NI-DSP™

Software Reference Manual for LabVIEW® for Windows

Digital Signal Processing Software for the PC

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About This Manual

The *NI-DSP Software Reference Manual for LabVIEW for Windows* explains how to use the NI-DSP software package for the LabVIEW for Windows environment. The NI-DSP software package contains the NI-DSP Analysis VIs, which are high-level digital signal processing (DSP) VIs that call the functions that execute on the AT-DSP2200 plug-in board for IBM AT bus and compatible computers. This manual describes how to use the NI-DSP Analysis VIs to develop applications in LabVIEW using the AT-DSP2200 board.

The NI-DSP software package also contains the NI-DSP Interface Utilities. The NI-DSP Interface Utilities are a set of tools and examples that help you customize your NI-DSP Analysis VIs and the DSP Library, which is resident on the board. This manual contains step-by-step instructions and useful examples to help the LabVIEW developer add custom algorithms to the NI-DSP Analysis VIs using the NI-DSP Interface Utilities.

Assumption of Previous Knowledge

The material in this manual is for users who are familiar with LabVIEW and the IBM family of computers and compatible computers.

Organization of This Manual

This manual is divided into four parts.

- Part 1, *Getting Started with NI-DSP*, contains a brief product overview, information about the NI-DSP for LabVIEW for Windows package, and the procedure for installing the software.

- Part 2, *Introduction to the NI-DSP Analysis VIs*, describes how to use the NI-DSP Analysis VIs in your LabVIEW applications. This part also describes how to manage memory on the DSP board from your LabVIEW application, and how to transfer data between your LabVIEW application and the NI-DSP functions on the board. This part contains general guidelines for developing NI-DSP applications within LabVIEW.

- Part 3, *NI-DSP Function Reference*, is intended as a reference for users familiar with Part 2. Part 3 is organized as follows:

  - Chapter 1, *NI-DSP Analysis VI Reference Overview*, contains an overview of the NI-DSP Analysis VIs and includes a list of the VIs. This chapter describes how the NI-DSP Analysis VIs are organized, and how to access them.

  - Chapter 2, *NI-DSP Analysis VI Reference,* contains a brief explanation of each NI-DSP Analysis VI. The VIs are arranged alphabetically.

- Part 4, *NI-DSP Interface Utilities*, explains how to customize the NI-DSP Analysis Library on the board and to create and run interfaces in LabVIEW to your custom library functions. Part 4 is organized as follows:

  - Chapter 1, *Introduction to the NI-DSP Interface Utilities*, contains an overview of the NI-DSP Interface Utilities, installation instructions, and explains how to use the NI-DSP Interface Utilities.

  - Chapter 2, *Getting Started with the NI-DSP Interface Utilities*, contains a step-by-step example for building a custom DSP Library, creating a LabVIEW interface to a custom function, and executing the custom function from the LabVIEW environment. The chapter demonstrates this concept with an example of how to add a custom function.

  - Chapter 3, *DSP Board Function Overview*, contains an overview of the prototypes of the C-callable NI-DSP Analysis functions on the DSP board that you can use in your custom programs.

  - Chapter 4, *Using the DMA VIs,* describes two special VIs that transfer data between the host computer and the DSP board without interfering with the DSP board.

- Appendix A, *Error Codes*, contains a list of the error codes returned by the NI-DSP Analysis VIs and the corresponding error messages.

- 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.

- The *Index* alphabetically lists topics covered in this manual, including the page where the topic can be found.

Conventions Used in This Manual

The following conventions are used in this manual:

<>	Angle brackets enclose the name of a key on the keyboard–for example, <enter>.
bold	Bold text denotes menus, command names, parameters, function panel items, error messages, and warnings.
DSP board	DSP board refers to the AT-DSP2200.
DSP Handle	DSP Handle refers to a 32-bit long integer code that constitutes an indirect reference to a buffer of memory in the memory space of an AT-DSP2200 board. The code contains information about the slot number of the board on which that buffer is allocated.
DSP Handle Cluster	DSP Handle Cluster refers to a cluster that constitutes two fields–a DSP Handle and a 32-bit size that indicates the number of elements the DSP Handle holds.
DSP Library	DSP Library refers to a common object file format (COFF) that constitutes NI-DSP software that resides and runs on the AT-DSP2200 board. The DSP Library consists of the Kernel, memory management routines, execution control routines, data communication routines, interrupt handling routines, data acquisition routines and a set of analysis functions.
<enter>	Key names are in lowercase letters.
Interface Library	Interface Library refers to the part of the NI-DSP software that resides on the PC and is linked with your application in order to communicate data to and from the DSP board. The Interface Library communicates with the DSP board using the driver.
italic	Italic text denotes emphasis, a cross reference, or an introduction to a key concept.
`italic monospace`	Italic text in this font denotes that you must supply the appropriate words or values in the place of these items.

LabVIEW Data Types — Each VI description includes a data type picture for each control and indicator, as illustrated in the following table:

Control	Indicator	Data Type
TF	TF	Boolean
abc	abc	String
I16	I16	Signed 16-bit integer
[I16]	[I16]	Array of signed 16-bit integers
I32	I32	Signed 32-bit integer
[I32]	[I32]	Array of signed 32-bit integers
SGL	SGL	32-bit floating-point number; by default, floating-point numbers are double precision
[SGL]	[SGL]	Array of 32-bit floating-point numbers
		Path
DSPHC	DSPHC	DSP Handle Cluster
[DSPHC]	[DSPHC]	Array of DSP Handle Clusters
		Error Cluster

`monospace` — 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, array names, structures, variables, filenames, and extensions, and for statements and comments taken from program code.

NI-DAQ — NI-DAQ is used throughout this manual to refer to the NI-DAQ software for DOS/Windows/LabWindows unless otherwise noted.

NI-DSP — NI-DSP is used throughout this manual to refer to the NI-DSP software for LabVIEW for Windows unless otherwise noted.

PC — PC refers to the IBM PC AT and compatible computers, and to EISA personal computers.

WE DSP32C tools — WE DSP32C tools is used throughout this manual to refer to the AT&T WE DSP32C Developer Toolkit.

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

Related Documentation

The following documentation available from National Instruments contains information that you may find helpful as you read this manual.

- *AT-DSP2200 User Manual*, part number 320435-01
- *LabVIEW Data Acquisition VI Reference Manual*, part number 320536-01
- *LabVIEW Getting Started Manual for Windows*, part number 320533-01
- *LabVIEW User Manual*, part number 320534-01
- *LabVIEW Utility VI Reference Manual*, part number 320543-01
- *NI-DAQ Function Reference Manual for DOS/Windows/LabWindows*, part number 320499-01
- *NI-DAQ Software Reference Manual for DOS/Windows/LabWindows*, part number 320498-01
- *NI-DSP Software Reference Manual for DOS/LabWindows*, part number 320436-01

The following documentation also contains information that you may find helpful as you read this manual:

- "A Computer Program for Designing Optimum FIR Linear Phase Digital Filters," McClellan, Parks, and Rabiner, *IEEE Transactions on Audio and Electroacoustics,* Vol. AU-21, No. 6, pp. 506-525, December 1973
- *Digital Filter Design,* Parks and Burrus, Wiley-Interscience
- *Discrete-Time Signal Processing,* Oppenheim and Schafer, Prentice Hall
- *Numerical Recipes,* Cambridge University Press

Additional Software

Additional DSP board software includes NI-DAQ for DOS/Windows/LabWindows and the Developer Toolkit.

NI-DAQ for DOS/Windows/LabWindows

Your AT-DSP2200 is shipped with the NI-DAQ for DOS/Windows/LabWindows software. NI-DAQ has a library of functions that can be called from your application programming environment. These functions include routines for analog input (A/D conversion), buffered data acquisition (high-speed A/D conversion), analog output (D/A conversion), waveform generation, digital I/O, counter/timer, SCXI, RTSI, and self-calibration. NI-DAQ maintains a consistent software interface among its different versions so you can switch between platforms with minimal modifications to your code. NI-DAQ comes with language interfaces for Professional BASIC, Turbo Pascal, Turbo C, Turbo C++, Borland C++, Microsoft C for DOS; and Visual Basic, Pascal, Microsoft C with SDK, and Borland C++ for Windows. NI-DAQ for DOS/Windows/LabWindows software is on high-density 5.25 in. and 3.5 in. diskettes. You can use the DSP board in conjunction with the National Instruments AT Series data acquisition boards and software to create a complete solution for integrated data acquisition and data analysis applications.

Developer Toolkit

The Developer Toolkit, an optional software package that you can purchase separately from National Instruments, is required for building custom libraries with the NI-DSP Interface Utilities. The Developer Toolkit contains an AT&T C compiler, assembler, linker, and documentation. With these tools, you can program AT Series DSP boards directly, using theboard flexibility to custom tailor the DSP Library. The C compiler optimizes WE DSP32C code and generates assembly language code that can be assembled and linked into a run-time module. When a run-time module is completed, use the download tools and the debugger to load, debug, and execute the code, set parameters, and report results. The Developer Toolkit also includes the *WE DSP32C Support Software Library User Manual* and the *WE DSP32 and WE DSP32C Language Compiler Library Reference Manual.*

Compatible Hardware

You can use DSP boards in conjunction with the National Instruments AT Series data acquisition boards. In particular, the National Instruments AT-DSP2200 is a high-performance, DSP board with high-accuracy audio frequency (DC to 51.2 kHz) analog input/output for the PC. The AT-DSP2200 gives the PC a dedicated high-speed numerical computation engine that can perform scientific calculations faster than the general-purpose 80x86 microprocessor on the PC.

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.

Part 1
Getting Started with NI-DSP

This part contains a brief product overview, information about the NI-DSP for LabVIEW for Windows package, and the procedure for installing the software.

Product Overview

The NI-DSP software package comes with a set of LabVIEW VIs that invoke the digital signal processing (DSP) board-resident high-performance functions that efficiently process large blocks of numerical data and perform numerically intensive computations. The NI-DSP Analysis VIs include numerical analysis, signal generation, DSP, windowing, digital filtering, and memory management that are suitable for simulation, modeling, and sophisticated data processing.

You can use NI-DSP to develop programs in the LabVIEW for Windows environment. This software comes with the NI-DSP Interface Utilities so you can customize the DSP Library by adding functions to or deleting functions from the Analysis Library on the DSP board and/or add interfaces to these custom functions in LabVIEW.

Figure 1-1 shows the development path for NI-DSP in the LabVIEW environment.

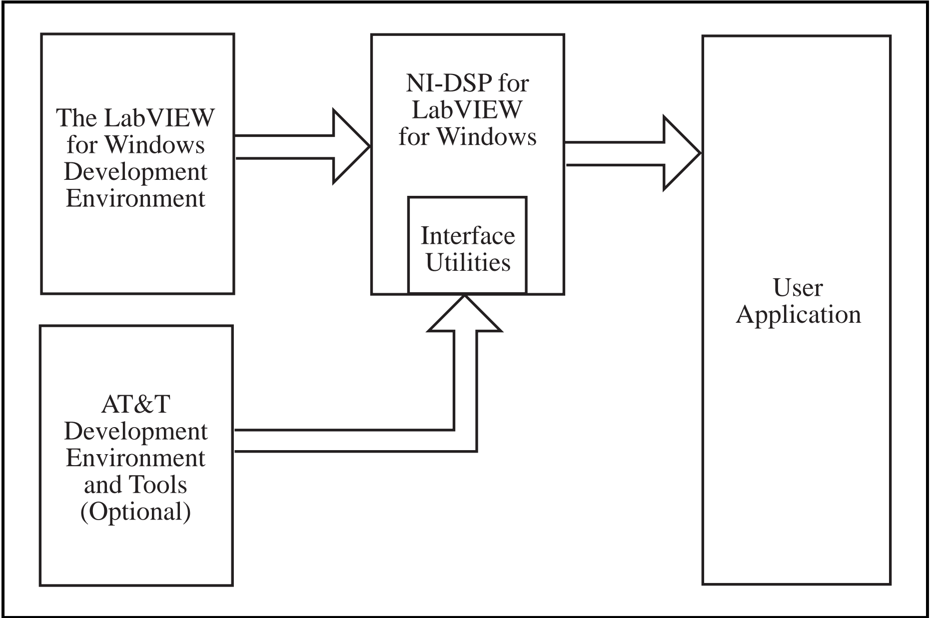


Figure 1-1. Development Paths with the NI-DSP Software

The NI-DSP Software

The NI-DSP software consists of the NI-DSP for LabVIEW for Windows diskettes.

The NI-DSP software contains a Warranty Registration Form. Please fill out this form and return it to National Instruments. The Warranty Registration Form entitles you to receive product upgrades and technical support.

What Your Distribution Diskettes Should Contain

The NI-DSP software package contains the NI-DSP for LabVIEW for Windows Disks (for licensed LabVIEW for Windows users). If your kit is missing any of these components, contact National Instruments.

Installing NI-DSP for LabVIEW for Windows

Note: NI-DSP for LabVIEW for Windows is intended for use with the standard LabVIEW for Windows software. *You must* install LabVIEW for Windows before installing NI-DSP. You must install the Data Acquisition Library of LabVIEW to run the NI-DSP software.

Before beginning the software installation, make backup copies of the NI-DSP for LabVIEW for Windows distribution disks. Copy each disk onto a correctly labeled backup disk and store the original distribution disks in a safe place.

You can install NI-DSP for LabVIEW for Windows from the DOS prompt, the Windows File Manager, or with the **Run...** command from the **File** menu of the Program Manager.

1. Insert Disk 1 into the disk drive and run the `SETUP.EXE` program on Disk 1 using one of the following three methods.

   - From the DOS prompt, type `X: \SETUP` (where X is the proper drive designation).
   - From Windows, select **Run...** from the **File** menu of the Program Manager. A dialog box appears. Type `X: \SETUP` (where X is the proper drive designation).
   - From Windows, launch the File Manager. Click on the drive icon that contains Disk 1. Find `SETUP.EXE` in the list of files on that disk and double-click on it.

2. The installer gives you the option of performing a full installation or a custom installation. Unless you do not have sufficient disk space (approximately 4 megabytes), National Instruments recommends that you perform a full installation. After you choose an installation, follow the instructions that appear on the screen.

After calling `SETUP`, the appropriate directories are created and the needed files are copied to your hard drive. `SETUP` can also install the NI-DSP Interface Utilities, discussed in Part 4, *NI-DSP Interface Utilities*, of this manual. If you choose Full Installation, `SETUP` does the following things:

1. `SETUP` creates a subdirectory called `DSP2200` of the `vi.lib` subdirectory of the `LabVIEW` directory. `SETUP` decompresses the NI-DSP Analysis VIs (as LabVIEW `.LLB` files) in the `DSP2200` directory.

2. `SETUP` copies `DSP.DLL` to your `windows` directory.

3. `SETUP` creates the subdirectories shown in Table 1-1 under the directory you specified during setup. These subdirectories make up the NI-DSP Interface Utilities.

4. `SETUP` creates a subdirectory called `DSP2200` of the `EXAMPLES` subdirectory of the `LabVIEW` directory. `SETUP` copies all of the NI-DSP VI examples there.

Table 1-1. Subdirectories Created by `SETUP`

Subdirectory Name	Description
`C:\NIDSP\LIB`	Library files for linking with stand-alone programs
`C:\NIDSP\DISPATCH`	The Dispatch utility and related files
`C:\NIDSP\EXAMPLES`	Contains source code for the examples

NIDSP is the name you specify during setup.

The SETUP program prompts you for information including the drive letter and directory in which you have installed the standard LabVIEW package. The program also verifies that your hard disk has enough space to hold the NI-DSP for LabVIEW for Windows files.

If you choose Custom Installation, SETUP installs only the files you specify.

Board Configuration

There are several board configuration parameters that must be established before an NI-DSP application can execute properly. These parameters are the board ID number, the board subtype, the base address, the interrupt level, the DMA channel, and the pathname of the DSP Library files. These parameters are established differently depending on whether you are installing the AT-DSP2200 in an ISA (or AT) bus computer or an EISA bus computer.

Installation on an ISA (or AT) Bus Computer

A configuration utility is supplied with the LabVIEW data acquisition software for establishing all the configuration parameters on ISA bus computers. This utility, called WDAQCONF.EXE, saves the configuration parameters in a file named WDAQCONF.CFG. Use the WDAQCONF utility to assign a board ID number to your AT-DSP2200, to choose the memory subtype (either 64 Kwords or ≥128 Kwords), to set the base address, interrupt level, and DMA channel, and to specify the pathname of the DSP Library file.

Two DSP Library files are supplied with your NI-DSP software–LV2200S.OUT and LV2200.OUT. LV2200S.OUT is intended for use with the 64 Kword version of the AT-DSP2200. LV2200.OUT is intended for use with any other version of the board. With the WDAQCONF utility, you can enter a complete path that will include the DSP Library file name. If you enter complete pathnames, you can configure the NI-DSP software to automatically load custom versions of the DSP Library files.

If you installed the Data Acquisition Library of LabVIEW, you can find WDAQCONF.EXE of your LabVIEW directory. From Windows, you run WDAQCONF.EXE by double-clicking on its icon.

Installation on an EISA Bus Computer

Installing the AT-DSP2200 board on an EISA bus computer involves two different configuration utilities.

- A system configuration utility is supplied with your computer by your computer vendor. This utility, along with the !NIC1100.CFG EISA configuration file installed by the LabVIEW SETUP program, is used to assign a slot number to the AT-DSP2200, to choose the memory subtype (either 64 Kwords or ≥128 Kwords), and to set the base address, interrupt level, and DMA channel.

  There are typically two methods for running your EISA system configuration utility.

  1. Boot from the diskette containing the utility and place a copy of the !NIC1100.CFG file on this diskette.

  2. Run the utility from a directory on your hard disk and place a copy of the !NIC1100.CFG file in this directory. The EISA system configuration utility is typically named CF.EXE.

- You must use the WDAQCONF utility to enter the DSP Library file pathname. You can only do this after completing the EISA system configuration. Refer to the information concerning the establishment of this pathname in the previous section titled, *Installation on an ISA (or AT) Bus Computer*.

Before using NI-DSP, you must run WDAQCONF.EXE to configure your DSP board.

For more information about board configuration, refer to Chapter 1, *Introduction and Configuration*, of the *LabVIEW Data Acquisition VI Reference Manual*.

Part 2
Introduction to the NI-DSP Analysis VIs

This part describes how to use the NI-DSP Analysis VIs in your LabVIEW applications. This part also describes how to manage memory on the DSP board from your LabVIEW application, and how to transfer data between your LabVIEW application and the NI-DSP functions on the board. This part contains general guidelines for developing NI-DSP applications within LabVIEW.

Using the NI-DSP VIs in LabVIEW

LabVIEW users use the NI-DSP Analysis VIs as if they were any other standard VIs, as described in the *LabVIEW User Manual.* Notice, however, that the NI-DSP Analysis VIs run analysis code on the DSP board rather than on the host CPU. One of the major features includes the memory management and data transfer VIs, which are discussed in detail in the section titled *Memory Management and Data Transfer* later in this chapter.

AT-DSP2200 Software Overview

The AT-DSP2200 board, working in conjunction with your personal computer, is a powerful numeric processor for high-speed analysis of data arrays. The NI-DSP for LabVIEW for Windows software includes a number of utilities and low-level memory management and data transfer VIs that facilitate communication between the DSP board and the host computer. Figure 1-1 is a block diagram of the software utilities and libraries that control the AT-DSP2200. The DSP Library and the low-level memory management and data transfer functions reside and execute on the board. You can customize the DSP Library to optimize performance, as described in Part 4, Chapter 2, *Getting Started with the NI-DSP Interface Utilities*, of this manual. Your application programs and the Interface Library, however, reside on the host PC.

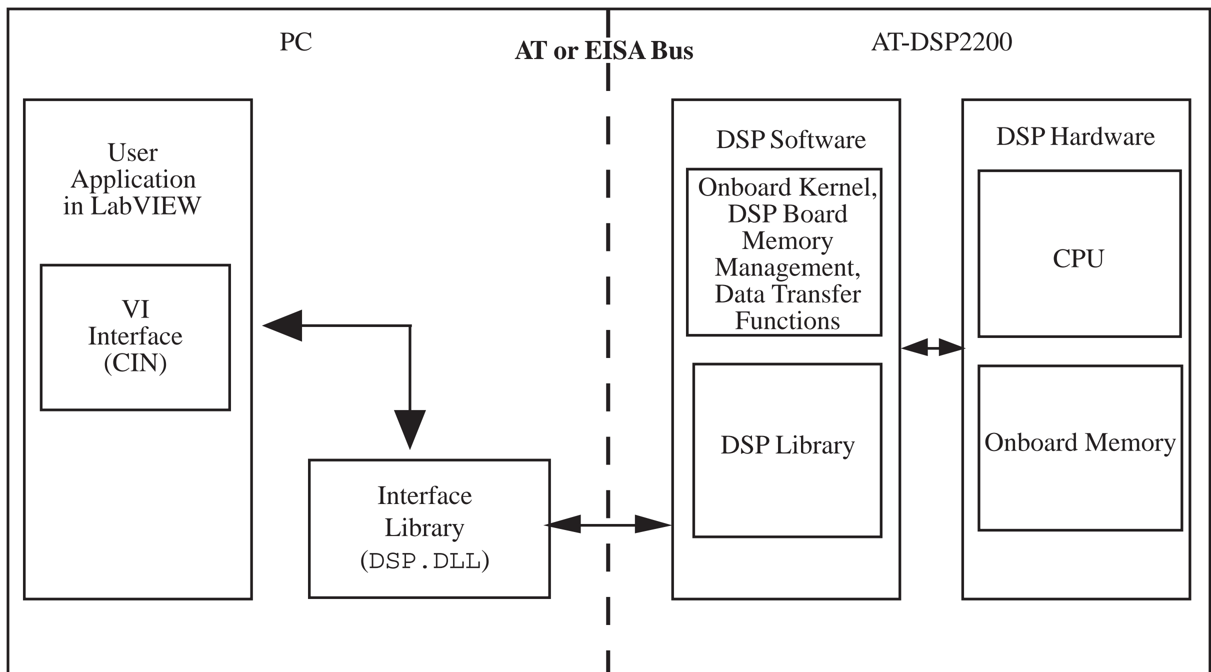


Figure 1-1. Communication between the PC and the DSP Board

The AT-DSP2200 can process large amounts of data, separately and distinctly from the host PC processor. The board consists not only of a signal processing chip, but also memory where data that the board processes must reside. The AT-DSP2200 does not have access to memory locations on the host PC. Therefore, you must download all data from your application programs to DSP board memory before processing it.

The Interface Library, `DSP.DLL`, and the Code Interface Node (CIN) interface, which reside on the PC, serve as a bridge between your application programs in LabVIEW and the DSP software running on the board. When you call an NI-DSP Analysis VI, the VI passes the parameters to the CIN first, which then passes the parameters to the Interface Library, `DSP.DLL`. The Interface Library determines what type of parameters are being passed, decides how to set up the data in DSP board memory, and then calls the actual functions that will run on the board.

When a function on the DSP board processes data, it assumes the data is resident in DSP board memory. Because transferring data between the PC and the DSP board slows down processing, none of the NI-DSP Analysis VIs transfer data back and forth internally except the data transferring VIs. The NI-DSP Analysis VIs process the data buffers that are already on the board and leave the results on the board.

If the data buffer you want to process using the DSP board is in PC memory, you must copy the data to the DSP board before you call a function on the DSP board to process the data. To see the results, you must then copy the data back to the PC. Several special NI-DSP Analysis VIs perform these transfers. For scalars, the NI-DSP Analysis VIs automatically perform the transfer for you.

The representation of data buffers in the NI-DSP Analysis VIs is not the normal LabVIEW data array representation because the data buffers indicate the data location on the DSP board instead of the PC address. A special structure, called a DSP Handle Cluster, represents the data buffer on the DSP board from LabVIEW. The DSP Handle Cluster is a coded DSP board memory address that indicates where the data buffer is on the DSP board. You must call the DSP Allocate Memory VI to obtain a valid DSP Handle Cluster. Several VIs can manage the memory on the DSP board. You can allocate and deallocate memory on the DSP board using these VIs. The next section, *Memory Management and Data Transfer,* discusses the VIs used to allocate memory and transfer data buffers to and from the DSP board.

Memory Management and Data Transfer

This section describes how to manage memory on the DSP board from your LabVIEW application and how to transfer data between your LabVIEW application and the DSP board.

The NI-DSP for LabVIEW package contains a set of VIs that manage memory space on the DSP board and help improve data transfers between the DSP board and your application. There are VIs for allocating memory buffers on the DSP board, for indexing into previously allocated buffers, for deallocating buffers and for copying data between DSP and LabVIEW. The following VIs, described in greater detail in Part 3 of Chapter 2, *NI-DSP Analysis VI Reference*, handle memory management on the DSP board and data transfers between the DSP board and your LabVIEW application:

- Copy Mem[DSP to DSP]
- Copy Mem[DSP to LV]
- Copy Mem[LV to DSP]
- DSP Allocate Memory
- DSP Free Memory
- DSP Index Memory
- DSP Init Memory

The DSP Allocate Memory VI allocates memory buffers on the DSP board and returns a DSP Handle Cluster, which has two fields that uniquely describe this buffer–a DSP Handle and a size.

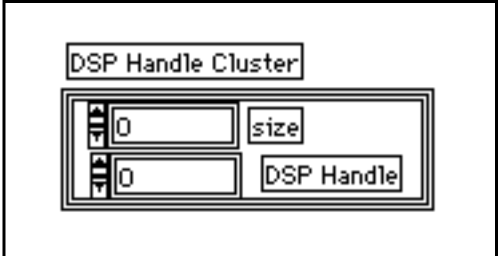


Figure 1-2. DSP Handle Cluster

DSP Handle is a 32-bit integer containing information that indicates the board on which the allocated buffer resides, and an index into an onboard Memory Look Up Table (MLUT) that holds the actual DSP address of the buffer that this handle represents. Figure 1-3 shows how a DSP Handle is encoded. The size field holds the number of elements in this buffer. An element can be 4 bytes (for 32-bit floating-point data or long integer data) or 2 bytes (for 16-bit integer data) depending on the bytes per element selector used in the DSP Allocate Memory VI.

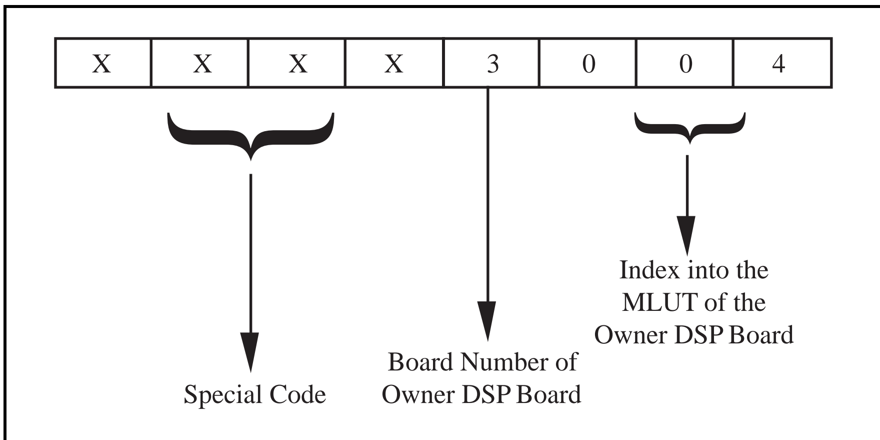


Figure 1-3. The Hexadecimal Encoding of a Typical DSP Handle

The first four hexadecimal numbers (upper 16 bits) of the DSP Handle, shown in Figure 1-3, are a special value. The interface code for a particular function that your application calls decodes these four hexadecimal numbers to determine if the argument is a valid DSP Handle.

Notes: Do not change the value of a DSP Handle Cluster. Keep in mind that a DSP Handle Cluster is just an entry of a table that indicates where the data buffer is on the DSP board. If you want to operate on part of the data in that buffer, use the DSP Index Memory VI or the DSP Subset VI to obtain a new DSP Handle Cluster to hold the part of the data**.** Then operate on the new DSP Handle Cluster.

The Memory Look-Up Table (MLUT) has only 128 entries. You can allocate only a total of 128 different DSP Handle Clusters. Although you might have physical memory on the DSP board, you will get an error message for not having enough memory if you already have 128 DSP Handle Clusters in use. Free the DSP Handle Clusters that are not in use. The DSP Init Memory VI will free all DSP Handle Clusters on the specified DSP board.

Figures 1-4 and 1-5 show how to allocate a DSP Handle Cluster of 2,048 4-byte-long elements on board 3. The board number on which the buffer is allocated is important for determining the ownership of the buffer. When making a VI call, the same DSP board on which the function is to execute must own all of the DSP Handle Clusters or an error code is returned. Only the DSP Allocate Memory VI and few other VIs that do not have DSP Handle Clusters as input parameters have a board slot parameter. VIs that have DSP Handle Clusters as input parameters obtain the board slot information from their own DSP Handle Clusters. All of the DSP Handle Clusters should have the same slot information, because the DSP VIs assume that all are executing on the same DSP board.

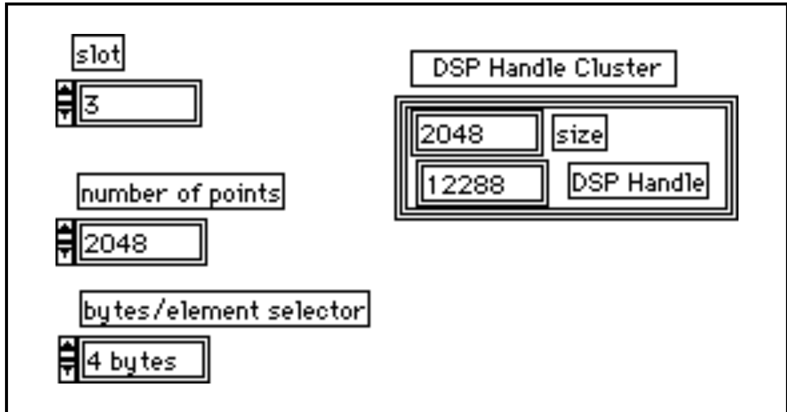


Figure 1-4. Front Panel–An Example of How to Allocate a DSP Handle Cluster

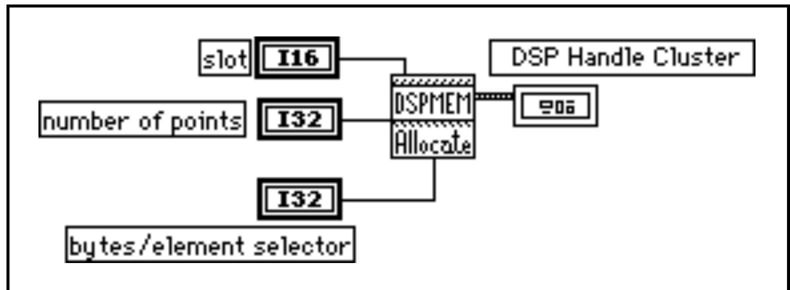


Figure 1-5. Block Diagram–An Example of How to Allocate a DSP Handle Cluster

For all of the NI-DSP Analysis VIs, the array data type is DSP Handle Cluster. Before you call any of these VIs, call the DSP Allocate Memory VI to obtain a valid DSP Handle Cluster, which you then use as a reference to your data buffer. The Analysis VIs assume that the data is already on the board and stores the results on the board. If you want to copy data between the PC and the DSP board, use either the Copy Mem(LV to DSP) VI or the Copy Mem(DSP to LV) VI to copy data back and forth.

If you use the DSP Allocate Memory VI in your program, use the DSP Free Memory VI to free buffers allocated when you do not need them any more. The board holds these allocations in memory even after your application has completed or you exit LabVIEW unless you execute the DSP Init Memory VI or reload the DSP Library. Thus, it is important to free all buffers your application allocated before you exit the application or you may run out of memory on the board.

Special Features of the NI-DSP Analysis VIs

This section describes the special features of the NI-DSP Analysis VIs that make them different from other LabVIEW VIs.

- **DSP Handle Cluster in/out**. The way you specify the output data buffers for NI-DSP Analysis VIs is different from the way you would specify output data buffers for other LabVIEW VIs. DSP Handle Clusters also represent all the output data buffers in the NI-DSP Analysis VIs. To use valid DSP Handle Clusters for the NI-DSP VI output data buffers, you must use the DSP Allocate Memory VI to obtain the DSP Handle Clusters before you use them. Supply all of the output DSP Handle Clusters as inputs to tell the DSP board where the output buffers are. Every output DSP Handle Cluster is identical to the corresponding input DSP Handle Cluster. The two DSP Handle Clusters are internally connected. The output is an indicator. The input is a control. For example, in the DSP Add VI, shown in Figure 1-6, **Z** is the DSP Handle Cluster that indicates where to store the results. You use **Z in** to connect to a valid DSP Handle Cluster that you previously allocated. **Z in** tells the DSP board where the results will be stored. **Z out** is the location where the results have already been stored. Because **Z in** and **Z out** are the same DSP Handle Cluster, you need to free only one of them when you want to deallocate their DSP board buffer.

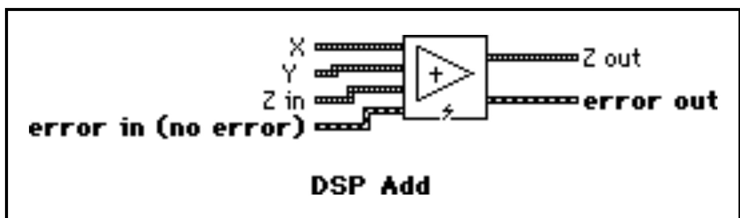


Figure 1-6. DSP Add VI

All of the controls and indicators in the NI-DSP Analysis VIs follow the **Z in**, **Z out** naming convention and work in the same way as previously described in the example, except for the **error in/error out** cluster.

- **Error Handling**. All of the NI-DSP Analysis VIs have an error input and an error output for managing and reporting errors. The **error in/error out** cluster used by the NI-DSP VIs and many other high-level I/O operations is a cluster containing a Boolean indicating whether the data should be treated as an error, a 32-bit error code, and a descriptive string that usually contains the name of the source of the error. The **error in/error out** cluster is shown in Figure 1-7.

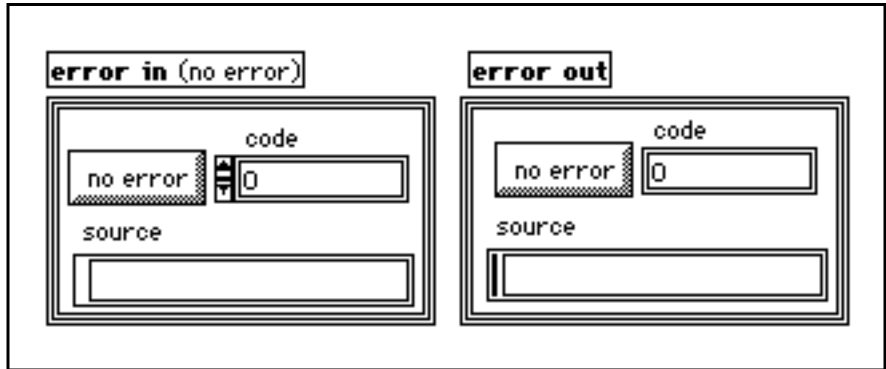


Figure 1-7. The **error in/error out** Cluster

The **error in/error out** cluster contains the following elements:

TF The boolean value is true if an error occurred, false if no error occurred.

I32 **code** is the error code.

abc **source** is the source of the error. If an error occurs during execution, the VI sets **source** to the name of the VI that produced the error.

Every VI checks **error in** first. If there is an error, the VI does not execute any DSP code but simply passes the contents of **error in** to the **error out** cluster. If there is no error, the VI executes. One advantage of this **error in/error out** design is that you can connect several I/O operations together so that, if an error occurs, subsequent VIs do not perform undesired actions. DSP Free Memory will execute even if an error occurs. This ensures that allocated buffers are freed even if an error occurred.

Another advantage of this **error in/error out** design is that you can establish the order of a set of operations, even if there is no other data flow between the operations. Connecting the **error out** of the first VI to the **error in** of the second VI establishes data flow and therefore execution order. You could do the same thing with a Sequence structure, but with the **error in/error out** design, you can establish the order with all of the operations at the top level of the block diagram. For example, in Figure 1-8, you allocate DSP Handle Clusters **X** and **Y** as inputs, and you want to free **X** and **Y** after the DSP Add VI has been executed. If you simply connect **X** to the DSP Free Memory VI as shown in Figure 1-8, there is no sequential order between the DSP Add VI and the DSP Free Memory VI. If the DSP Free Memory VI executes first, the DSP Add VI will receive an invalid handle because that DSP Handle Cluster was deallocated.

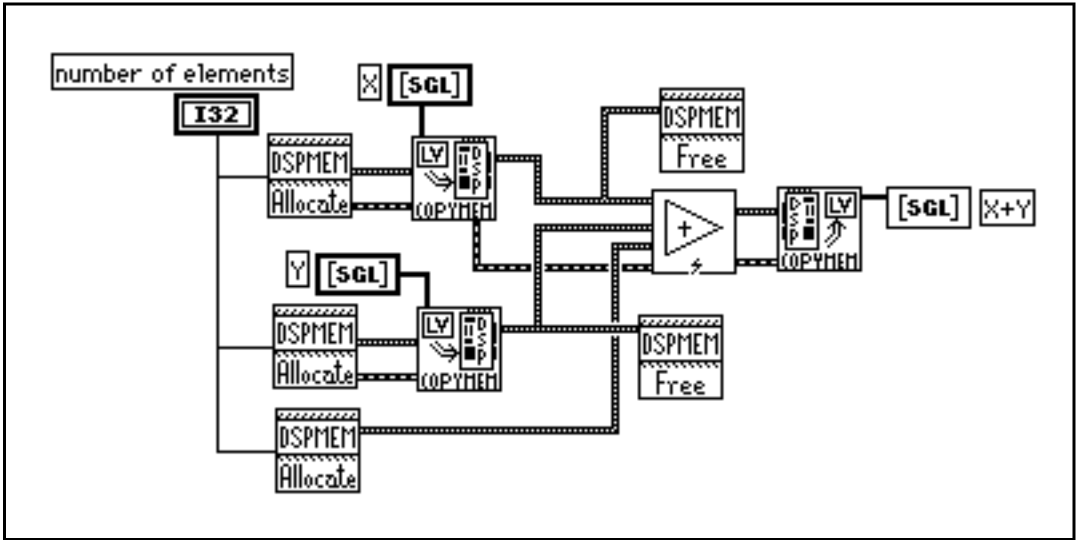


Figure 1-8. An Example That Does Not Use **error in/error out** for Sequencing VIs

If you connect the VIs as shown Figure 1-9 instead, you ensure that the DSP Add VI executes before the DSP Free Memory VI.

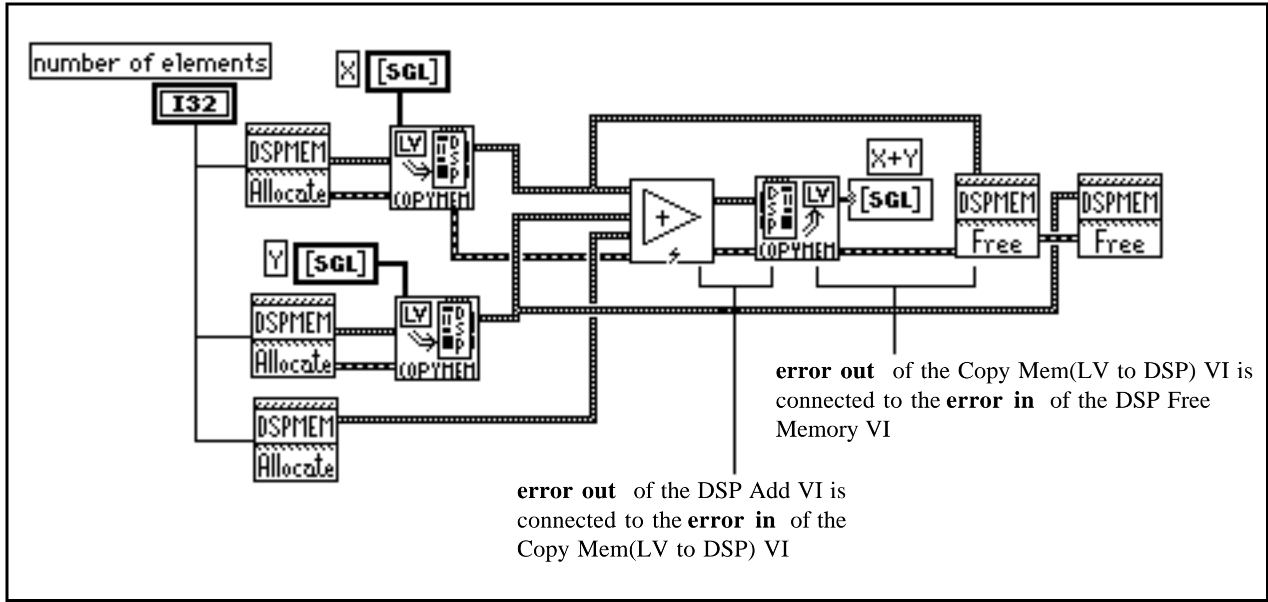


Figure 1-9. An Example of Using the **error in/error out** Cluster for Sequential VI Execution

For more information about the **error in/error out** cluster, refer to Chapter 2, *Error Handler VIs*, in the *LabVIEW Utility VI Reference Manual.*

Hints for Improving the Execution Speed on the DSP Board

Check each of the following things to maximize your DSP board performance:

- Allocate as many of the DSP Handle Clusters as you can before you operate on the data. Keep all data on the DSP board until you finish all of the processing. Reduce the number of data transfers between the DSP board and the PC as much as possible. The functions that run on the DSP board are very fast, but transferring data between the DSP board and the PC and memory allocation slows the total processing performance.

- Use the **error in/error out** cluster for sequencing VI execution. Be sure all of your VIs run in the correct sequence. Use the **error in/error out** cluster to propagate the errors. If an error occurs, you can tell where the error happens. You can use the LabVIEW error handler VIs in the **Utility** option of the **Functions** menu to obtain pop-up error messages. Refer to the *LabVIEW Utility VI Reference Manual* for more information about these VIs.

- Many analysis routines on the DSP board can be performed in place; that is, the input and output array can be the same array. This is very important to remember when you are processing large amounts of data. Large 32-bit floating-point arrays consume a lot of memory. If the results you want do not require that you keep the original array or an intermediate array of data, perform analysis operations in place whenever possible. For example, use the same DSP Handle Cluster for the input and output data buffers in your diagram in LabVIEW. This will save your DSP board memory.

- Several intermediate-level data acquisition VIs work with DSP Handle Clusters. These VIs can acquire data and leave it on the board. You can use the NI-DSP Analysis VIs to operate on this data and then copy the processed results back to the PC. In this way, you dramatically reduce the data transfer overhead between the PC and the DSP board, and improve the overall performance. For more information about these data acquisition VIs, refer to the *LabVIEW Data Acquisition VI Reference Manual.* An example that shows you how to use a DSP Handle Cluster to acquire data and process this data on the DSP board can be found in the `DSP2200` subdirectory of the `EXAMPLES` subdirectory of your `LabVIEW` directory.

An Example of Using NI-DSP Analysis VIs

Figures 1-10 and 1-11 show the front panel and block diagram, respectively, of an example using NI-DSP Analysis VIs.

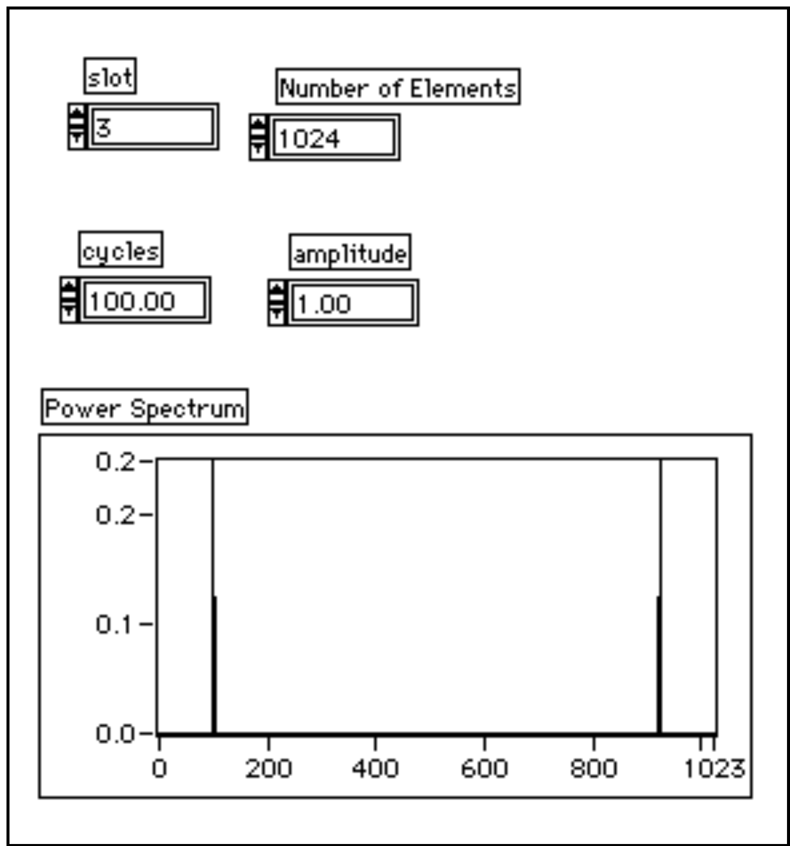


Figure 1-10. Front Panel–An Example of Using NI-DSP Analysis VIs

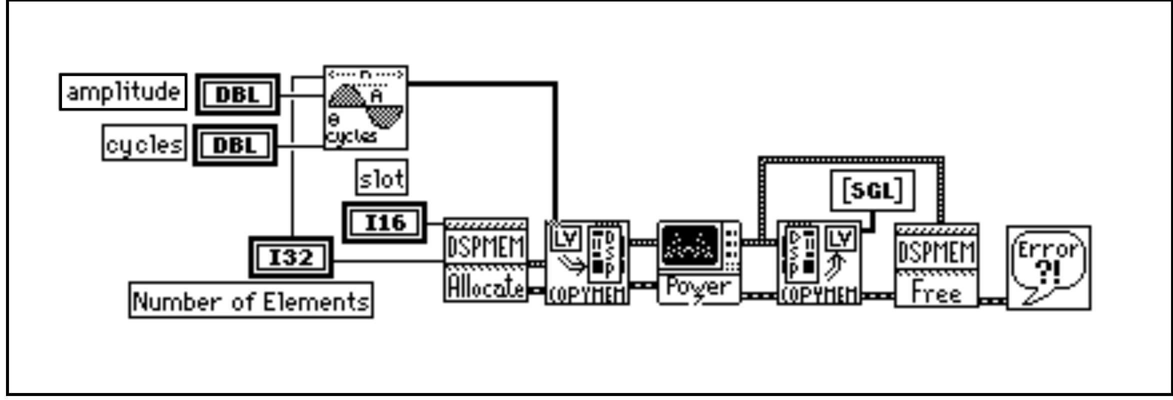


Figure 1-11. Block Diagram–An Example of Using NI-DSP Analysis VIs

This example shows you how to obtain the power spectrum of a sine wave signal. First, generate a sine wave that you want to analyze using the LabVIEW Analysis VIs, then use the Copy Mem(LV to DSP) VI to copy the data of this sine signal to the DSP board. Before you copy the data, you must call the DSP Allocate Memory VI to allocate a DSP Handle Cluster that reserves a data buffer on the DSP board. Connect this DSP Handle Cluster to the **destination in** terminal of the Copy Mem(LV to DSP) VI to indicate where the data will be stored on the DSP board. After the data is copied to the DSP board, call the DSP Power Spectrum VI to perform a power spectrum on the data. After you finish the analysis, the results are stored in the data buffer indicated by the DSP Handle Cluster you previously allocated. If you want to see the results, call the Copy Mem(DSP to LV) VI to copy data back to LabVIEW. Figure 1-10 shows the power spectrum of a sine wave. The last step is to call the DSP Free Memory VI to free the DSP Handle Cluster that you allocated. This example connects all of the **error out** clusters of the previous VIs to the **error in** clusters of the subsequent VIs to establish the proper sequence and to pass through an error, should it occur, without executing the rest of the VIs.

Chapter 1
NI-DSP Analysis VI Reference Overview

This chapter contains an overview of the NI-DSP Analysis VIs and includes a list of the VIs. This chapter describes how the NI-DSP Analysis VIs are organized and how to access them.

The NI-DSP Analysis VI Overview

The NI-DSP Analysis VIs are a set of high-performance VIs that efficiently process large blocks of numerical data and perform numerically intensive computations. The NI-DSP Analysis VIs include numerical analysis, signal generation, digital signal processing, digital filtering, and windowing operations that are suitable for simulation, modeling, and sophisticated data processing.

The NI-DSP Analysis VIs are presented in alphabetical order in Chapter 2, *NI-DSP Analysis VI Reference*. Table 1-1 lists these VIs by group.

Table 1-1. The NI-DSP Analysis VI Groups

Group	VI
Signal Generation	
	DSP Sine Pattern
	DSP Pulse Pattern
	DSP Impulse Pattern
	DSP Impulse Train Pattern
	DSP Ramp Pattern
	DSP Sinc Pattern
	DSP Square Pattern
	DSP Triangle Pattern
	DSP Triangular Train
	DSP Sawtooth Pattern
	DSP Uniform White Noise
	DSP Random Pattern
	DSP Gaussian White Noise
Frequency Domain	
	DSP ReFFT
	DSP Complex FFT
	DSP Inverse FFT
	DSP Power Spectrum
	DSP Cross Power
	DSP FHT
	DSP Inverse FHT
	DSP Zero Padder
Time Domain	
	DSP Convolution
	DSP Deconvolution
	DSP Correlation
	DSP Decimate
	DSP Derivative
	DSP Integral

(continues)

Table 1-1. The NI-DSP Analysis VI Groups (Continued)

Filters
- DSP Butterworth Coefficients
- DSP Chebyshev Coefficients
- DSP Inverse Chebyshev Coeff
- DSP Elliptic Coefficients
- DSP IIR Filter
- DSP Equi-Ripple LowPass
- DSP Equi-Ripple HighPass
- DSP Equi-Ripple BandPass
- DSP Equi-Ripple BandStop
- DSP Parks-McClellan
- DSP Median Filter

Windows
- DSP Blackman Window
- DSP Exact Blackman Window
- DSP Blackman Harris Window
- DSP Hanning Window
- DSP Hamming Window
- DSP Flat Top Window
- DSP General Cosine Window
- DSP Exponential Window
- DSP Force Window
- DSP Kaiser-Bessel Window
- DSP Triangular Window

Array Functions
- DSP Add
- DSP Subtract
- DSP Multiply
- DSP Divide
- DSP Absolute
- DSP Square Root
- DSP Product
- DSP Sum
- DSP Log
- DSP Clip
- DSP Reverse
- DSP Shift
- DSP Sort
- DSP Linear Evaluation
- DSP Max & Min
- DSP Polynomial Evaluation
- DSP Subset
- DSP Set
- DSP Unwrap
- DSP Polar to Rectangular
- DSP Rectangular to Polar

Memory Management
- DSP Allocate Memory
- Copy Mem(LV to DSP)
- Copy Mem(DSP to LV)
- Copy Mem(DSP to DSP)
- DSP Free Memory
- DSP Index Memory
- DSP Init Memory

Table 1-1. The NI-DSP Analysis VI Groups (Continued)

Utility Functions
`DSP Reset`
`DSP Load`
`DSP Start`
`DSP Timeout`
`DSP Custom`
`DSP DMA Copy(DSP to LV)`
`DSP DMA Copy(LV to DSP)`
`DSP Handle to Address`

Analysis VI Organization

After installation, the NI-DSP Analysis VIs reside in the following VI library files (LabVIEW `.LLB` files) within the **DSP2200** option:

- **Signal Generation** contains VIs that generate digital patterns.
- **Frequency Domain** contains VIs that perform frequency domain transformations, frequency domain analysis, and other transforms such as the Hartley transform.
- **Time Domain** contains VIs that perform direct time series analysis of signals.
- **Filters** contains VIs that perform IIR and FIR filtering functions.
- **Windows** contains VIs that perform smoothing windowing.
- **Array** contains VIs that perform arithmetic operations on arrays.
- **Memory Management** contains VIs to perform allocating, indexing, copying, and freeing memory on the AT-DSP2200 board.
- **Utility** contains VIs for controlling the operation of the AT-DSP2200 board.

After installation, the eight analysis VI libraries appear in the **Functions** menu in the order shown in the preceding list under the **DSP2200** option**.** You can reorganize the folders and the VIs to suit your needs and applications.

Accessing the NI-DSP Analysis VIs

To access the analysis VIs from the block diagram window, choose **DSP2200** from the **Functions** menu as shown in Figure 1-1, proceed through the hierarchical menus, and select the VI you want. The icon corresponding to that VI appears in the block diagram and is ready to be wired.

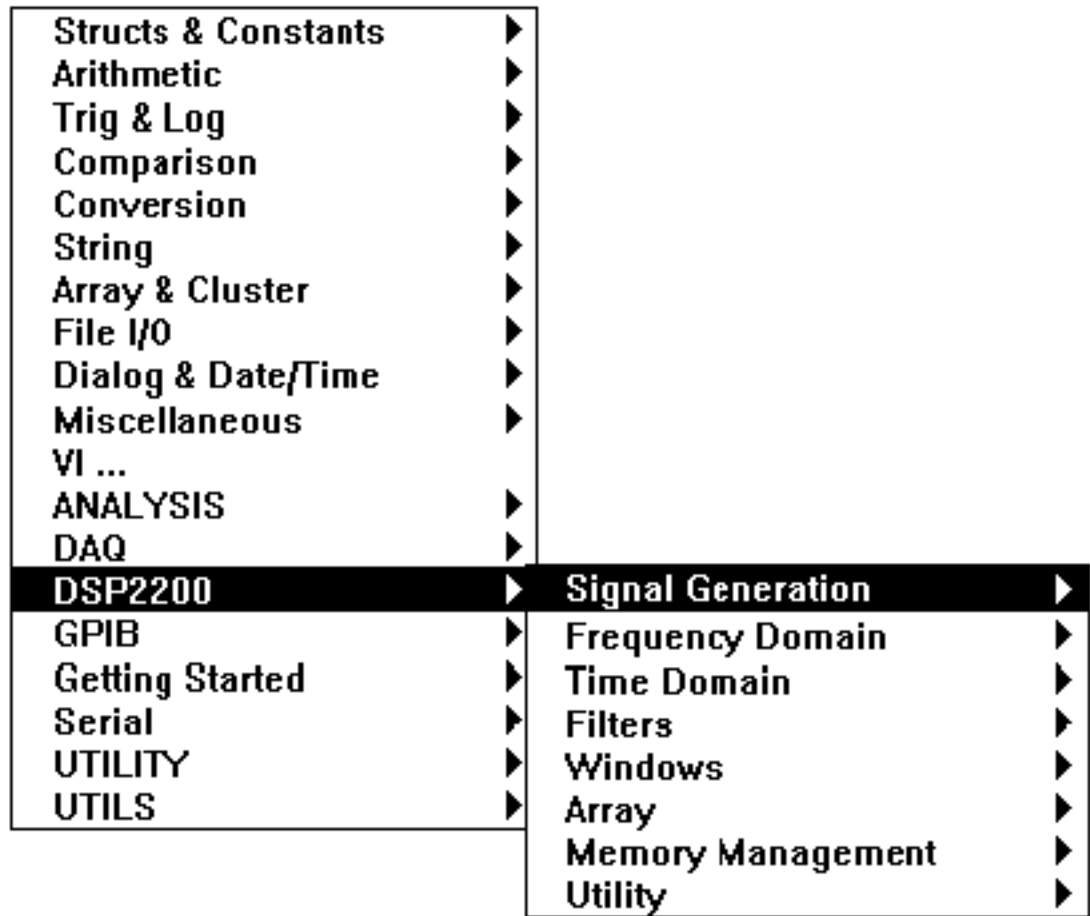


Figure 1-1. Choosing DSP2200 from the Functions Menu

About the Fast Fourier Transform (FFT)

The VIs in the Frequency Domain group are based upon the discrete implementation and optimization of the Fourier Transform integral. The Discrete Fourier Transform (DFT) of a complex sequence X containing n elements is obtained using the following formula:

$$Y[i] = \sum_{k=0}^{n-1} X[k] * \exp(-j2\pi ik / n), \quad \text{for } i = 0, 1, \ldots, n-1$$

where Y[i] is the ith element of the DFT of X and $j = \sqrt{-1}$.

The DFT of X also results in a complex sequence Y of n elements. Similarly, the Inverse Discrete Fourier Transform (IDFT) of a complex sequence Y containing n elements is obtained using the following formula:

$$X[i] = (1/n) \sum_{k=0}^{n-1} Y[k] * \exp(j2\pi ik / n), \quad \text{for } i = 0, 1, \ldots, n-1$$

where X[i] is the ith element of the IDFT of Y and $j = \sqrt{-1}$.

The discrete implementation of the DFT is a numerically intense process. However, it is possible to implement a fast algorithm when the size of the sequence is a power of two. These algorithms are known as FFTs, and can be found in many introductory digital signal processing (DSP) texts.

The resulting complex FFT sequence has the conventional DSP format as described in this section.

If there are n number of elements in the complex sequence and k = n/2, then the output of the FFT is organized as follows:

Y[0]	DC component
Y[1]	Positive first harmonic
Y[2]	Positive second harmonic
:	:
Y[k-1]	Positive k-1 harmonic
Y[k]	Nyquist frequency
Y[k+1]	Negative k-1 harmonic
:	:
Y[n-2]	Negative second harmonic
Y[n-1]	Negative first harmonic

The following conventions and restrictions apply to the VIs in the Frequency Domain folder:

- All arrays must be a power of two: $n = 2^m$, m = 1, 2, 3, …, 24 (limited by onboard memory).
- Complex sequences are manipulated using two arrays. One array represents the real elements. The other array represents the imaginary elements.

The following notation is used to describe the FFT operations performed in the Frequency Domain class:

- Y = FFT {X}, the sequence Y is the FFT of the sequence X.
- Y = FFT^{-1} {X}, the sequence Y is the inverse FFT of the sequence X.

X is usually a complex array but can be treated as a real array.

About Filtering

All of the VIs in the Filters group are digital filters that can be represented by the computational algorithm that best describes the relationship between the input and output discrete time sequences. This computational algorithm is referred to as the Linear Constant Coefficient Difference Equation. This equation relates the input and output by the basic operations of addition, delay and multiplication. The following equation relates the input and output sequences x and y, respectively at the discrete time instant n:

$$a(0)*y(n) = \sum_{i=0}^{i=N-1} (x[n-i]*b(i)) - \sum_{i=1}^{i=M-1} (y[n-i]*a(i)) \qquad \text{(a)}$$

where :

- x is the discrete time input signal to the system represented by the filter
- y is the discrete time output signal of the system represented by the filter
- a is the set of coefficients applied to the input in the linear difference equation and represent the multiplication factors for delays. The equation suggests that there are N such coefficients.
- b is the set of coefficients applied to the output in the linear difference equation and represent the multiplication factors for delays. The equation suggests that there are M such coefficients.

The set of coefficients a and b are often referred to as the numerator and denominator coefficients, respectively. Another common way to refer to them is as the feedforward and feedback coefficients. This is due to the mathematical derivation that led to equation a. Refer to *Discrete-Time Signal Processing* by Oppenheim and Schafer for more information. Another frequent assumption is that a(0)=1.0. For example, let us assume M = 4, N = 4 and n = 2. Using the filtering equation produces:

$$a(0)*y(2) = b(0)*x(2) + b(1)*x(1) + b(2)* x(0) + b(3)* x(-1) - a(1)*y(1) - a(2)*y(0) - a(3)*y(-1) \qquad (b)$$

The filtering equation suggests that in order to compute the value of the output at n = 2, you not only need the coefficients that represent the filter that is producing this output sequence, but also the value of the current input and output, the values of the input and output one time unit ago, the values of the input and output two time units ago, and the values of the input and output three time units ago. You need some “history” about the previous outputs of the filter as well as the previous inputs to the filter. The amount of history (number of previous output and input samples) depends on the lengths of the arrays a and b (filter coefficients arrays).

At time n = 0, it is important to note that the filtering equation becomes:

$$a(0)*y(0) = b(0)*x(0) + b(1)*x(-1) + b(2)* x(-2) + b(3)* x(-3) - a(1)*y(-1) - a(2)*y(-2) - a(3)*y(-3) \qquad (c)$$

Thus, for the function filter to properly operate as of time n = 0, you need to supply some history about previous behavior. The filter function then updates the history as time goes on, keeping track of previous input values and corresponding outputs. This history, at time n = 0, is referred to as the initial conditions on the input and output of the filter.

Digital filters fall into two classes–Infinite Impulse Response filters (IIR filters) and Finite Impulse Response filters (FIR filters). Notice that IIR filters are represented by equation (a) while the FIR filters can be represented by the same equation provided all a's are zero except for a(0) as shown in the following equation:

$$a(0)*y(n) = \sum_{i=0}^{i=N} (x[n-i]*b(i)) \qquad (d)$$

The NI-DSP Analysis VIs have a set of VIs that implement IIR and FIR filters. Because all digital filters are approximations of their analog design counterparts, there are several techniques for designing a digital filter.

For the IIR filter design, the NI-DSP Analysis VIs have four approaches representing four different techniques of obtaining digital filter specifications (coefficients to equation (a))–Butterworth, Chebyshev, inverse Chebyshev, and elliptic techniques. With each design technique, you can obtain the coefficients for lowpass, highpass, bandpass, and bandstop filters from the respective NI-DSP Analysis VIs.

For the FIR filters, the NI-DSP Analysis VIs allow the design of a multiband FIR linear phase filter using the Parks-McClellan algorithm. The frequency response in each band has equal ripples that can be adjusted by a weighting factor. For more information, please refer to *Digital Filter Design* by Parks and Burrus, or "A Computer Program for Designing Optimum FIR Linear Phase Digital Filters," by McClellan, Parks, and Rabiner.

The IIR VIs generate filter coefficients that represent the a's and b's in the equations a and b. No filtering is actually performed. The IIR filter design coefficients all have a(0) = 1.0. You can use the general-purpose VI that accepts filter coefficients, initial conditions on the input and output sequences, and an input sequence to filter any of the filter specifications. This VI filters the input sequence and provides the final conditions on the output and input of the filter.

About Windowing

Almost every application requires you to use finite length signals. This requires that continuous signals be truncated, using a process called windowing.

The simplest window is a rectangular window. Because this window requires no special effort it is commonly referred to as the no window option. Remember, however, that a discrete signal and its spectrum is always affected by a window. Let x[n] be a digitized time-domain waveform that has a finite length of n. w[n] is a window sequence of n points. The windowed output is calculated as follows:

$$y[i] = x[i] * w[i] \tag{1}$$

If X, Y, and W are the spectra of x, y, and w, respectively, the time-domain multiplication in formula 1 is equivalent to the convolution shown as follows:

$$Y[k] = X[k] \Theta W[k]. \tag{2}$$

Convolving with the window spectrum always distorts the original signal spectrum in some way. A window spectrum consists of a big mainlobe and several sidelobes.

The mainlobe is the primary cause of lost frequency resolution. When two signal spectrum lines are too close to each other, they may fall in the width of the mainlobe, causing the output of the windowed signal spectrum to have only one spectrum line. Use a window with a narrower mainlobe to reduce the loss of frequency resolution. It has been shown that a rectangular window has the narrowest mainlobe, so that it provides the best frequency resolution.

The sidelobes of a window function affect frequency leakage. A signal spectrum line will leak into the adjacent spectrum if the sidelobes are large. Once again, the leakage results from the convolution process. Select a window with relatively smaller sidelobes to reduce spectral leakage. Unfortunately, a narrower mainlobe and smaller sidelobes are mutually exclusive. For this reason, selecting a window function is application dependent. An example of a windowed spectrum in the continuous case is shown in Figure 1-2.

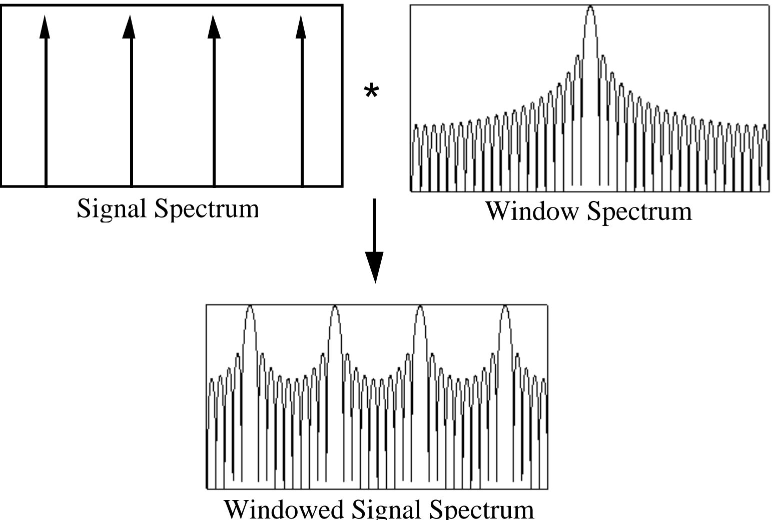


Figure 1-2. Spectral Leakage Demonstrated Using Convolution

The original signal spectrum is convolved with the window spectrum and the output is a smeared version of the original signal spectrum. In this example, you can still see four distinctive peaks from the original signal, but each peak is smeared and the frequency leakage effect is clear.

Window definitions used in National Instruments analysis libraries are designed in such a way that the window operations in the time domain are exactly equivalent to the operations of the same window in the frequency domain. To meet this requirement, the windows are not symmetrical in the time domain, that is:

$$w[0] \neq w[N-1] \quad (3)$$

where N is the window length. They are usually symmetrical in the frequency domain, however. For example, the Hamming window definition uses the formula:

$$w[i] = 0.54 - 0.46 \cos(2\pi i/N) \quad (4)$$

Other manufacturers may use a slightly different definition, such as:

$$w[i] = 0.54 - 0.46 \cos(2\pi i/N-1) \quad (5)$$

The difference is small if N is large.

Formula 4 is not symmetrical in the time domain, but it ensures that the time domain windowing is equivalent to the frequency domain windowing. If you want to have a perfectly symmetrical sequence in the time domain, you must write your own windowing function using formula 5.

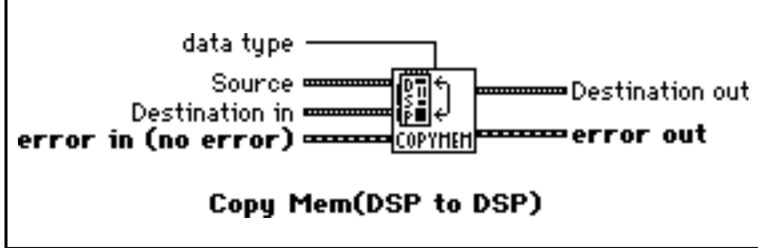
The choice of a window depends on the application. For most applications, the Hamming or Hanning windows deliver good performance.

Chapter 2
NI-DSP Analysis VI Reference

This chapter contains a brief explanation of each NI-DSP Analysis VI . The VIs are arranged alphabetically.

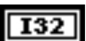
Copy Mem(DSP to DSP)

Copies a buffer of data from the **Source** buffer on the DSP board that is referred to by a DSP Handle Cluster to the **destination** buffer on the DSP board, which is referred to by another DSP Handle Cluster. **Source** and **destination** buffers should be on the same DSP board.



To copy data correctly from one DSP buffer to another DSP buffer, you must set the **data type** to the appropriate type to indicate what kind of data is on the DSP board. The VI has three data types–32-bit floating-point data, 16-bit short integer data, and 32-bit long integer data.

Source is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the data that you want to copy to the destination.



data type indicates the type of data that **Source** contains. The VI has three data types:

0: 32-bit floating-point data.
1: 16-bit short integer data.
2: 32-bit long integer data.

data type defaults to 32-bit floating-point data.

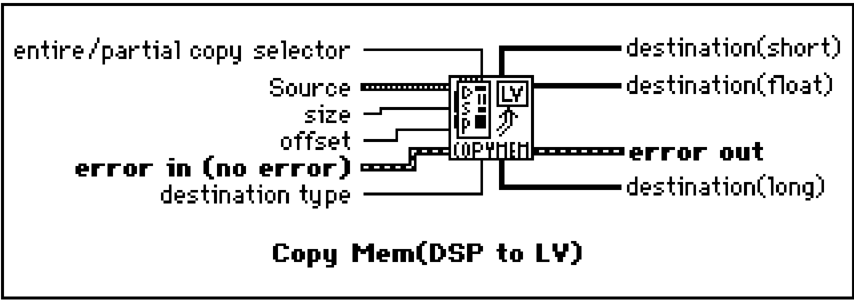
Destination in is a DSP Handle Cluster that indicates the destination memory buffer on the DSP board that will contain the data copied from source buffer.

Destination out is a DSP Handle Cluster that is identical to the **Destination in**, but with the source buffer data already copied to the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

Copy Mem(DSP to LV)



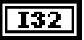
Copies an entire or partial buffer of data according to the **entire/partial copy selector** from the **Source** buffer on the DSP board that is referred to by a DSP Handle Cluster to the **destination** buffer in LabVIEW.

To copy data correctly from the DSP board to LabVIEW, you must indicate what type of data is stored in the **Source** buffer on the DSP board. You must set the **destination type** to the appropriate type and wire to the corresponding destination terminal. The VI has three destination types–float (32-bit), short (16-bit), and long (32-bit). To copy different types of data, you must wire to the appropriate destination terminal that corresponds to the **destination type** you choose. Remember, you must wire only one kind of destination terminal.

If you set the **entire/partial copy selector** to entire copy, all the data in the **Source** buffer will be copied back to the **destination** without considering the values of **offset** and **size**. Otherwise, only the number of **size** the data in the **Source** buffer, starting from the **offset**, will be copied back to the **destination**.

entire/partial copy selector has two types: entire copy and partial copy. It defaults to entire copy.
0: entire copy.
1: partial copy.

Source is a DSP Handle Cluster that indicates the memory buffer on the DSP board which contains the data that you want copy to the **destination** in LabVIEW.



destination type indicates the type of the destination data on the DSP board. It has three options:
0: 32-bit floating-point.
1: 16-bit short integer.
2: 32-bit long integer.

destination type defaults to 32-bit floating point.

offset indicates where to start to copy data from the **Source** buffer on the DSP board to the **destination** buffer in the LabVIEW. **offset** defaults to 0. This parameter is ignored when **entire/partial copy selector** is set to entire copy.

size indicates how much data to copy from the **Source** buffer on the DSP board to the **destination** buffer in LabVIEW. **size** defaults to 0. If **offset** plus **size** is greater than the number of elements in the **Source** buffer, the VI returns an error. This parameter is ignored when **entire/partial copy selector** is set to entire copy.

destination(float) is the terminal to which you should wire the destination buffer if the data you want to copy is 32-bit floating-point data array.

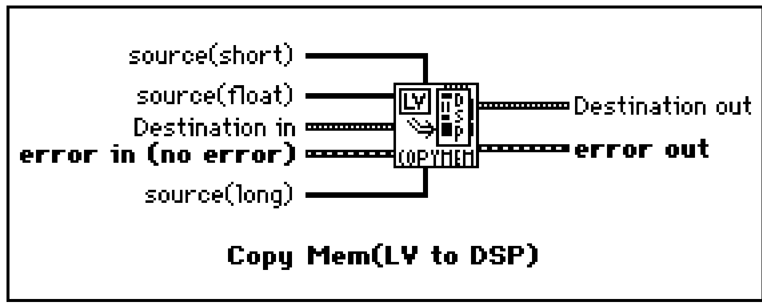
[I32] **destination(long)** is the terminal to which you should wire the destination buffer if the data you want to copy is 32-bit long data array.

[I16] **destination(short)** is the terminal to which you should wire the destination buffer if the data you want to copy is 16-bit short data array.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

Copy Mem(LV to DSP)



Copies a buffer of data from the **source** buffer in LabVIEW to the destination buffer on the DSP board, which is referred to by a DSP Handle Cluster.

The source buffer can contain one of three kinds of data–float (32-bit), short (16-bit), and long (32-bit). To copy different types of data, you must wire to the appropriate source terminal. You must wire only one kind of source terminal. The destination buffer must be large enough to contain all of the data from the **source** buffer.

source(float) is the terminal to which you should wire the source buffer if the data you want to copy is 32-bit floating point data array.

source(long) is the terminal to which you should wire the source buffer if the data you want to copy is 32-bit long data array.

source(short) is the terminal to which you should wire the source buffer if the data you want to copy is 16-bit short data array.

Destination in is a DSP Handle Cluster that indicates the destination memory buffer on the DSP board that will contain the data copied from the LabVIEW source buffer.

Destination out is a DSP Handle Cluster that is identical to **Destination in,** but with the source buffer data already copied to the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

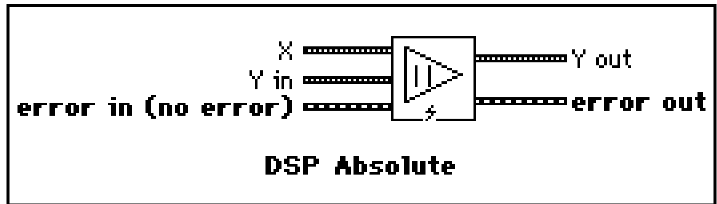
DSP Absolute

Find the absolute value of input array **X**. The ith element of the output array **Y** is obtained using the following formula:

$Y(i) = |X(i)|$.

for $i = 0, 1, 2, \ldots, n-1$

where n is the number of elements in **X**.

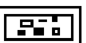


X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the absolute results of X(i).

Y out is a DSP Handle Cluster that is identical to **Y in**, but with the absolute results already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.



error out contains the error information for this call.

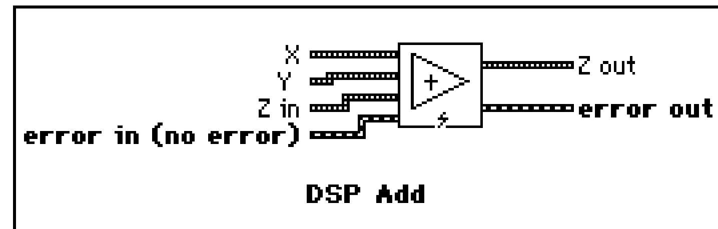
The operation can be performed in place; that is, the input and output arrays can be the same DSP Handle Cluster.

DSP Add

Add array **X** to array **Y.** The ith element of the output array **Z** is obtained using the following formula:

$$Z(i) = X(i) + Y(i).$$

for $i = 0, 1, 2, \ldots, n-1$



where n is the smaller number of elements in **X** and **Y**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Y is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array Y.

Z in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of X(i) + Y(i).

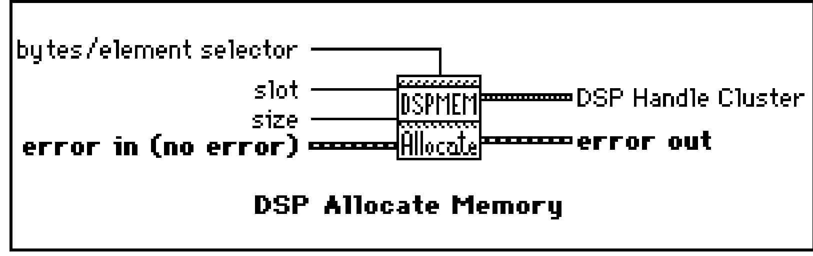
Z out is a DSP Handle Cluster that is identical to **Z in**, but with the added results already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation can be performed in place; that is, the input and output arrays can be the same DSP Handle Cluster.

DSP Allocate Memory



Allocates a block of memory buffer on the DSP board specified by **slot** and returns a DSP Handle Cluster that contains the coded DSP board memory and the number of elements in this buffer.

The number of bytes allocated for this buffer depends on **size** and **bytes/element selector**. If **bytes/element selector** selects 4 bytes, then the number of bytes = **size***4. If **bytes/element selector** selects 2 bytes, then the number of bytes = **size***2.

The allocation routines on the board assure alignment to the nearest 4-byte boundary ≥ number of bytes to guarantee memory alignment on 4-byte addresses. For example, if you request to allocate a buffer of size 1,022 or 1,024 bytes, the allocation routines allocate 1,024 bytes in both cases.

slot is the board ID number. **slot** defaults to 3.

size is the number of elements to allocate for this buffer. **size** defaults to 0.

bytes/element selector specifies the number of bytes per element. It has two options:
0: 4 bytes.
1: 2 bytes.

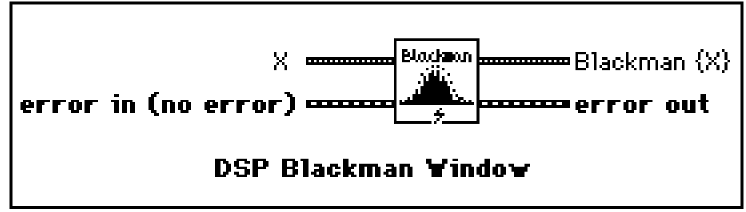
bytes/element selector defaults to 4 bytes.

DSP Handle Cluster indicates the allocated memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Blackman Window



Applies a Blackman window to the input sequence **X**. If Y represents the output sequence **Blackman{X}**, the elements of Y are obtained from the following formula:

$$y_i = x_i [0.42 - 0.50 \cos(w) + 0.08 \cos(2w)]$$

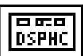
for $i = 0, 1, 2, \ldots, n-1$,

$$w = \frac{2\pi i}{n},$$

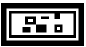
where n is the number of elements in **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Blackman {X}.**



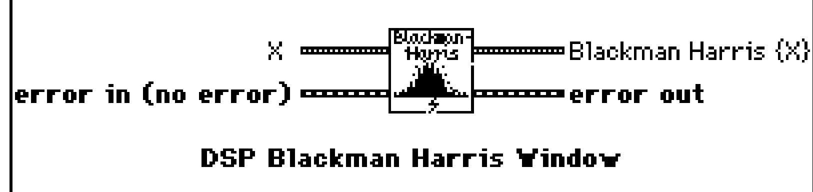
Blackman {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Blackman {X}** already stored in the memory buffer on the DSP board.



error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Blackman Harris Window



Applies a Blackman Harris window to the input sequence **X**. If Y represents the output sequence **Blackman Harris{X}**, the elements of Y are obtained using the following formula:

$$y_i = x_i [0.42323 - 0.49755 \cos(w)$$

$$+0.07922 \cos(2w)]$$

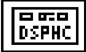
for $i = 0, 1, 2, \ldots, n-1$

$$w = \frac{2\pi i}{n} \ ,$$

where n is the number of elements in **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Blackman Harris {X}.**



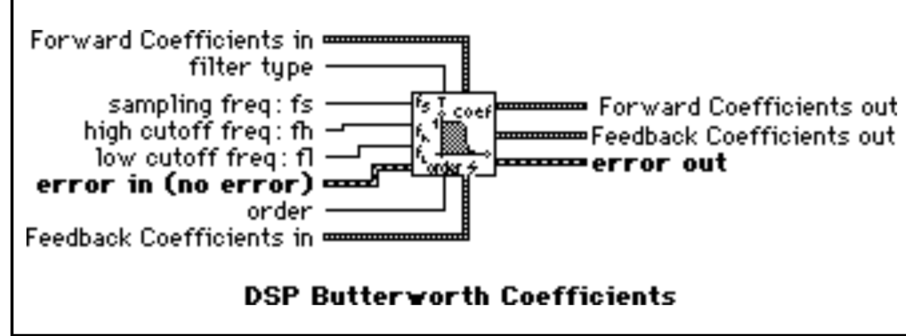
Blackman Harris {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Blackman Harris {X}** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Butterworth Coefficients

Generates the set of filter coefficients to implement an IIR filter as specified by the Butterworth filter model. You can then pass these coefficients to the DSP IIR Filter VI to filter a sequence of data.



filter type specifies the passband of the filter. It has four options:
0: lowpass.
1: highpass.
2: bandpass.
3: bandstop.

filter type defaults to lowpass.

sampling freq : fs is the sampling frequency and must be greater than 0. If it is less than or equal to zero, the VI returns an error. **sampling freq : fs** defaults to 1.0.

high cutoff freq: fh is the high cutoff frequency. The VI ignores this parameter when **filter type** is lowpass. **high cutoff freq: fh** defaults to 0.45.

low cutoff freq: fl is the low cutoff frequency. The VI ignores this parameter when **filter type** is highpass. **low cutoff freq: fl** defaults to 0.125.

Note: fh and fl must observe the Nyquist criterion: $0 \leq fl \leq fh \leq fs/2$.

Forward Coefficients in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the forward coefficients.

Feedback Coefficients in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the feedback coefficients.

Forward Coefficients out is a DSP Handle Cluster that is identical to the **Forward Coefficients in**, but with the forward coefficients already stored in the memory buffer on the DSP board.

Feedback coefficients out is aDSP Handle Cluster that is identical to the **Feedback Coefficients in**, but with the feedback coefficients already stored in the memory buffer on the DSP board.

order must be greater than zero. If **order** is less than or equal to zero, the VI returns an error. **order** defaults to 2.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

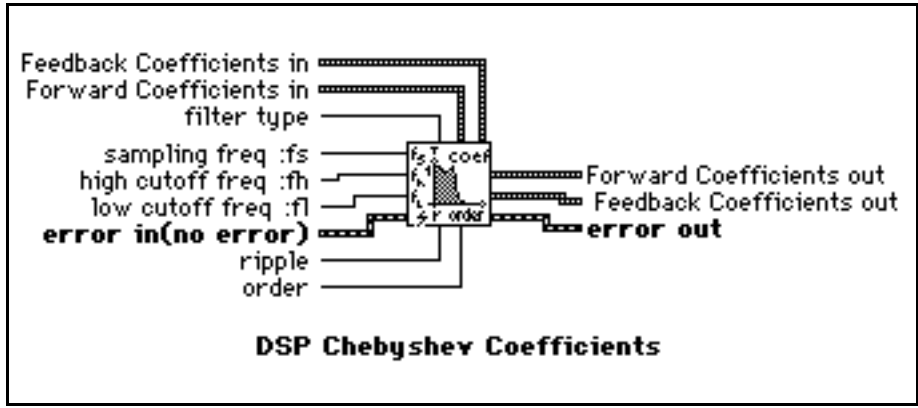
error out contains the error information for this call.

Parameter Discussion

The arrays **Forward Coefficients in/out** and **Feedback Coefficients in/out** must have a size of at least (**order** + 1) for lowpass and highpass filters. The arrays **Forward Coefficients in/out** and **Feedback Coefficients in/out** must have a size of at least (2***order** + 1) for bandpass and bandstop filters.

DSP Chebyshev Coefficients

Generates the set of filter coefficients to implement an IIR filter as specified by the Chebyshev filter model. You can then pass coefficients to the DSP IIR Filter VI to filter a sequence of data.



filter type specifies the passband of the filter. It has four options:

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

filter type defaults to lowpass.

sampling freq : fs is the sampling frequency and must be greater than 0. If it is less than or equal to zero, the VI returns an error. **sampling Freq : fs** defaults to 1.0.

high cutoff freq: fh is the high cutoff frequency. The VI ignores this parameter when **filter type** is lowpass. **high cutoff freq: fh** defaults to 0.45.

low cutoff freq: fl is the low cutoff frequency. The VI ignores this parameter when **filter type** is highpass. **low cutoff freq: fl** defaults to 0.125.

Note: fh and fl must observe the Nyquist criterion: $0 \le fl \le fh \le fs/2$.

ripple must be greater than 0. If **ripple** is less than or equal to zero, the VI returns an error. **ripple** defaults to 0.1.

order must be greater than zero. If **order** is less than or equal to zero, the VI returns an error. **order** defaults to 2.

Forward Coefficients in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the forward coefficients.

Feedback Coefficients in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the feedback coefficients.

Forward Coefficients out is a DSP Handle Cluster that is identical to the **Forward Coefficients in**, but with the forward coefficients already stored in the memory buffer on the DSP board.

Feedback Coefficients out is a DSP Handle Cluster that is identical to the **Feedback Coefficients in**, but with the feedback coefficients already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

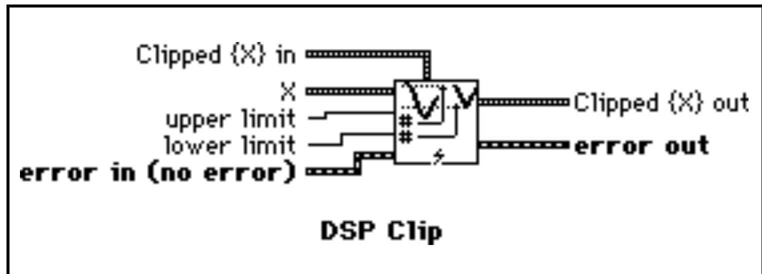
error out contains the error information for this call.

Parameter Discussion

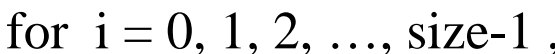
The arrays **Forward Coefficients in/out** and **Feedback Coefficients in/out** have to have a size of at least (**order** + 1) for lowpass and highpass filters.The arrays **Forward Coefficients in/out** and **Feedback Coefficients in/out** have to have a size of at least (2***order** + 1) for bandpass and bandstop filters.

DSP Clip

Clips the input array values. The range of the resulting output array is [**lower**:**upper**]. Let Y represent the output array. The ith element of the resulting array is obtained by using the following formulas:



$$\mathbf{Y(i)} = \begin{cases} \mathbf{upper} \text{ if } \mathbf{X}(i) > \mathbf{upper} \\ \mathbf{lower} \text{ if } \mathbf{X}(i) < \mathbf{lower} \\ \mathbf{X}(i) \text{ otherwise} \end{cases}$$



for $i = 0, 1, 2, \ldots, \text{size-1}$,

where n is the number of elements in **X**.

The operation can be performed in place; that is, the input and output arrays can be the same DSP Handle Cluster.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

upper limit. **upper limit** ≥ **lower limit. upper limit** defaults to 1.0**.**

lower limit. **lower limit** defaults to 0.0**.**

Clipped{X} in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output array of **Clipped{X}**.

Clipped{X} out is a DSP Handle Cluster that is identical to **Clipped {X} in**, but with the **Clipped{X}** already stored in the memory buffer on the DSP board.

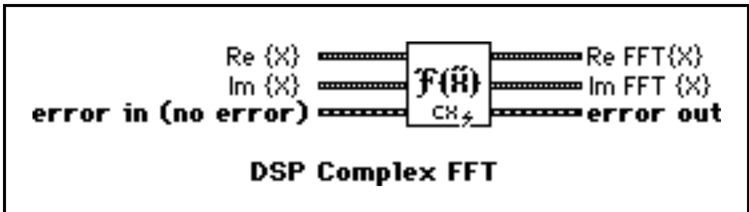
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Complex FFT

Computes the Fast Fourier transform of the complex input sequence X. If Y represents the complex output sequence, then:

$$Y = \boldsymbol{F}\{X\}.$$



Re{X} is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the real part of the input signal array X.

Im{X} is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the imaginary part of input signal array X.

Notes: The number of elements for the input array must be a power of 2.

If the size of **Re{X}** is different from the size of **Im{X}**, the VI uses the smaller number as the size of the Complex FFT.

The operation is performed in place and the input array **Re{X}** and **Im{X}** is overwritten by the **Re FFT{X}** and **Im FFT{X}.**

The largest complex FFT that can be computed depends upon the amount of memory on your board.

Re FFT{X} is a DSP Handle Cluster that is identical to **Re{X}**, but with the real part of FFT{X}already stored in the memory buffer on the DSP board.

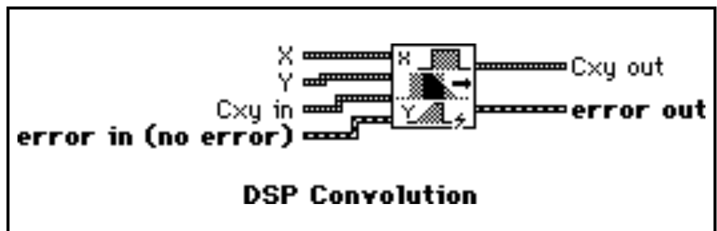
Im FFT{X} is a DSP Handle Cluster that is identical to **Im{X}**, but with the imaginary part of FFT{X}already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Convolution

Computes the convolution of the input sequences **X** and **Y**. The convolution Cxy(t), of the signals x(t) and y(t), is defined as follows:



$$\mathbf{C}xy(t) = x(t) * y(t) = \int_{-\infty}^{\infty} x(\tau)\, y(t-\tau)\, d\tau,$$

where the symbol $*$ denotes convolution.

For the discrete implementation of the convolution, let Cxy represent the output sequence **X * Y**, n be the number of elements in the input sequence **X**, and m be the number of elements in the input sequence **Y**. Assuming that indexed elements of **X** and **Y** that lie outside their range are zero,

$$x_i = 0, \qquad i < 0 \quad \text{or} \quad i \geq n$$

and

$$y_j = 0, \qquad j < 0 \quad \text{or} \quad j \geq m,$$

then you obtain the elements of Cxy using:

$$Cxy_i = \sum_{k=0}^{n-1} x_k\, y_{i-k} \qquad \text{for } i = 0, 1, 2, \ldots, \text{size-1},$$

$$\text{size} = n + m - 1,$$

where size is the total number of elements in the output sequence **X * Y**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Y is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array Y.

Cxy in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the convolution of X with Y.

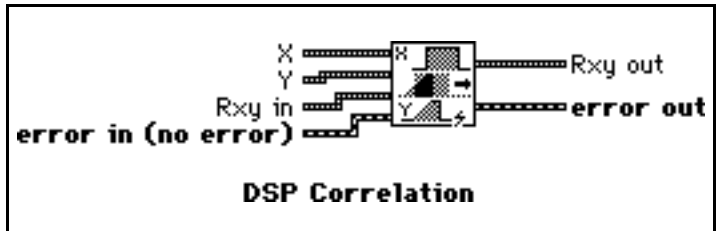
Note: The size of **Cxy in** must be (n+m-1) elements long. n is the size of **X**, m is the size of **Y**. You cannot perform the operation in place.

Cxy out is a DSP Handle Cluster that is identical to the **Cxy in** but with the convolution of X and Y already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Correlation



Computes the cross correlation of the input sequences **X** and **Y**. The cross correlation $R_{xy}(t)$ of the signals x(t) and y(t) is defined as follows:

$$R_{xy}(t) = x(t) \otimes y(t) = \int_{-\infty}^{\infty} x(\tau)\, y(t+\tau) \quad d\tau ,$$

where the symbol ⊗ denotes correlation.

For the discrete implementation of the correlation, let R_{xy} represent the output sequence X ⊗ Y, n be the number of elements in the input sequence X, and m be the number of elements in the input sequence Y. You then obtain the elements of R_{xy} using the following formula:

$$R_{xy\,i} = \sum_{k=0}^{n-1} x[k]\, y[k+m-1] \qquad \text{for } i = 0, 1, 2, \ldots, m+n-1 ,$$

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Y is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array Y.

Rxy in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the correlation results.

Note: The size of **Rxy in** must be at least (n+m-1) elements long. n is the size of **X**, m is the size of **Y**.

Rxy out is a DSP Handle Cluster that is identical to the **Rxy in**, but with the correlation of X and Y already stored in the memory buffer on the DSP board.

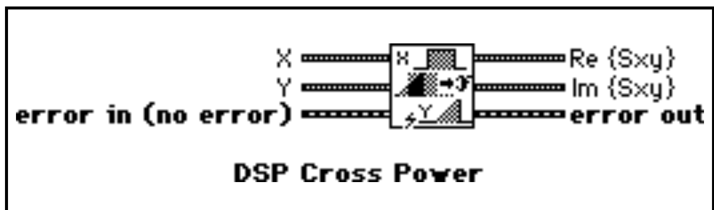
Note: If **X** and **Y** are the same array, an auto correlation is performed, otherwise, a cross correlation is performed. You cannot perform the operation in place.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Cross Power

Computes the cross power spectrum of the input sequences **X** and **Y**. The cross power, $S_{xy}(f)$, of the signals x(t) and y(t) is defined as follows:



$$S_{xy}(f) = X^*(f)Y(f)$$

where $X^*(f)$ is the complex conjugate of X(f),
$X(f) = \boldsymbol{F}\{x(t)\}$, and
$Y(f) = \boldsymbol{F}\{y(t)\}$.

This VI uses the FFT routine to compute the cross power spectrum, that is given by the following formula:

$$S_{xy} = \frac{1}{n^2} \boldsymbol{F}^*\{\mathbf{X}\} \boldsymbol{F}\{\mathbf{Y}\},$$

$$n = 2^k \qquad \text{for } k = 1, 2, 3, \ldots, 23,$$

where S_{xy} represents the complex output sequence **Sxy**, and
n is the number of samples that can accommodate both input sequences **X** and **Y**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Y is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array Y.

Notes: If the size of **X** is different than **Y**, the VI uses the smaller number as the size of input arrays X and Y. The number of elements for the input arrays must be a power of two.

The operation is performed in place and the input arrays **X** and **Y** are overwritten by the outputs **Re{Sxy}** and **Im{Sxy}**.

The largest Cross Power Spectrum that can be computed depends upon the amount of memory in your DSP board.

This VI allocates a temporary workspace on the DSP board equal to the size of the input signal array.

Re{Sxy} is a DSP Handle Cluster that is identical to **X**, but with the real part of **Sxy** already stored in the memory buffer on the DSP board.

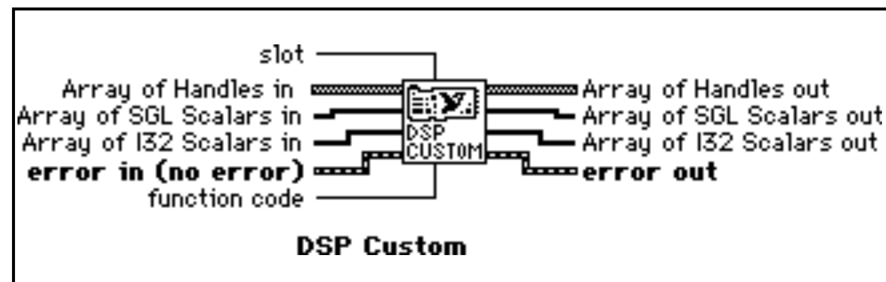
Im{Sxy} is a DSP Handle Cluster that is identical to **Y**, but with the imaginary part of **Sxy** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Custom

Use this VI as the interface to call your own custom functions written on the DSP board from LabVIEW. For more details about how to use this VI, refer to Part 4, *NI-DSP Interface Utilities*, of this manual.



[I16] **slot** is the board ID number. **slot** defaults to 3.

[I32] **function code** indicates which function on the DSP board LabVIEW is calling. **function code** defaults to 0.

[DSPHC] **Array of Handles in** is an array of DSP Handle Clusters that hold all of the references to the input and output data buffers used in your custom functions on the DSP board.

[SGL] **Array of SGL Scalars in** is an array of 32-bit floating-point scalars that hold all of the input 32-bit floating-point scalars used in your custom functions on the DSP board.

[I32] **Array of I32 Scalars in** is an array of 32-bit long integer scalars that hold all of the input 32-bit long integer scalars used in your custom functions on the DSP board.

[DSPHC] **Array of Handles out** is an array of DSP Handle Clusters that is identical to **Array of Handles in**, but with the output results already stored in the memory buffers on the DSP board.

[SGL] **Array of SGL Scalars out** is an array of 32-bit floating-point scalars that is identical to **Array of SGL Scalars in**, but with the output results already stored in the array.

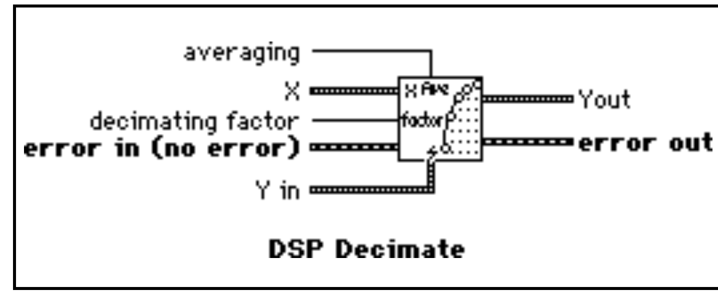
[I32] **Array of I32 Scalars out** is an array of 32-bit long integer scalars that is identical to **Array of I32 Scalars in**, but with the output results already stored in the array.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Decimate

Decimates the input sequence **X** by the **decimating factor** and the **averaging** control. If Y represents the output sequence **Decimated Array**, the elements of the sequence Y are obtained using:



$$Y_i = \begin{cases} X_{im} & \text{if ave = no averaging} \\ \frac{1}{m}\sum_{k=0}^{m-1} X_{i(m+k)} & \text{if ave = averaging} \end{cases}$$

for i = 0, 1, 2, …, size -1

$$\text{size} = \text{trunc}\left(\frac{n}{m}\right),$$

where n is the number of elements in **X**,
m is the **decimating factor**,
ave is the **averaging** option, and
size is the number of elements in the output sequence **Y**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the decimated output array.

Y out is a DSP Handle Cluster that is identical to **Y in**, but with the results of the decimated array already stored in the memory buffer on the DSP board.

decimating factor must be greater than zero:

0 < **decimating factor** ≤ n.

If **decimating factor** is greater than the number of samples in **X** or less than or equal to zero, the VI returns an error. **decimating factor** defaults to 1.

averaging has two options:
0: averaging.
1: no averaging.

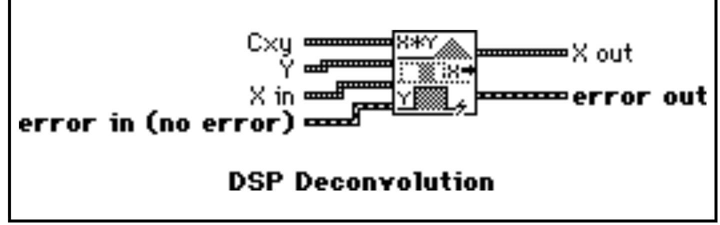
averaging defaults to no averaging.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation can be performed in place; that is, the input X and the output Y can be the same DSP Handle Cluster.

DSP Deconvolution



Computes the deconvolution of the input sequences **X** and **Y**. The convolution operation can be realized using Fourier identities because

$$x(t) * y(t) \Leftrightarrow X(f)\,Y(f)$$

is a Fourier transform pair, where the symbol $*$ denotes convolution, and the deconvolution is the inverse of the convolution operation. If h(t) is the signal resulting from the deconvolution of the signals x(t) and y(t), the VI obtains h(t) using the equation

$$h(t) = \boldsymbol{F}^{-1}\left\{\frac{X(f)}{Y(f)}\right\},$$

where X(f) is the Fourier transform of x(t), and
Y(f) is the Fourier transform of y(t).

The VI performs the discrete implementation of the deconvolution using the following steps.

1. Compute the Fourier transform of the input sequence **X * Y**.
2. Compute the Fourier transform of the input sequence **Y**.
3. Divide the Fourier transform of **X * Y** by the Fourier transform of **Y**. Call the new sequence H.
4. Compute the inverse Fourier transform of H to obtain the deconvoluted sequence **X**.

Cxy is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array Cxy. The number of elements in **Cxy** must be greater than or equal to the number of elements in **Y**.

Y is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array Y.

X in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the deconvolution results.

Note: The size of **X in** must be n elements long, although only (n-m+1) elements are valid. n is the size of **Cxy**, m is the size of **Y**.

X out is a DSP Handle Cluster that is identical to **X in**, but with the deconvolution results already stored in the memory buffer on the DSP board.

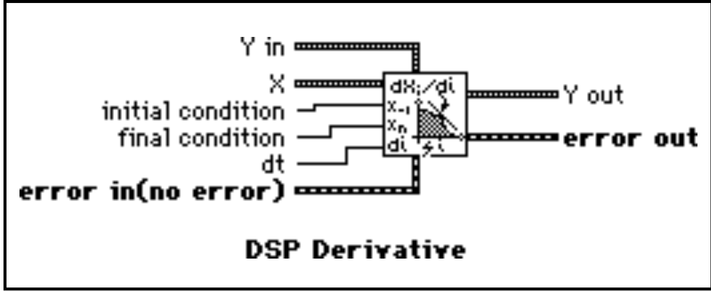
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

Note: The deconvolution operation is a numerically unstable operation, and it is not always possible to solve the system numerically. Computing the deconvolution via FFTs is perhaps the most stable generic algorithm that does not require sophisticated DSP techniques. However, it is not free of errors (for example, when there are zeros in the Fourier transform of the input sequence **Y**).

DSP Derivative

Performs a discrete differentiation of the sampled signal **X**. The differentiation f(t) of a function F(t) is defined as follows:



$$f(t) = \frac{d}{dt} F(t).$$

Let Y represent the sampled output sequence **d/dt X**. The discrete implementation is given by the following formula:

$$y_i = \frac{1}{2\mathbf{dt}} (x_{i+1} - x_{i-1}) \qquad \text{for } i = 0, 1, 2, \ldots, n\text{-}1.$$

where n is the number of samples in **x(t)**
x_{-1} is specified by **initial condition** when i = 0, and
x_n is specified by **final condition** when i = n-1.

initial condition and **final condition** minimize the error at the boundaries.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

initial condition defaults to 0.0.

final condition defaults to 0.0.

dt is the sampling interval and must be greater than zero. If **dt** is less than or equal to zero, the VI returns an error. **dt** defaults to 1.0.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of the differentiation of X.

Y out is a DSP Handle Cluster that is identical to **Y in**, but with the results of differentiation already stored in the memory buffer on the DSP board.

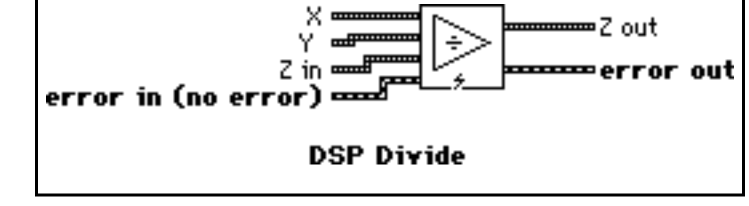
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation can be performed in place; that is, the input X and the output Y can be the same DSP Handle Cluster.

DSP Divide

Divide array **X** by array **Y.** The ith element of the output array **Z** is obtained using the following formula:



Z(i) = X(i) / Y(i).

for i = 0, 1, 2, …, n-1.

where n is the smaller number of elements in **X** and **Y**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Y is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array Y.

Z in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of X(i) / Y(i).

Z out is a DSP Handle Cluster that is identical to **Z in**, but with the divided results already stored in the memory buffer on the DSP board.

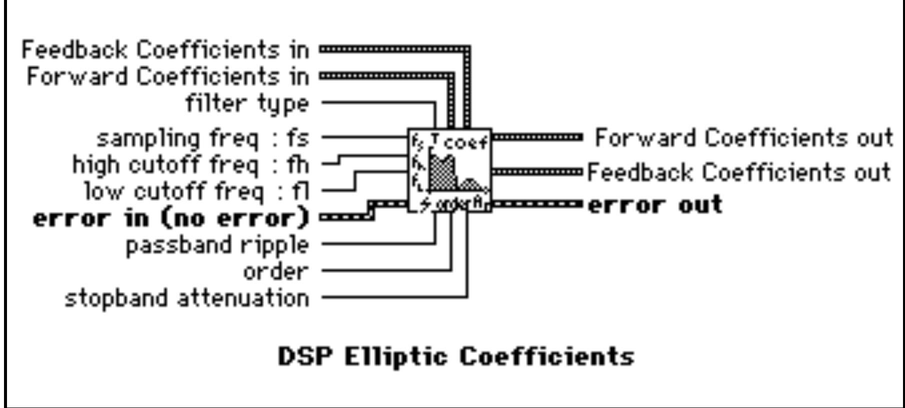
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation can be performed in place; that is, the input and the output arrays can be the same DSP Handle Cluster.

DSP Elliptic Coefficients

Generates the set of filter coefficients to implement a digital elliptic IIR filter. You can then pass these coefficients to the DSP IIR Filter VI.



filter type specifies the passband of the filter. **filter type** has four options:
- 0: lowpass.
- 1: highpass.
- 2: bandpass.
- 3: bandstop.

filter type defaults to lowpass.

sampling freq : fs is the sampling frequency and must be greater than 0. If it is less than or equal to zero, the VI returns an error. **sampling freq : fs** defaults to 1.0.

high cutoff freq: fh is the high cutoff frequency. The VI ignores this parameter when **filter type** is lowpass. **high cutoff freq: fh** defaults to 0.45.

low cutoff freq: fl is the low cutoff frequency. The VI ignores this parameter when **filter type** is highpass. **high cutoff freq: fh** defaults to 0.125.

Note: fh and fl must observe the Nyquist criterion: $0 \le fl \le fh \le fs/2$.

passband ripple must be greater than 0, and you must express it in decibels. If **passband ripple** is less than or equal to zero, the VI returns an error. **passband ripple** defaults to 1.0.

order must be greater than zero. If **order** is less than or equal to zero, the VI returns an error. **order** defaults to 2.

stopband attenuation must be greater than 0, and you must express it in decibels. If **stopband attenuation** is less than or equal to zero, the VI returns an error. **stopband attenuation** defaults to 60.0.

Forward Coefficients in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the forward coefficients.

Feedback Coefficients in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the feedback coefficients.

Forward Coefficients out is a DSP Handle Cluster that is identical to **Forward Coefficients in**, but with the forward coefficients already stored in the memory buffer on the DSP board.

Feedback Coefficients out is a DSP Handle Cluster that is identical to **Feedback Coefficients in**, but with the feedback coefficients already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

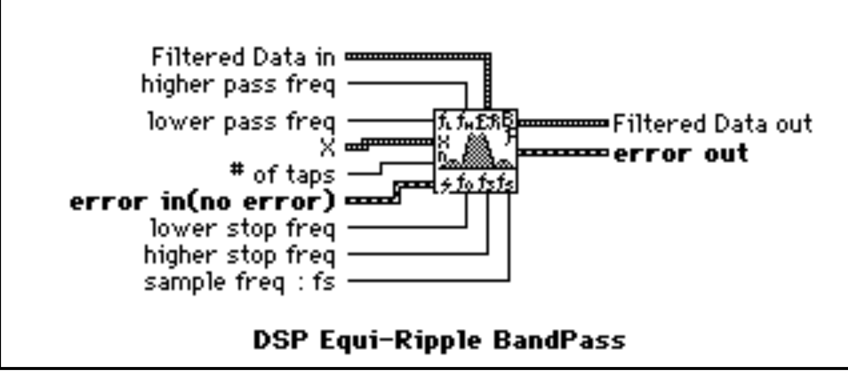
error out contains the error information for this call.

Parameter Discussion

The arrays **Forward Coefficients in/out** and **Feedback Coefficients in/out** must have a size of at least (**order** + 1) for lowpass and highpass filters. The arrays **Forward Coefficients in/out** and **Feedback Coefficients in/out** must have a size of at least (2***order** + 1) for bandpass and bandstop filters.

DSP Equi-Ripple BandPass

Generates a bandpass FIR filter with equi-ripple characteristics using the Parks-McClellan algorithm and the number of taps, lower stop frequency, higher stop frequency, lower pass frequency, higher pass frequency, and sampling frequency. The VI then filters the input sequence **X** to obtain the bandpass filtered linear-phase sequence **Filtered Data**.



The first stopband of the filter region goes from zero (DC) to the lower stop frequency. The passband region goes from the lower pass frequency to the higher pass frequency, and the second stopband region goes from the higher stop frequency to the Nyquist frequency.

SGL **higher pass freq** must be greater than **lower pass freq** frequency. If **higher pass freq** is less than or equal to **lower pass freq**, the VI returns an error. **higher pass freq** defaults to 0.3.

SGL **lower pass freq** must be greater than the **lower stop freq**. If **lower pass freq** is less than or equal to **lower stop freq**, the VI returns an error. **lower pass freq** defaults to 0.2.

DSPHC **X** is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input data array.

I32 **# of taps** must be greater than 0. If the number of taps is less than or equal to zero, the VI returns an error. **# of taps** defaults to 32.

SGL **lower stop freq** must be greater than zero. If **lower stop freq** is less than or equal to zero, the VI returns an error. **lower stop freq** defaults to 0.1.

SGL **higher stop freq** must be greater than **higher pass freq** and must observe the Nyquist criterion:

$$0 < f_0 < f_1 < f_2 < f_3 \leq 0.5f_s,$$

where f_0 is **lower stop freq**, f_1 is **lower pass freq**, f_2 is **higher pass freq**, f_3 is the **higher stop freq**, and f_s is the sampling frequency. If any of these conditions is violated, the VI returns an error. **higher stop freq** defaults to 0.4.

SGL **sampling freq: fs** defaults to 1.0.

DSPHC **Filtered Data in** is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the filtered data output.

DSPHC **Filtered Data out** is a DSP Handle Cluster that is identical to **Filtered Data in**, but with the filtered data already stored in the memory buffer on the DSP board. Because the VI filters via convolution, the number of elements, k, in **Filtered Data** is as follows:

$$k = n + m - 1,$$

where n is the number of elements in **X**, and m is the number of taps.

A delay is also associated with the output sequence

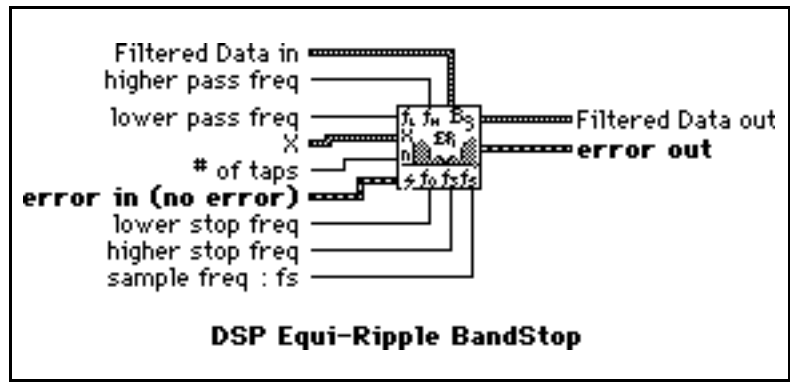
$$\text{delay} = \frac{m-1}{2}.$$

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation cannot be performed in place; that is, the input **X** and the output **Filtered Data** cannot be the same DSP Handle Cluster.

DSP Equi-Ripple BandStop



Generates a bandstop FIR digital filter with equi-ripple characteristics using the Parks-McClellan algorithm and number of taps, lower pass frequency, higher pass frequency, lower stop frequency, higher stop frequency, and sampling frequency. The VI then filters the input sequence **X** to obtain the bandstop filtered linear-phase sequence **Filtered Data**.

The first passband region of the filter goes from zero (DC) to the lower pass frequency. The stopband region goes from the lower stop frequency to the higher stop frequency, and the second passband region goes from the higher pass frequency to the Nyquist frequency.

higher pass freq must be greater than **higher stop freq** and observe the Nyquist criterion:

$$0 < f_0 < f_1 < f_2 < f_3 \leq 0.5f_s,$$

where f_0 is the **lower pass freq**, f_1 is the **lower stop freq**, f_2 is the **higher stop freq**, f_3 is the **higher pass freq**, and f_s is the **sampling freq**. If any of these conditions is violated, the VI returns an error. **higher pass freq** defaults to 0.4.

lower pass freq must be greater than zero. If **lower pass freq** is less than or equal to 0, the VI returns an error. **lower pass freq** defaults to 0.1.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input data array.

# of taps must be odd and must be greater than 0. If the number of taps is an even number or is less than or equal to zero, the VI returns an error. **# of taps** defaults to 31.

lower stop freq must be greater than **lower pass freq**. If **lower stop freq** is less than or equal to **lower pass freq**, the VI returns an error. **lower stop freq** defaults to 0.2.

higher stop freq must be greater than **lower stop freq**. If **higher stop freq** is less than or equal to **lower stop freq**, the VI returns an error. **higher stop freq** defaults to 0.3.

sampling freq: fs defaults to 1.0.

Filtered Data in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the filtered data output.

Filtered Data out is a DSP Handle Cluster that is identical to **Filtered Data in**, but with the filtered data already stored in the memory buffer on the DSP board. Because the VI filters via convolution, the number of elements, k, in **Filtered Data** is as follows:

$$k = n + m - 1,$$

where n is the number of elements in **X**, and m is the number of taps.

A delay is also associated with the output sequence:

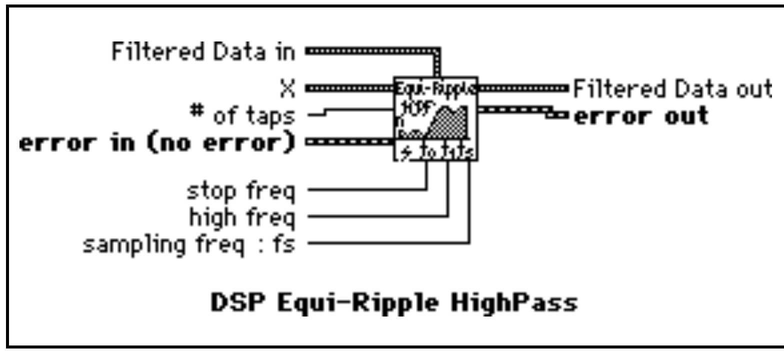
$$\text{delay} = \frac{m-1}{2}.$$

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation cannot be performed in place; that is, the input X and the output Filtered Data cannot be the same DSP Handle Cluster.

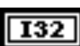
DSP Equi-Ripple HighPass



Generates a highpass FIR filter with equi-ripple characteristics using the Parks-McClellan algorithm and the number of taps, high frequency, stop frequency, and sampling frequency. The VI then filters the input sequence **X** to obtain the highpass filtered linear-phase sequence **Filtered Data**.

The stopband of the filter goes from zero (DC) to the stop frequency. The transition band goes from the stop frequency to the high frequency, and the passband goes from the high frequency to the Nyquist frequency.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input data array.



# of taps must be greater than 0 and must be an odd number. If the number of taps is an even number or is less than or equal to zero, the VI returns an error. **# of taps** defaults to 31.

stop freq must be greater than zero. If **stop freq** is less than or equal to 0, the VI returns an error. **stop freq** defaults to 0.2.

high freq must be greater than **stop freq** and observe the Nyquist criterion:

$$0 < f_0 < f_1 \leq 0.5f_s,$$

where f_0 is the **stop freq**, f_1 is the **high freq**, and f_s is the sampling frequency. If any of these conditions is violated, the VI returns an error. **high freq** defaults to 0.3.

sampling freq: fs defaults to 1.0.

Filtered Data in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the filtered data output.

Filtered Data out is a DSP Handle Cluster that is identical to **Filtered Data in**, but with the filtered data already stored in the memory buffer on the DSP board. Because the VI filters via convolution, the number of elements, k, in **Filtered Data** is:

$$k = n + m - 1,$$

where n is the number of elements in **X**, and m is the number of taps.

A delay is also associated with the output sequence:

$$\text{delay} = \frac{m-1}{2}.$$

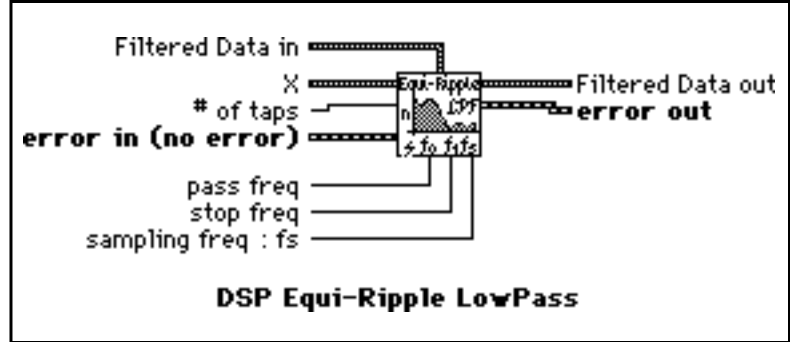
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation cannot be performed in place; that is, the input **X** and the output **Filtered Data out** cannot be the same DSP Handle Cluster.

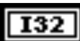
DSP Equi-Ripple LowPass

Generates a lowpass FIR filter with equi-ripple characteristics using the Parks-McClellan algorithm and the number of taps, pass frequency, stop frequency, and sampling frequency. The VI then filters the input sequence **X** to obtain the lowpass filtered linear-phase sequence **Filtered Data**.



The passband of the filter goes from zero (DC) to **pass freq**. The transition band goes from **pass freq** to **stop freq**, and the stopband goes from **stop freq** to the Nyquist frequency.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input data array.



# of Taps must be greater than 0. If the number of taps is less than or equal to zero, the VI returns an error. **# of Taps** defaults to 31.

pass freq must be greater than 0. If **pass freq** is less than or equal to zero, the VI returns an error. **pass freq** defaults to 0.2.

stop freq must be greater than the **pass freq** and observe the Nyquist criterion:

$$0 < f_0 < f_1 \leq 0.5f_s,$$

where f_0 is the **pass freq**, f_1 is the **stop freq**, and f_s is the sampling frequency. If any of these conditions is violated, the VI returns an error. **stop freq** defaults to 0.3.

sampling freq: fs defaults to 1.0.

Filtered Data in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the filtered data output.

Filtered Data out is a DSP Handle Cluster that is identical to **Filtered Data in**, but with the filtered data already stored in the memory buffer on the DSP board. Because the VI filters via convolution, the number of elements, k, in **Filtered Data out** is:

$$k = n + m - 1,$$

where n is the number of elements in **X**, and
m is the number of taps.

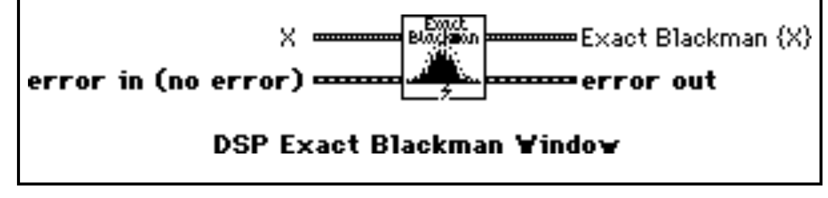
A delay equal to $\frac{m-1}{2}$ is also associated with the output sequence.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation cannot be performed in place; that is, the input **X** and the output **Filtered Data** cannot be the same DSP Handle Cluster.

DSP Exact Blackman Window



Applies an Exact Blackman window to the input sequence **X**. If Y represents the output sequence **Exact Blackman{X}**, the elements of Y are obtained using the formula:

$y_i = x_i$ [0.42659071

- 0.49656062 cos(w)

+ 0.07684867 cos(2w)]

for $i = 0, 1, 2, \ldots, n-1$

$$w = \frac{2\pi i}{n} ,$$

where n is the number of elements in **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Exact Blackman {X}.**

Exact Blackman {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Exact Blackman {X}** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Exponential Window

Applies an exponential window to the input sequence **X**. If Y represents the output sequence **Exponential{X}**, the elements of Y are obtained using the formula:

$$y_i = x_i \exp(a * i)$$

for $i = 0, 1, 2, \ldots, n-1$,

$$a = \frac{\ln |f|}{n - 1},$$

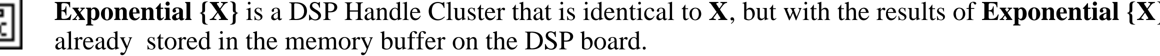
where f is the **final value**, and
n is the number of elements in **X**.

You can use the Exponential Window VI to analyze transients.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Exponential {X}.**

final value defaults to 0.10.



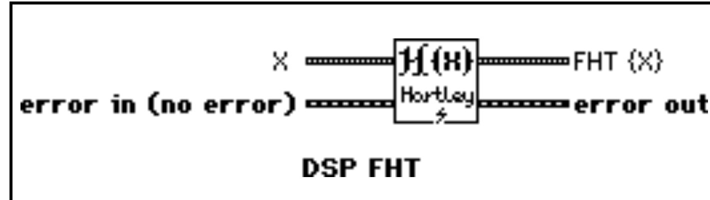
Exponential {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Exponential {X}** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP FHT

Computes the fast Hartley transform (FHT) of the input sequence **X**. The Hartley transform of a function x(t) is defined as follows:



$$X(f) = \int_{-\infty}^{\infty} x(t)\, \mathrm{cas}(2\pi ft)\, dt$$

where cas(x) = cos(x) + sin(x).

If Y represents the output sequence **FHT {X}** obtained via the FHT, then Y is obtained through the discrete implementation of the Hartley integral:

$$Y_k = \sum_{i=0}^{n-1} X_i \ \mathrm{cas}\left(\frac{2\pi ik}{n}\right), \qquad \text{for } k = 0, 1, 2, \ldots, n-1 .$$

where n is the number of elements in **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Notes: The number of elements for the input array must be a power of two.

The operation is performed in place and the input array **X** is overwritten by the output **FHT{X}**.

The largest FHT that can be computed depends upon the amount of memory on your DSP board.

FHT{X} is a DSP Handle Cluster that is identical to **X**, but with the results of **FHT{X}** already stored in the memory buffer on the DSP board.

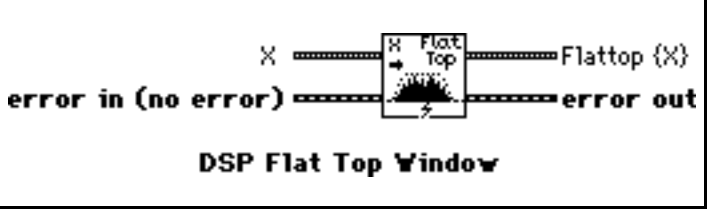
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The Hartley transform maps real-valued sequences into real-valued frequency domain sequences. You can use it instead of the Fourier transform to convolve signals, deconvolve signals, correlate signals, and find the power spectrum. Furthermore, you can derive the Fourier transform from the Hartley transform.

When the sequences to be processed are real-valued sequences, the Fourier transform produces complex-valued sequences in which half of the information is redundant. The advantage of using the Hartley transform instead of the Fourier transform is that the Hartley transform uses half the memory to produce the same information the FFT produces. Further, the FHT is calculated in place and is as efficient as the Fourier transform.

DSP Flat Top Window



Applies a flat top window to the input sequence **X**. If Y represents the output sequence **Flattop{X}**, the elements of Y are obtained using the formula:

$$y_i = x_i\,[0.2810639 - 0.5208972\cos(w) + 0.1980399\cos(2w)]$$

for $i = 0, 1, 2, \ldots, n-1$

$$w = \frac{2\pi i}{n},$$

where n is the number of elements in **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Flattop {X}.**

Flattop {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Flattop {X}** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Force Window

Applies a force window to the input sequence **X** If Y represents the output sequence **Force{X}**, the elements of Y are obtained using the formula:

$$y_i = \begin{cases} x_i & \text{if } 0 \le i \le d \\ 0 & \text{elsewhere} \end{cases}$$

for $i = 0, 1, 2, \ldots, n-1$,

$d = (0.01)(n)(\textbf{duty})$

where n is the number of elements in **X**.

You can use the Force Window VI to analyze transients.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Force {X}.**

duty. 0.0 ≤ **duty** ≤ 100.0. **duty** defaults to 50.0.

Force {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Force {X}** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Free Memory

Frees the memory space referred by **DSP Handle Cluster** on the specified DSP board.

DSP Handle Cluster is a DSP Handle Cluster that indicates the memory buffer to free.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and the VI also frees the DSP Handle Cluster.

error out contains the error information for this call.

DSP Gaussian White Noise

Generates a Gaussian distributed pseudorandom pattern whose statistical profile is as follows:

$$(\mu, \sigma) = (0, s),$$

where s is the absolute value of the specified **standard deviation**.

standard deviation defaults to 1.0.

seed < 65536.0. If **seed** > 0.0, the generated noise will be the same in repeated invocations if the **seed** value does not change. If **seed** ≤ 0.0, the VI generates a random value to use as the seed, and the noise will differ in repeated invocations although the value in **seed** does not change. **seed** defaults to 0.0.

Gaussian Pattern in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output Gaussian pattern.

Gaussian Pattern out is a DSP Handle Cluster that is identical to **Gaussian Pattern in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest Gaussian pattern that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP General Cosine Window

Applies a general cosine window to the input sequence **X**. If A represents the **Cosine Coefficients** input sequence and Y represents the output sequence **GenCos{X}**, the elements of Y are obtained using the formula:

X
Cosine Coefficients
error in (no error)
GenCos {X}
error out
DSP General Cosine Window

$$y_i = x_i \sum_{k=0}^{m-1} (-1)^k a_k \cos(kw)$$

for $i = 0, 1, 2, \ldots, n-1$

$$w = \frac{2\pi i}{n},$$

where n is the number of elements in **X**, and
m is the number of elements in **Cosine Coefficients**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **GenCos {X}.**

Cosine Coefficients is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the cosine coefficients.

GenCos {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **GenCos {X}** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Hamming Window

Applies a Hamming window to the input sequence **X**. If Y represents the output sequence **Hamming {X}**, the elements of Y are obtained from the formula:

$$y_i = x_i \, [0.54 - 0.46 \cos(w)]$$

for $i = 0, 1, 2, \ldots, n-1$,

$$w = \frac{2\pi i}{n},$$

where n is the number of elements in the input sequence **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input data array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Hamming {X}.**

Hamming {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Hamming {X}** already stored in the memory buffer on the DSP board.

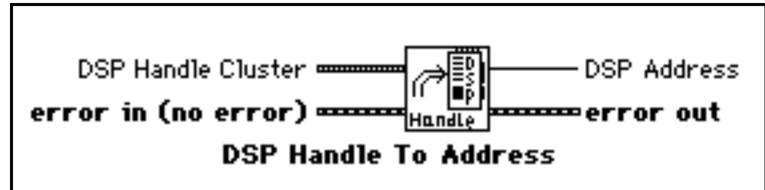
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

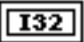
DSP Handle To Address

Finds the actual **DSP address** value of **DSP Handle Cluster** that indicates a memory buffer on the DSP board.

You can use the output **DSP Address** as the input of the DSP Address terminal in the DSP DMA Copy(DSP to LV) VI or the DSP DMA Copy(LV to DSP) VI.



DSP Handle Cluster indicates a memory buffer on the DSP board.



DSP Address is the actual DSP address of the memory buffer on the DSP board referred to by **DSP Handle Cluster**.

error in(no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Hanning Window

Applies a Hanning window to the input sequence **X**. If Y represents the output sequence **Hanning {X}**, the elements of Y are obtained using the formula:

$$y_i = 0.5\, x_i\, [1 - \cos(w)]$$

for $i = 0, 1, 2, \ldots, n-1$,

$$w = \frac{2\pi i}{n},$$

where n is the number of elements in the input sequence **X.**

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input data array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Hanning {X}.**

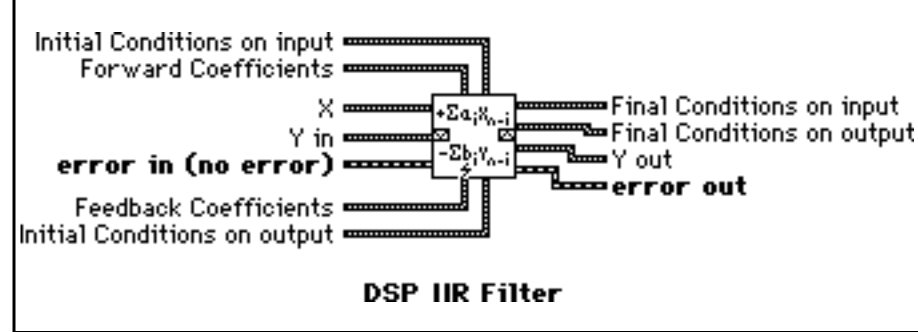
Hanning {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Hanning {X}** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP IIR Filter

Performs IIR filtering on the **X** input array and reports the result in **Y**. It uses the arrays **a** and **b** of sizes **sza** and **szb** respectively in implementing the linear difference equation that describes IIR filtering:



$$\mathbf{y}[i] = \sum_{k=0}^{\mathbf{sza}-1} \mathbf{a}(k) * \mathbf{x}(i-k) - \sum_{k=1}^{\mathbf{szb}-1} \mathbf{b}(k) * \mathbf{y}(i-k) \qquad i = 0,1 \,.....\, ,\mathbf{n}-1$$

where:

a is the array of forward coefficients describing an IIR filter (obtained from DSP Butterworth Coefficients VI for example)

b is the array of feedback coefficients describing an IIR filter (obtained from the same filter design VI)

Notes: **b**`(0) = 1.0` in the above equation.

This VI may be called in a loop to perform filtering on data frames that are part of the same data stream.

At time i = 0, the initial conditions on **X** (for example, **X**(1-**sza**) through **X**(-1)) are obtained from the array **Initial Conditions on input.** At exit (the end of the VI execution), this array holds the final conditions on **X** that could be used as the initial conditions for the next DSP IIR Filter VI call if you were to call it in a loop.

At time i = 0, the initial conditions on **Y** (for example, **Y**(1-**szb**) through **Y**(-1)) are obtained from the array **Initial Conditions on output**. At exit (the end of the VI execution), this array holds the final conditions on **Y** that could be used as the initial conditions for the next DSP IIR Filter VI call if you were to call it in a loop.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output array of filtered data.

Initial Conditions on input is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the initial conditions on input at time zero.

Initial Conditions on output is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the initial conditions on output at time zero.

Forward Coefficients is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the forward coefficients of the IIR filter.

Feedback Coefficients is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the feedback coefficients of the IIR filter.

Y out is a DSP Handle Cluster that is identical to **Y in**, but with the filtered data already stored in the memory buffer on the DSP board.

Final Conditions on input is a DSP Handle Cluster that is identical to **Initial Conditions on input**, but contains the final conditions on input.

Final Conditions on output is a DSP Handle Cluster that is identical to **Initial Conditions on output**, but contains the final conditions on output.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

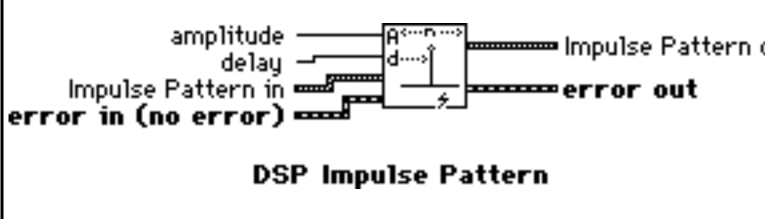
error out contains the error information for this call.

The operation can be performed in place; that is, the input X and the output Y can be the same DSP Handle Cluster.

Parameter Discussion

The arrays **X** and **Y** should be the same size. The arrays **Forward Coefficients** and **Initial/Final Conditions on input** should be the same size. The arrays **Feedback Coefficients** and **Initial/Final Conditions on output** should be the same size. In successive calls to this VI, while performing filtering on large data streams, one buffer at a time, the final conditions of the filter are stored in **Final Condition on input** and in **Final Condition on output**, and are used as the initial conditions of the filter in the subsequent call. For example, if you designed a filter with all of the initial conditions set to zero, and you then wanted to filter a stream of 8,192 data points, 1,024 points at a time, after the first call, the final conditions of the filter are stored in **Final Condition on input** and in **Final Condition on output**. These can be used as the initial conditions to the filter in the filtering of the next 1,024 points, and so on.

DSP Impulse Pattern



Generates an array containing an impulse pattern. If the **Impulse Pattern** is represented by the sequence X, the VI generates the pattern according to the following formula:

$$x_i = \begin{cases} a & \text{if } i = d \\ 0 & \text{elsewhere} \end{cases}$$

for $i = 0, 1, 2, \ldots, n-1$,

where a is the **amplitude**,
d is the **delay**, and
n is the number of elements in **Impulse Pattern**.

amplitude defaults to 1.0.

delay must be greater than or equal to 0. If **delay** is less than zero or greater than or equal to the size of **Impulse Pattern**, the VI returns an error. **delay** defaults to 0.

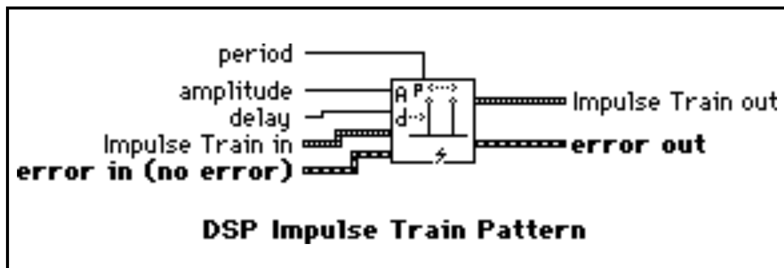
Impulse Pattern in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output impulse pattern.

Impulse Pattern out is a DSP Handle Cluster that is identical to **Impulse Pattern in**, **but** with the generated pattern already stored in the memory buffer on the DSP board. The largest impulse pattern that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Impulse Train Pattern



Generates a train of impulses of value **amplitude** at sample **delay**.

If the impulse train pattern is represented by the sequence X, the VI generates the pattern according to the following formula:

$$x_i = \begin{cases} \textbf{amplitude} & \text{if } i \text{ modulo } P = d \\ 0.0 & \text{elsewhere} \end{cases}$$

for $i = 0, 1, 2, \ldots, n-1$,

where P is the **period**,
d is the **delay**, and
n is the number of elements in the impulse train.

amplitude defaults to 1.0.

delay defaults to 0.

period defaults to 1.

Impulse Train in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output impulse train.

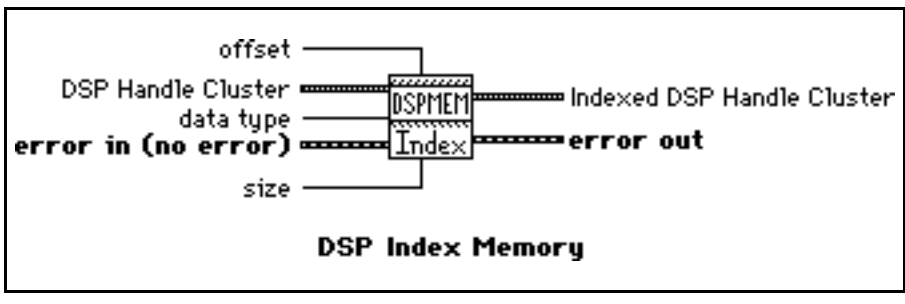
Impulse Train out is a DSP Handle Cluster that is identical to **Impulse Train in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest impulse train that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

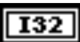
DSP Index Memory

Indexes into a DSP buffer allocated in the memory space of the specified DSP board. The return value is another DSP Handle Cluster. This VI does not allocate memory.



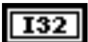
In order to index into a buffer correctly, you need to indicate what type of data is in the DSP buffer. You must set **data type** to the appropriate type. **data type** has two options–4 bytes (for 32-bit long and floating-point data) and 2 bytes (for 16-bit short data).

DSP Handle Cluster is a DSP Handle Cluster into which you want to index.



data type indicates the type of data in the buffer on the DSP board that is referred to by **DSP Handle Cluster**. **data type** has two options:
0: 4 bytes (for 32-bit long and floating-point data).
1: 2 bytes (for 16-bit short data).

data type defaults to 4 bytes.



offset is the index where to start to index into the array referenced by **DSP Handle Cluster. offset** defaults to 0.

Indexed DSP Handle Cluster is a DSP Handle Cluster that contains the number of **size** data in **DSP Handle Cluster** starting from **offset**.

Notes: If you free a DSP Handle Cluster using the DSP Free Memory VI, then all other DSP Handle Clusters obtained by using DSP Index Memory VI will no longer be valid.

This VI returns a new DSP Handle Cluster. If you no longer need this DSP Handle Cluster, remember to free it.

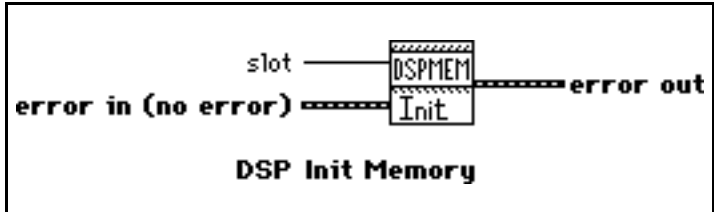
size indicates how many elements you want to index starting from the **offset**. **size** defaults to 0.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Init Memory

Initializes the memory heaps and frees all allocations of memory on the specified DSP board.



slot is the board ID number. **slot** defaults to 3.

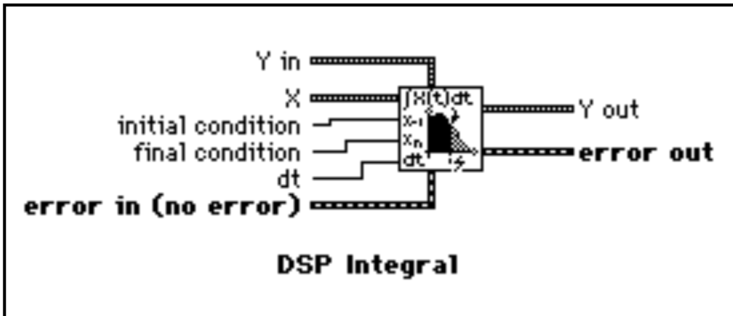
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Integral

Performs the discrete integration of the sampled signal **X**. The integral F(t) of a function f(t) is defined as follows:

$$F(t) = \int f(t)\ dt\ .$$



Let Y represent the sampled output sequence **Integral X**. The VI obtains the elements of Y using the following formula:

$$y_i = \frac{1}{6} \sum_{j=0}^{i} (x_{j-1} + 4x_j + x_{j+1})\, \mathbf{dt} \qquad \text{for } i = 0, 1, 2, \ldots, n-1\ ,$$

where n is the number of elements in **X**,
x_{-1} is specified by **initial condition** when i = 0, and
x_n is specified by **final condition** when i = n-1.

The **initial condition** and **final condition** minimize the overall error by increasing the accuracy at the boundaries, especially when the number of samples is small. Determining boundary conditions before the fact enhances accuracy.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

initial condition defaults to 0.0.

final condition defaults to 0.0.

dt is the sampling interval and must be greater than zero. If **dt** is less than or equal to zero, the VI returns an error. **dt** defaults to 1.0.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of the integration of X.

Y out is a DSP Handle Cluster that is identical to **Y in**, but with the results of integration already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation can be performed in place; that is, the input X and the output Y can be the same DSP Handle Cluster.

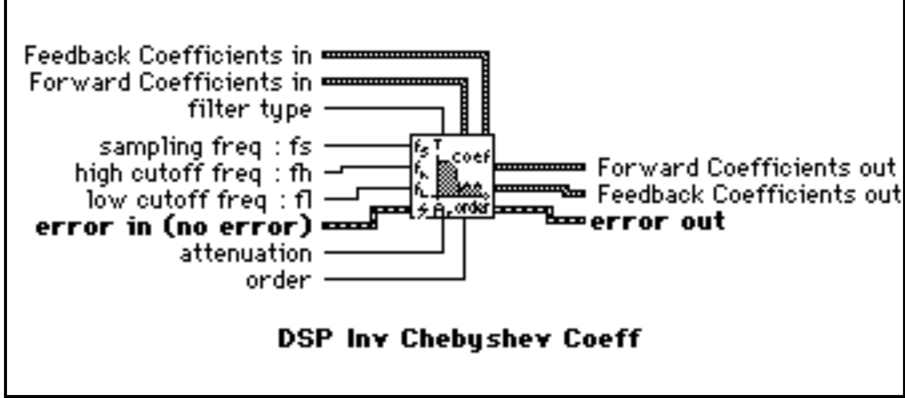
Note: You can also use the DSP Integral x(t) VI to numerically evaluate the finite integral:

$$\int_a^b f(t)\ dt = F(b) - F(a) \approx y_{n-1}\ ,$$

by extracting the last element of the output sequence Y.

DSP Inv Chebyshev Coeff

Generates the set of filter coefficients to implement an inverse IIR filter as specified by the Chebyshev II Filter mode. You can then pass these coefficients to the DSP IIR Filter VI to filter a sequence of data.



filter type specifies the passband of the filter. **filter type** has four options:
0: lowpass.
1: highpass.
2: bandpass.
3: bandstop.

filter type defaults to lowpass.

sampling Freq : fs is the sampling frequency and must be greater than 0. If it is less than or equal to zero, the VI returns an error. **sampling Freq : fs** defaults to 1.0.

high cutoff freq: fh is the high cutoff frequency. The VI ignores this parameter when **filter type** is lowpass. **high cutoff freq: fh** defaults to 0.45.

low cutoff freq: fl is the low cutoff frequency. The VI ignores this parameter when **filter type** is highpass. **low cutoff freq: fl** defaults to 0.125.

Note: fh and fl must observe the Nyquist criterion: $0 \leq fl \leq fh \leq fs/2$.

attenuation must be greater than 0.0. If **attenuation** is less than or equal to zero, the VI returns an error. **attenuation** defaults to 60.0.

order must be greater than zero. If the filter **order** is less than or equal to zero, the VI returns an error. **order** defaults to 2.

Forward Coefficients in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the forward coefficients.

Feedback Coefficients in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the feedback coefficients.

Forward Coefficients out is a DSP Handle Cluster that is identical to **Forward Coefficients in**, but with the forward coefficients already stored in the memory buffer on the DSP board.

Feedback Coefficients out is a DSP Handle Cluster that is identical to **Feedback Coefficients in**, but with the feedback coefficients already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

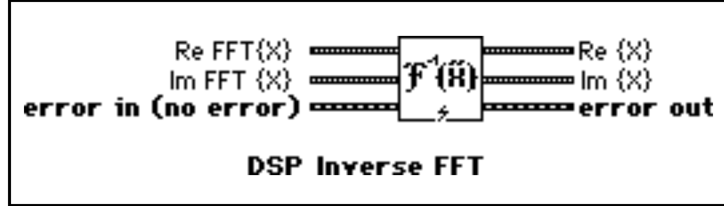
Parameter Discussion

The arrays **Forward Coefficients in/out** and **Feedback Coefficients in/out** must have a size of at least (**order** + 1) for lowpass and highpass filters. The arrays **Forward Coefficients in/out** and **Feedback Coefficients in/out** must have a size of at least (2***order** + 1) for bandpass and bandstop filters.

DSP Inverse FFT

Computes the inverse Fourier transform of the complex input sequence **FFT {X}**. If Y represents the output sequence, then:

$$Y = \boldsymbol{F}^{-1}\{X\}.$$



Re FFT{X} is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the real part of input signal array FFT {X}.

Im FFT{X} is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the imaginary part of input signal array FFT {X}.

Notes: The number of elements for the input array must be the power of two. If the size of **Re FFT{X}** is different from the size of **Im FFT{X}**, the VI uses the smaller number as the size of the inverse FFT.

The operation is performed in place and the input array **Re FFT{X }** and **Im FFT{X}** is overwritten by **Re {X}** and **Im {X}.**

The largest inverse FFT that can be computed depends upon the amount of memory on your DSP board.

Re {X} is a DSP Handle Cluster that is identical to **Re FFT{X }**, but with the real part of X already stored in the memory buffer on the DSP board.

Im {X} is a DSP Handle Cluster that is identical to the **Im FFT{X}** but with the imaginary part of X already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Inverse FHT

Computes the inverse fast Hartley transform of the input sequence **FHT {X}**. The inverse Hartley transform of a function X(f) is defined as

$$x(t) = \int_{-\infty}^{\infty} X(f)\, \mathrm{cas}(2\pi ft)\, df$$

where cas(x) = cos(x) + sin(x).

If Y represents the output sequence **X**, the VI calculates Y through the discrete implementation of the inverse Hartley integral:

$$Y_k = \frac{1}{n} \sum_{i=0}^{n-1} X_i \ \mathrm{cas}\left(\frac{2\pi\, ik}{n}\right) \qquad \text{for } k = 0, 1, 2, \ldots, n-1$$

where n is the number of elements in **X**.

The inverse Hartley transform maps real-valued frequency sequences into real-valued sequences. You can use it instead of the inverse Fourier transform to convolve, deconvolve, and correlate signals. Furthermore, you can derive the Fourier transform from the Hartley transform.

See the description of the DSP FHT VI for a comparison of the Fourier and Hartley transforms.

FHT {X} is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Notes: The number of elements for the input array must be a power of 2.

The operation is performed in place and the input array **FHT {X}** is overwritten by the output array **X.**

The largest inverse FHT that can be computed depends upon the amount of memory in your DSP board.

X is a DSP Handle Cluster that is identical to **FHT {X}**, but with the results of inverse **FHT {X }** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Kaiser-Bessel Window

Applies a Kaiser-Bessel window to the input sequence **X**. If Y represents the output sequence **Kaiser-Bessel{X}**, the elements of Y are obtained using the formula:

X
beta
error in (no error)
Kaiser-Bessel {X}
error out
DSP Kaiser-Bessel Window

$$y_i = x_i \frac{I_o\left(\beta\sqrt{1.0 - a^2}\right)}{I_o(\beta)}$$

for $i = 0, 1, 2, \ldots n - 1$

$$a = \frac{i - k}{k}$$

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

where n is the number of elements in **X**, and
$I_O(\bullet)$ is the zeroth-order modified Bessel function.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Kaiser-Bessel {X}.**

beta is proportional to the sidelobe attenuation–that is, the larger **beta** is, the greater the sidelobe attenuation is. **beta** defaults to 0.0.

Kaiser-Bessel {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Kaiser-Bessel {X}** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

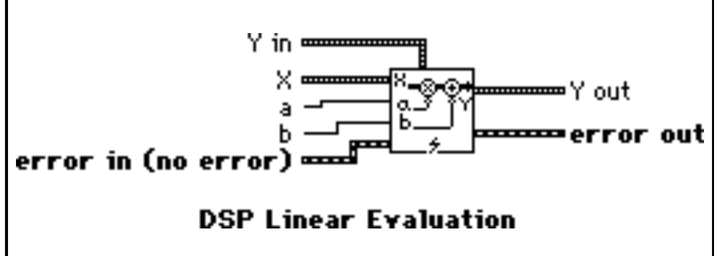
error out contains the error information for this call.

DSP Linear Evaluation

Performs a linear evaluation of the input array X. The ith element of the output array Y is obtained using the following formula:

$$Y(i) = a * X(i) + b \quad \text{for } i = 0, 1, 2, \ldots, n-1,$$

where n is the number of elements in **X**.



X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

a is the multiplicative constant. **a** defaults to 1.0.

b is the additive constant. **b** defaults to 0.0.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of output array.

Y out is a DSP Handle Cluster that is identical to **Y in**, but with the results already stored in the memory buffer on the DSP board.

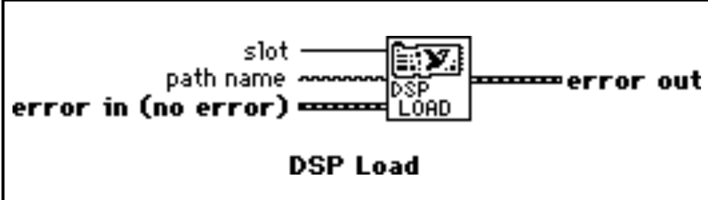
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation can be performed in place; that is, the input and output arrays can be the same DSP Handle Cluster.

DSP Load

Downloads a library specified by **path name** to the DSP board in **slot**. **path name** must contain the full path name of a valid COFF file (`.out` file).



slot is the board ID number. **slot** defaults to 3.

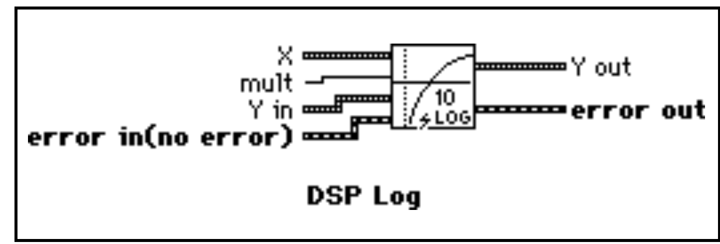
path name is the path name of the `.out` file.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Log

Computes the logarithm base 10 of the **X** input array. The ith element of resulting array is obtained by using the following formula:



$$y(i) = \log10(X(i)) * \textbf{mult}$$

for $i = 0, 1, 2, \ldots, n-1$,

where n is the number of elements in **X**.

This VI is useful for converting values that represent power to decibels. This VI returns the most negative number for any input less than or equal to zero.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

mult defaults to 1.0.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of logarithm X.

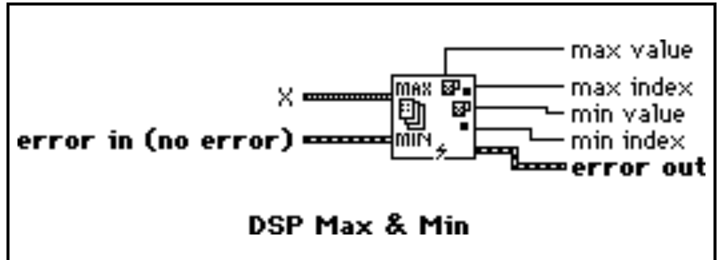
Y out is a DSP Handle Cluster that is identical to **Y in**, but with the results of logarithm X already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Max & Min

Finds the maximum and minimum values in the input array, as well as the respective indices of the occurrence of the maximum and minimum values.



X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

max value is the maximum value of the input array.

max index is the index into the input array where maximum occurred.

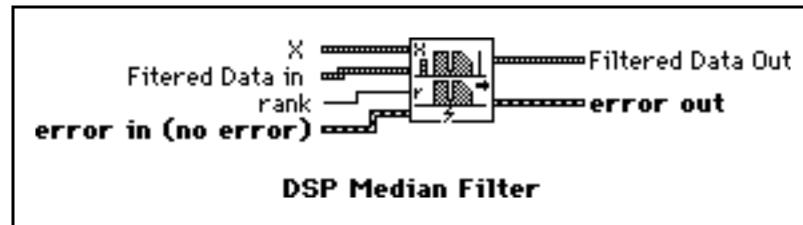
min value is the minimum value of the input array.

min index is the index into the input array where minimum occurred.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Median Filter



Applies a median filter of **rank** to the input sequence **X**. The median filter is a nonlinear filter that combines lowpass filters characteristics (to remove high-frequency noise) and high-frequency characteristics (to detect edges).

If Y represents the output sequence **Filtered Data**, if J_i represents a subset of the input sequence **X** centered about the ith element of **x:**

$$J_i = \{x_{i-r}, x_{i-r+1}, \ldots, x_{i-1}, x_i, x_{i+1}, \ldots, x_{i+r-1}, x_{i+r}\},$$

and if the indexed elements outside the range of X equal zero, the VI obtains the elements of Y using:

$$y_i = \text{Median}(J_i) \qquad \text{for } i = 0, 1, 2, \ldots, n-1,$$

where n is the number of elements in the input sequence **X**, and
r is the filter **rank**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input data array. The number of elements in **X** must be greater than the **rank**:

$$n > r \geq 0.$$

If the number of elements in **X** is less than or equal to **rank**, the VI returns an error.

rank must be greater than or equal to zero. If **rank** is less than zero, the VI returns an error. **rank** defaults to 2.

Filtered Data in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the filtered data output.

Filtered Data out is a DSP Handle Cluster that is identical to **Filtered Data in**, but with the filtered data already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

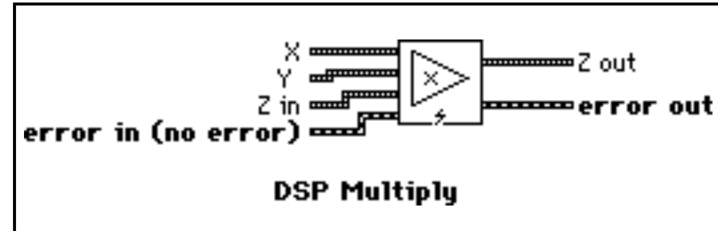
The operation cannot be performed in place; that is, the input **X** and the output **Filtered Data** cannot be the same DSP Handle Cluster.

DSP Multiply

Multiply array **X** by array **Y.** The ith element of the output array **Z** is obtained using the following formula:

$$Z(i) = X(i) * Y(i) \quad \text{for } i = 0, 1, 2, \ldots, n-1,$$

where n is the smaller number of elements in **X** and **Y**.



X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Y is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array Y.

Z in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of X(i) * Y(i).

Z out is a DSP Handle Cluster that is identical to **Z in**, but with the multiplied results already stored in the memory buffer on the DSP board.

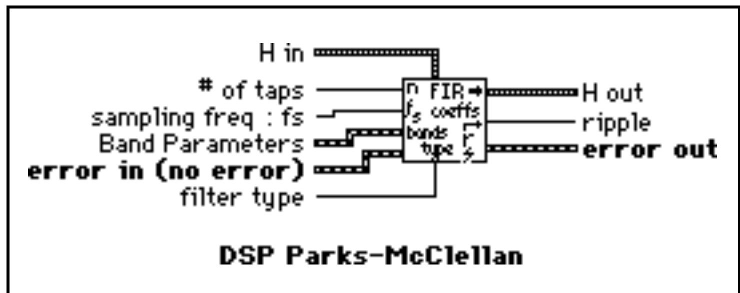
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation can be performed in place; that is, the input and output arrays can be the same DSP Handle Cluster.

DSP Parks-McClellan

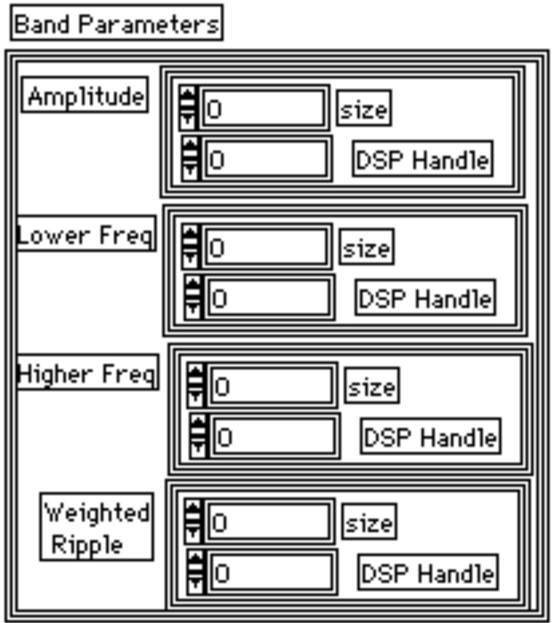
Generates a set of linear-phase finite impulse response multiband digital filter coefficients using the number of taps, sampling frequency, filter type, and **Band Parameters**.



# of taps is the total number of coefficients in **H**. A tap corresponds to a multiplication and an addition. If there are n taps, every filtered sample requires n multiplications and n additions. **# of taps** must be greater than 0. If **# of taps** is less than or equal to zero, the VI returns an error. **# of taps** defaults to 32.

sampling freq : fs defaults to 1.0.

Band Parameters is a cluster. Each cluster element contains the necessary information associated with each band for the FIR design. Each cluster contains four elements, as shown in the following figure:



The **Band Parameters** cluster must contain at least one element, that is, one band.

amplitude is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the desired amplitude for each band.

lower freq is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the lower frequency bands.

higher freq is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the upper frequency bands.

weighted ripple is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the weighting factor for each band.
For each band, **higher freq** must be greater than **lower freq**, and for adjacent bands, **lower freq** in the higher band must be greater than **higher freq** in the lower band,

$$f_{h_i} > f_{l_i}, \quad \text{for } i = 0, 1, 2, \ldots, m\text{-}1,$$

$$f_{l_{i+1}} > f_{h_i}, \quad \text{for } i = 0, 1, 2, \ldots, m\text{-}2,$$

where f_{l_i} represents the **lower freq** in the ith band, and f_{h_i} represents the **higher freq** in the ith band.

The **higher freq** in the last band must observe the Nyquist criterion:

$$f_{h_{m-1}} \leq 0.5 f_s,$$

where f_s is the sampling frequency.

If **Band Parameters** does not contain any elements, or if any of the preceding frequency conditions is violated, the VI returns an error.

filter type has three options:
0: multiband.
1: differentiator.
2: Hilbert.

filter type defaults to multiband.

ripple is the optimal ripple the VI computes and is a measure of deviation from the ideal filter specifications.

H in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the FIR coefficients.

H out is a DSP Handle Cluster that is identical to **H in**, but with the coefficients already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

Note: The DSP Parks-McClellan VI finds the coefficients using iterative techniques based upon an error criterion. Although you specify valid filter parameters, the algorithm may fail to converge.

The DSP Parks-McClellan VI generates only the filter coefficients. It does not perform the filtering function. To filter a sequence X using the set of FIR filter coefficients **H**, use the DSP Convolution VI with **X** and **H** as the input sequences.

The equi-ripple filters use a similar technique to filter the data.

Parameter Discussion

The weights are usually the same for every band and inversely proportional to frequency *f* for a differentiate. The amplitudes of the bands are usually the same for every band and form a slope for a differentiation. The designed filter is unconstrained, that is the frequency response in transitional period specifications is not guaranteed. Adjust the parameters if there are discrepancies between specifications and results.

Generally, when type = multiband and **bands** ≥ 2, the DSP Parks-McClellan VI designs a multiband filter. When **filter type** = differentiator and **bands** = 1, the DSP Parks-McClellan VI designs a differentiation. When **filter type** = Hilbert and **bands** = 1, the DSP Parks-McClellan VI designs a Hilbert transform. The following tables contain the brief requirements you must meet when you design different types of filters. For more information, please refer to *Digital Filter Design* by Parks and Burrus, or "A Computer Program for Designing Optimum FIR Linear Phase Digital Filters," by McClellan, Parks, and Rabiner.

When **filter type** = Multiband:

Filter Band Type	LowPass	HighPass	BandPass	BandStop
# of Bands (n)	2	2	≥3	≥3
# of taps	odd/even	odd	odd/even	odd
fl[0]	0.0	0.0	0.0	0.0
fh[n-1]	0.5fs	0.5fs	0.5fs	0.5fs
amp[0]	>0.0	≥0.0	≥0.0	>0.0
amp[n-1]	0.0(for even taps) ≥0.0(for odd taps)	>0.0	0.0(for even taps) ≥0.0(for odd taps)	>0.0

fl[0] represents the first value in the array of **Lower Freq.**
fh[n-1] represents the last value in the array of **Higher Freq**.
amp[0] represents the first value in the array of **Amplitude**.
amp[n-1] represents the last value in the array of **Amplitude**.
fs is the sample frequency.

When **filter type** = Differentiator or Hilbert:

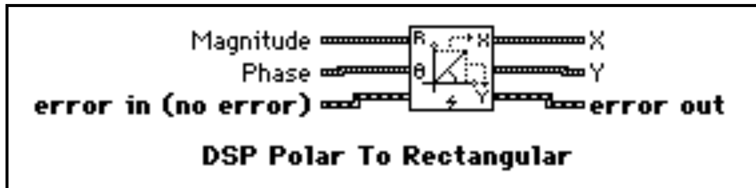
Filter Type	Diff(odd)	Diff(even)	Hilbert(odd)	Hilbert(even)
bands(n)	≥1	≥1	≥1	≥1
fl[0]	0.0	0.0	0.0+Δ	0.0+Δ
fh[n-1]	0.5fs-Δ	0.5fs	0.5fs-Δ	0.5fs

odd or even in the table refers to the value of the **# of taps.**
fl[0] represents the first value in the array of **Lower Freq.**
fh[n-1] represents the last value in the array of **Higher Freq**.
fs is the sample frequency.
Δ is a small number that specifies the transitional band.

Although the DSP Parks-McClellan VI is the most flexible way to design a FIR linear phase filter, it has more complex parameters and requires some DSP knowledge. You may find it more convenient to use DSP Equi-Ripple LowPass, DSP Equi-Ripple HighPass, DSP Equi-Ripple BandPass, and DSP Equi-Ripple BandStop. These functions, which provide lowpass, highpass, bandpass and bandstop FIR filters with equal weighting factors in all bands, are special cases of the DSP Parks-McClellan VI with simplified parameters.

DSP Polar to Rectangular

Converts a set of polar coordinate points (**Magnitude, Phase**) to a set of rectangular coordinate points (**X**, **Y**). The ith elements of the rectangular set is obtained using the following formulas:



X(i) = **Magnitude**(i) * cos(**Phase**(i))
Y(i) = **Magnitude**(i) * sin(**Phase**(i))

Note: The operation is performed in place and the input arrays **Magnitude** and **Phase** are overwritten by **X** and **Y**.

Magnitude is the DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array **Magnitude**.

Phase is the DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array **Phase**.

X is the DSP Handle Cluster that is identical to **Magnitude** but with the caculated rectangular coordinate values already stored in the memory buffer on the DSP board.

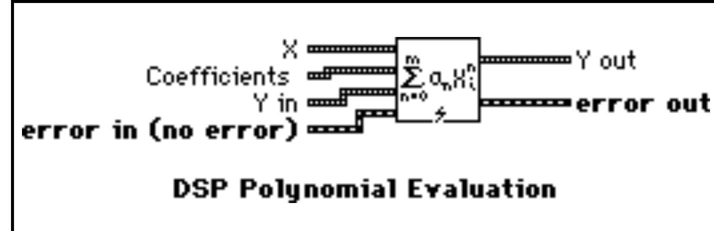
Y is the DSP Handle Cluster that is identical to **Phase** but with the caculated rectangular coordinate values already stored in the memory buffer on the DSP board.

error in(no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Polynomial Evaluation

Performs a polynomial evaluation on the input array X. The ith element of the output array Y is obtained using the following formula:



$$\mathbf{Y}(i) = \sum_{j=0}^{\mathbf{k}-1} (\mathbf{Coefficients}(j) * \mathbf{X}(i)^j)$$

for $i = 0, 1, 2, \ldots, n-1$,

where n is the number of elements in **X** and k is the number of elements in **Coefficients**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Coefficients is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the coefficients array.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of output array.

Y out is a DSP Handle Cluster that is identical to **Y in**, but with the results already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

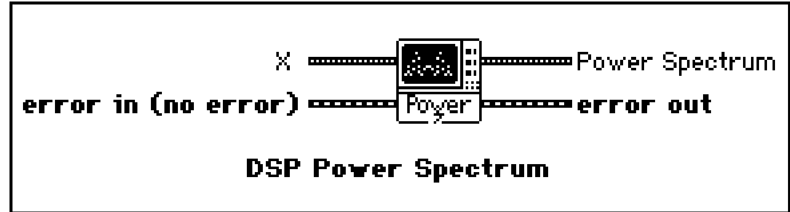
The operation can be performed in place; that is, the input X and the output Y can be the same DSP Handle Cluster.

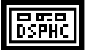
DSP Power Spectrum

Computes the **Power Spectrum** of the input sequence **X**. The **Power Spectrum** $S_{xx}(f)$ of a function x(t) is defined as

$$S_{xx}(f) = X^*(f)X(f) = |X(f)|^2,$$

where $X(f) = \boldsymbol{F}\{x(t)\}$, and $X^*(f)$ is the complex conjugate of X(f).





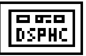
X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Notes: The number of elements for the input array must be a power of two.

The operation is performed in place and the input array **X** is overwritten by the output **Power Spectrum.**

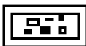
The largest power spectrum that can be computed depends upon the amount of memory on your DSP board.

This VI allocates a temporary workspace on the DSP board equal to the size of the input signal array.



Power Spectrum is a DSP Handle Cluster that is identical to **X**, but with the results of **Power Spectrum** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

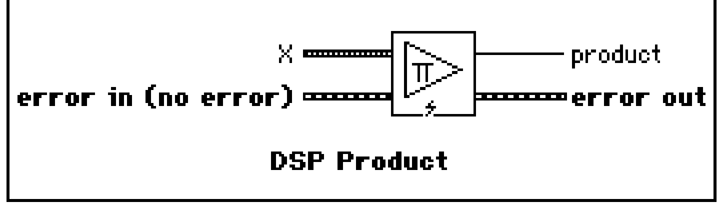


error out contains the error information for this call.

DSP Product

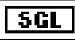
Finds the product of the elements of the input array **X.** The product of the elements is obtained using the following formula:

$$\textbf{product} = \prod_{i=0}^{n-1} X(i)$$



where n is the smaller number of elements in **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

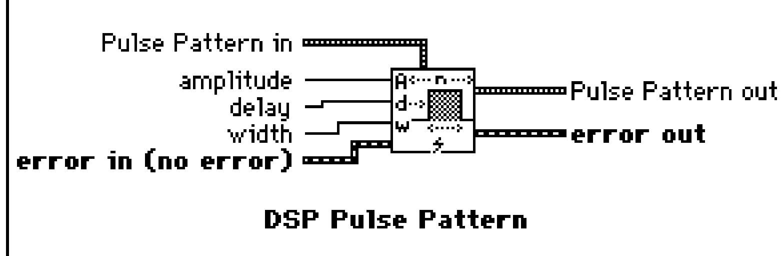


product is the product of the elements in **X.**

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Pulse Pattern



Generates an array containing a pulse pattern. If **Pulse Pattern** is represented by the sequence X, then the pattern is generated according to the following formula:

$$x_i = \begin{cases} a & \text{if } d \le i < last \\ 0 & \text{elsewhere} \end{cases}$$

$$\text{for } i = 0, 1, 2, \ldots, n-1,$$

$$last = d + w,$$

where a is the **amplitude**,
d is the **delay**,
w is the **width**, and
n is the number of elements in **Pulse Pattern**.

amplitude defaults to 1.0.

delay must be greater than or equal to 0. **delay** defaults to 0.

width must be greater than or equal to 0. **width** defaults to 1.

Pulse Pattern in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output pulse pattern.

Pulse Pattern out is a DSP Handle Cluster that is identical to **Pulse Pattern in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest pulse pattern that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

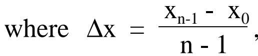
DSP Ramp Pattern

Generates an array containing a ramp pattern. If the **Ramp Pattern** is represented by the sequence X, then the pattern is generated according to the formula:

$$x_i = x_0 + i\,\Delta x$$

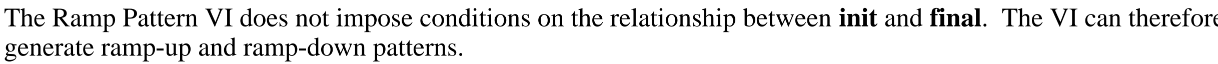
for $i = 0, 1, 2, \ldots, n-1$,

where $\Delta x = \dfrac{x_{n-1} - x_0}{n - 1}$,



x_{n-1} is the **final**,
x_0 is the **init**, and
n is the number of elements in **Ramp Pattern**.

The Ramp Pattern VI does not impose conditions on the relationship between **init** and **final**. The VI can therefore generate ramp-up and ramp-down patterns.



final defaults to 0.0.

init defaults to 1.0.

Ramp Pattern in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output ramp pattern.

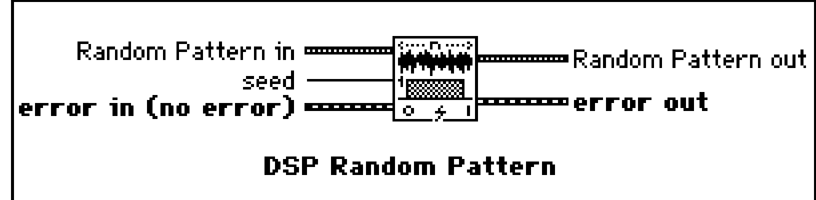
Ramp Pattern out is a DSP Handle Cluster that is identical to **Ramp Pattern in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest ramp pattern that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

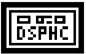
error out contains the error information for this call.

DSP Random Pattern

Generates a uniformly distributed pseudorandom pattern whose values are in the range [0:1]. The sequence is generated using the Very-Long-Cycle random number generator algorithm.



seed < 65536.0. If **seed** > 0.0, the generated noise will be the same in repeated invocations if the **seed** value does not change. If **seed** ≤ 0.0, the VI generates a random value to use as the **seed**, and the noise differs in repeated invocations although the value in **seed** does not change. **seed** defaults to 0.0.



Random Pattern in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output random pattern.

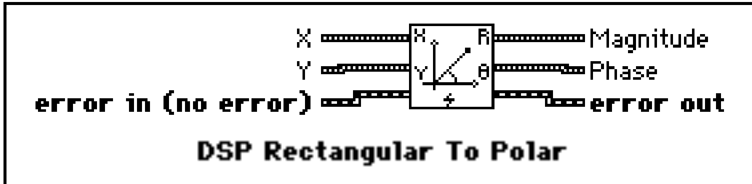
Random Pattern out is a DSP Handle Cluster that is identical to **Random Pattern in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest random pattern that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Rectangular To Polar

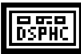
Converts a set of rectangular coordinate points (**X**, **Y**) to a set of polar coordinate points (**Magnitude**, **Phase**). The ith element of the polar coordinate set is obtained by using the following fomulas:



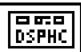
Magnitude(i) = $\sqrt{X(i)^2 + Y(i)^2}$
Phase(i) = arctan(**Y**(i)/**X**(i))

Note*:* The operation is perfomed in place and the input arrays **X** and **Y** are overwritten by **Magnitude** and **Phase**.

X is the DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array **X**.



Y is the DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array **Y**.



Magnitude is the DSP Handle Cluster that is identical to **X** but with the caculated polar coordinate amplitude values already stored in the memory buffer on the DSP board.

Phase is the DSP Handle Cluster that is identical to **Y** but with the caculated polar coordinate phase values already stored in the memory buffer on the DSP board.

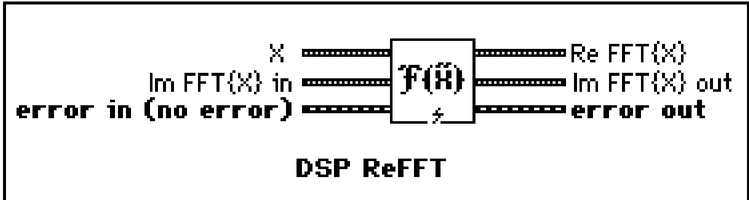
error in(no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP ReFFT

Computes the Fast Fourier transform of a real input sequence X. If Y represents the complex output sequence, then:

$$Y = \boldsymbol{F}\{X\}.$$



X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

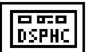
Notes: The number of elements for input X must be the power of two.

The operation is performed in place and the input array **X** is overwritten by **Re FFT{X}.**

The largest FFT that can be computed depends upon the amount of memory in your DSP board.

Im FFT{X} in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the imaginary part of FFT{X}.

Re FFT{X} is a DSP Handle Cluster that is identical to **X**, but with the real part of FFT{X}already stored in the memory buffer on the DSP board.



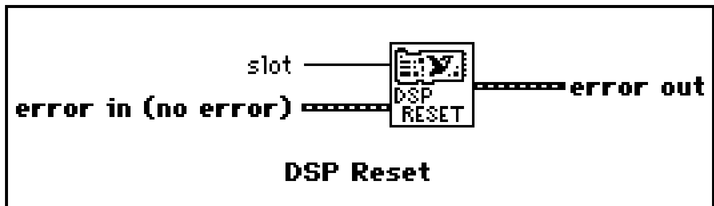
Im FFT{X} out is a DSP Handle Cluster that is identical to **Im FFT{X} in**, but with the imaginary part of FFT{X} already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Reset

DSP Reset aborts any currently running function on the board, stops the library, and reloads the DSP Library from the directory specified in the `WDAQCONF.EXE` file.



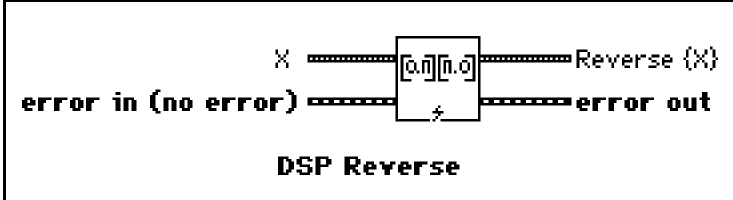
slot is the board ID number. **slot** defaults to 3.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Reverse

Reverse the order of the elements of the input array X.



X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

The operation can only be performed in place; that is, the input **X** is overwritten by the output **Reverse {X}**.

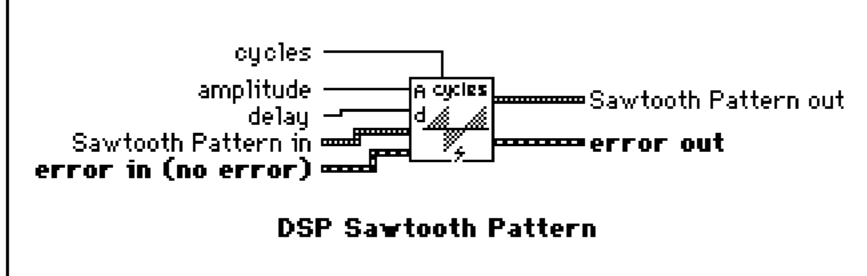
Reverse {X} is a DSP Handle Cluster that is identical to the **X**, but with the reverse {X} already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Sawtooth Pattern

Generate a sawtooth pattern with positive-slope, zero-crossing at sample **delay**. If the sawtooth pattern is represented by the sequence Y, then the pattern is generated according to the following formula:



$$y_i = \begin{cases} \frac{2a}{T} k & \text{if } 0 \le k < \frac{T}{2} \\ \frac{2a}{T} (k - T) & \text{if } \frac{T}{2} \le k < T \end{cases} \quad \text{for } i = 0, 1, 2, ..., n - 1$$

where $k = (i - d) \text{ modulo } T$ and $T = \frac{n}{cycles}$,
d is the delay, and
n is the number of elements in the sawtooth pattern.

amplitude defaults to 1.0.

delay defaults to 0.

cycle defaults to 1.0.

Sawtooth in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output sawtooth pattern.

Sawtooth out is a DSP Handle Cluster that is identical to **Sawtooth in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest sawtooth pattern that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

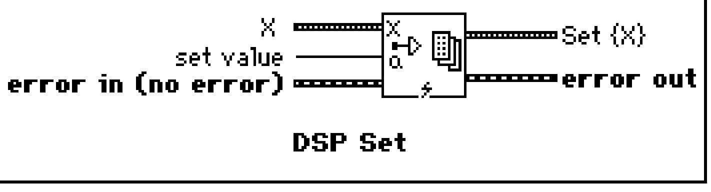
error out contains the error information for this call.

DSP Set

Set the elements of the input array X to the constant value **set value**. If the output **Set {X}** is represented by the sequence Y, then:

y_i = set value

for i = 0, 1, 2, …, n-1 ,



where n is the number of elements in **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X. The operation is performed in place; that is, the input **X** is overwritten by the output **Set {X}**.

set value defaults to 0.0.

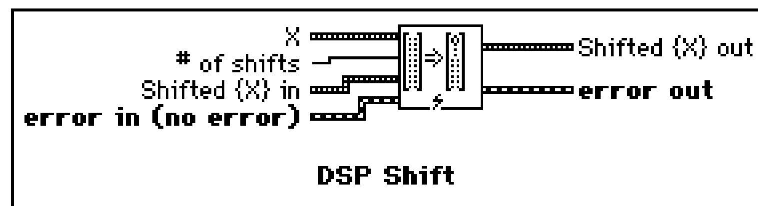
Set {X} is a DSP Handle Cluster that is identical to **X**, but with the set {X} already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Shift

Shifts the elements in the input array **X**, replacing the new values with zeros. The number of shifts selected can be in the positive (right) or negative (left) direction.



X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

# of shifts defaults to 0**.**

Shifted {X} in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output array of shifted {X}.

Shifted {X} out is a DSP Handle Cluster that is identical to **Shifted {X} in**, but with the shifted {X} already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation can be performed in place; that is, the input and output arrays can be the same DSP Handle Cluster.

Parameter Discussion

Shifting can be performed right or left by setting **# of shifts** to a positive or negative value, respectively. If **# of shifts** is greater than n, (n is the number of elements of input array **X**), the outputs are all zero.

DSP Sinc Pattern

Generates an array containing a sinc pattern. If the **Sinc Pattern** is represented by the sequence Y, then the pattern is generated according to the following formula:

$$y_i = a\,\text{sinc}(i\Delta t - d),$$

$$\text{for } i = 0, 1, 2, \ldots, n-1,$$

$$\text{where } \text{sinc}(x) = \frac{\sin(\pi x)}{\pi x},$$

a is the **amplitude**,
Δt is the sampling interval **delta t**,
d is the **delay**, and
n is the number of elements in **Sinc Pattern**.

The main lobe of the sinc function, sinc(x), is the part of the sinc curve bounded by the region $-1 \leq x \leq 1$.

When $|x| = 1$, the sinc(x) = 0.0, and the peak value of the sinc function occurs when x = 0. You can show using l'Hôpital's Rule that sinc(0) = 1 and that it is also its peak value. Thus, the main lobe is the region of the sinc curve encompassed by the first set of zeros to the left and the right of its peak value.

amplitude defaults to 1.0.

delay shifts the peak value within the **Sinc Pattern** as the VI generates the pattern. This condition is determined from the preceding formula and occurs when $i\Delta t = d$. **delay** defaults to 0.0.

delta t is the sampling interval. It is a floating-point number inversely proportional to the width of the main sinc lobe. That is, the smaller the sampling interval, the wider the main lobe, and the larger the sampling interval, the smaller the main lobe. Notice that when **delta t** is 1 and d is an integer value, the VI sets **Sinc Pattern** to zero except at the point where i = d, at which point the value is equal to **amplitude**. The recommended range of values for the sampling interval is 0 < **delta t** ≤ 1 . **delta t** must be greater than 0.0. If **delta t** is less than or equal to zero, the VI returns an error. **delta t** defaults to 1.0.

Sinc Pattern in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output sinc pattern.

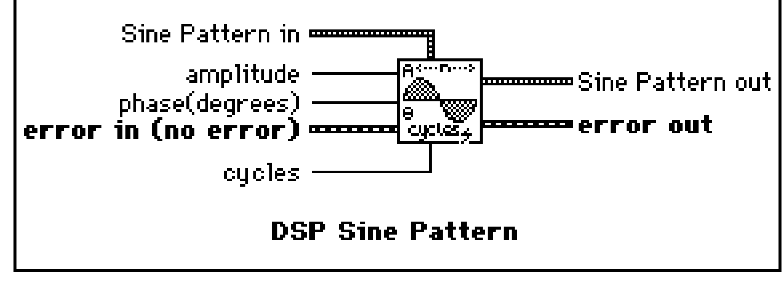
Sinc Pattern out is a DSP Handle Cluster that is identical to the **Sinc Pattern in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest sinc pattern that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Sine Pattern

Generates an array containing a sinusoidal pattern. If the **Sine Pattern** is represented by the sequence Y, the pattern is generated according to the following formula:



$y_i = a \sin(x_i)$,

for $i = 0, 1, 2, \ldots, n\text{-}1$,

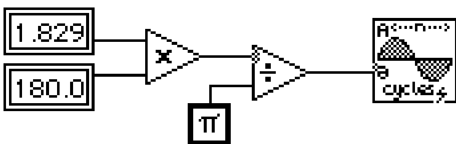
where $x_i = \frac{2\pi i k}{n} + \frac{\pi \phi_o}{180}$,

a is the **amplitude**,
k is the number of **cycles** in the pattern,
ϕ_o is the initial **phase** in degrees, and
n is the number of elements in **Sine Pattern**.

amplitude defaults to 1.0.

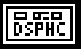
phase defaults to 0.0.

Note: **phase** must be in degrees rather than radians. If **phase** is in radians, make sure you convert it to degrees, as shown in the following figure, before using the Sine Pattern VI.



cycles defaults to 1.0.

Note: Because **cycles** is a floating-point number, fractional cycles are possible for the **Sine Pattern**. Furthermore, setting **cycles** to a negative number does *not* generate an error condition because it is mathematically correct and useful to consider negative frequencies in Fourier and spectral analysis.



Sine Pattern in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output sine pattern.

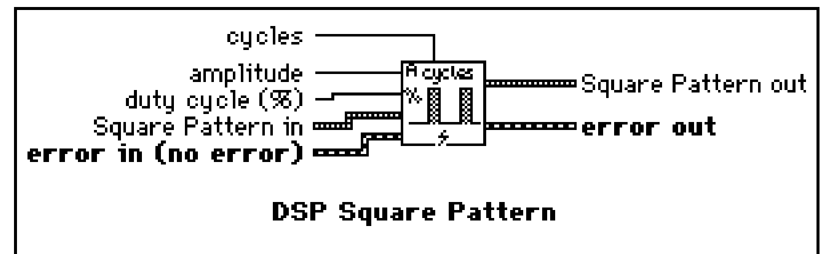
Sine Pattern out is a DSP Handle Cluster that is identical to **Sine Pattern in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest sine pattern that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Square Pattern

Generates an array containing a square pattern. If the **Square Pattern** is represented by the sequence X, then the pattern is generated according to the following formula:



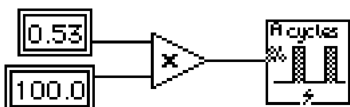
$$x_i = \begin{cases} a & \text{if } 0 \leq \text{remainder}\left(\frac{i}{T}\right) < 0.01\,dT \\ 0.0 & \text{elsewhere} \end{cases} \quad \text{for } i = 0, 1, \ldots, n-1\,,$$

where $T = \frac{n}{\textbf{cycles}}$ is the time period of one cycle of the **Square Pattern**,
a is the **amplitude**,
d is the **duty cycle** in percent, and
n is the number of elements in **Square Pattern**.

amplitude defaults to 1.0.

duty cycle must be greater than or equal to 0 and less than or equal to 100. If **duty cycle** is less than zero or greater than 100, the VI returns an error. **duty cycle** defaults to 50.0.

Note: **duty cycle** must be a percentage. Make sure you convert the fractions of a cycle into percentages, as shown below, before using the DSP Square Pattern VI in a block diagram.

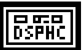


Special Cases: Two special cases can occur when the **duty cycle** assumes its extreme values of 0 and 100. The VI sets **Square Pattern** to zero:

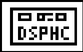
duty cycle = 0 or 100 ⇒ X = 0.

cycles must be greater than 0. If **cycles** is less than or equal to zero, the VI returns an error. **cycles** defaults to 1.0.

Note: Because **cycles** is a floating-point number, fractional cycles of the **Square Pattern** are permitted.



Square Pattern in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output square pattern.

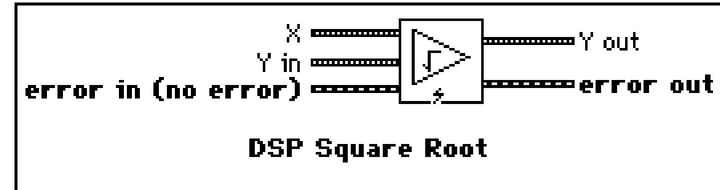


Square Pattern out is a DSP Handle Cluster that is identical to **Square Pattern in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest square pattern that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Square Root



Find a square root estimate of the absolute value of each element of the input array **X .** The ith element of the output array **Y** is obtained using the following formula:

$$Y(i) = \sqrt{|X(i)|} \qquad \text{for } i = 0, 1, 2, \ldots, n-1 ,$$

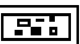
where n is the number of elements in **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the square root results of X(i).

Y out is a DSP Handle Cluster that is identical to **Y in**, but with the square root results already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.



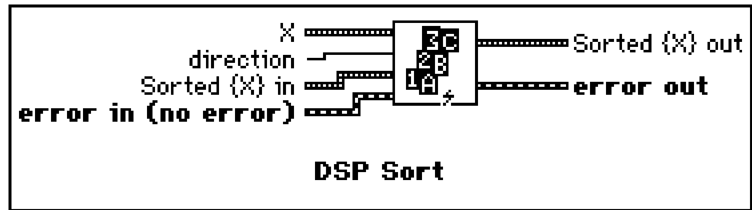
error out contains the error information for this call.

The operation can be performed in place; that is, the input and output arrays can be the same DSP Handle Cluster.

If the input value is negative, it calculates the square root of the absolute value of that number.

DSP Sort

Sort the input array X in ascending or descending order.

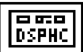


X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

direction is the direction to sort:
0: ascending.
1: descending.

direction defaults to ascending**.**

Sorted {X} in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output array of sorted {X}.



Sorted {X} out is a DSP Handle Cluster that is identical to **Sorted {X} in**, but with the sorted {X} already stored in the memory buffer on the DSP board.

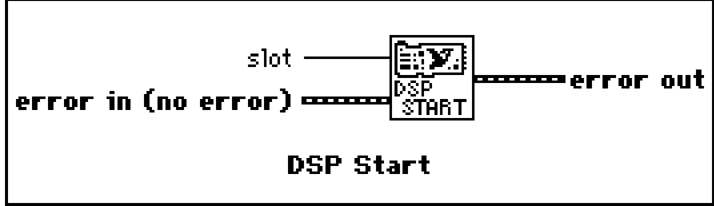
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

The operation can be performed in place; that is, the input and output arrays can be the same DSP Handle Cluster.

DSP Start

Enables the DSP board to run. Use DSP Start with the DSP Load and DSP Reset VIs after downloading a custom application.



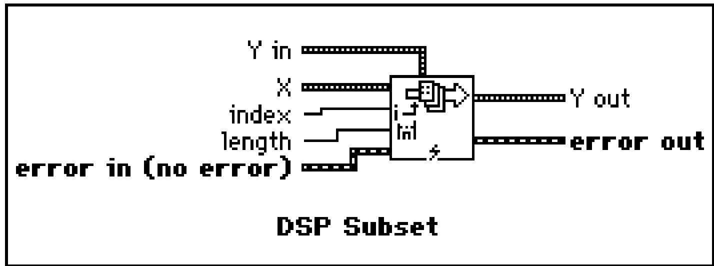
slot is the board ID number. **slot** defaults to 3.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Subset

Extracts a subset of array **X** of **length** beginning at **index** and stores it in array **Y**.



X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

index defaults to 0.

length is the length of the output subset. **length** defaults to 1.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the subset output array.

Y out is a DSP Handle Cluster that is identical to **Y in**, but with the subset array already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

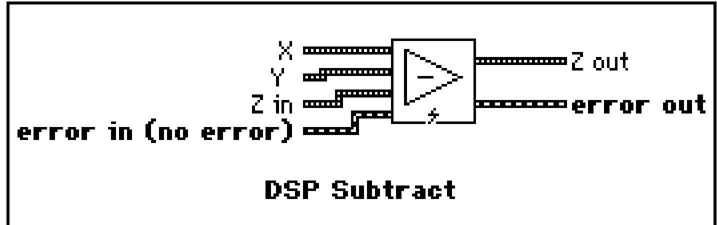
This VI is particularly useful when performing onboard processing with large arrays, with **index** different from zero, or **Y** in a memory space different from **X**. If **index** is less than zero, then the first element of the subset is the first element of the input array.

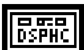
DSP Subtract

Subtract array **Y** from array **X.** The ith element of the output array **Z** is obtained using the following formula:

$$Z(i) = X(i) - Y(i). \quad \text{for } i = 0, 1, 2, \ldots, n-1,$$

where n is the smaller number of elements in **X** and **Y**.




X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

Y is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array Y.

Z in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of X(i) - Y(i).

Z out is a DSP Handle Cluster that is identical to **Z in**, but with the subtracted results already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

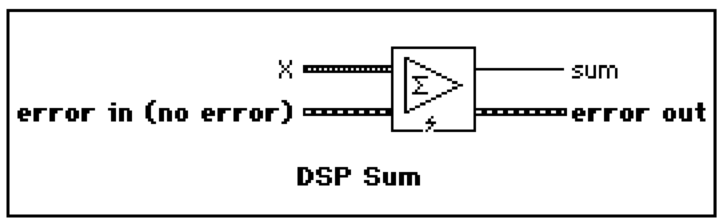
The operation can be performed in place; that is, the input and output arrays can be the same DSP Handle Cluster.

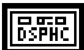
DSP Sum

Find the sum of the elements of the input array **X.** The sum of the elements is obtained using the following formula:

$$\textbf{sum} = \sum_{i=0}^{n-1} X(i)$$

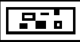
where n is the number of elements in **X**.




X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array X.

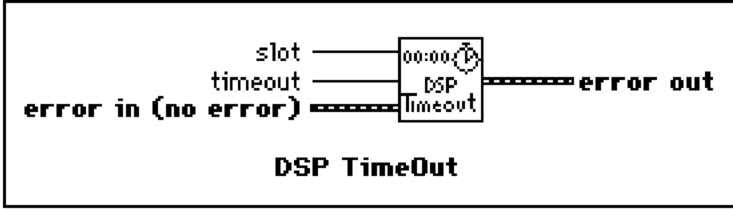
sum is the sum of the elements in **X.**

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.


error out contains the error information for this call.

DSP TimeOut

Selects the timeout limit in seconds to wait for a function on DSP board to complete execution.



The default timeout setting at startup and after a DSP Reset call is 10 s.

All subsequent calls from the VI to the onboard functions on the specified board return timeout errors if function execution on the board calling the VI is not completed in the timeout limit.

I32 **slot** is the board ID number. **slot** defaults to 3.

SGL **timeout** is the timeout in seconds. **timeout** defaults to 10.0 s.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Triangle Pattern

Generates an array containing a triangle pattern. If the **Triangle Pattern** is represented by the sequence Y, the pattern is generated according to the following formula:

Triangle Pattern in
amplitude
delay
delta t
error in (no error)
width
Triangle Pattern out
error out

DSP Triangle Pattern

$$y_i = a \, \text{tri}(x_i),$$

$$\text{for } i = 0, 1, 2, \ldots, n-1,$$

where

$$x_i = \frac{i\Delta t - d}{w}, \text{ and}$$

$$\text{tri}(x) = \begin{cases} 1 - |x| & \text{if } |x| \le 1 \\ 0 & \text{elsewhere} \end{cases}$$

a is the **amplitude**,
d is the **delay**,
w is the **width**, and
n is the number of elements in **Triangle Pattern**.

amplitude is the value of the waveform at the peak. **amplitude** defaults to 1.0.

delay is the distance in seconds between the beginning of the pattern and the peak. **delay** shifts the peak value within **Triangle Pattern**. **delay** defaults to 0.0.

delta t is the duration of the pattern in seconds, or the sampling interval, and must be greater than 0.0 to avoid undefined arguments. If **delta t** is less than or equal to zero, the VI returns an error. **delta t** defaults to 1.0.

width is the distance in seconds between the peak and end of the pattern. In other words, **width** sets the width from the peak value to the first zero value in the pattern. Thus the actual duration of the pattern is 2**width** (twice the value of **width**). **width** must be greater than 0.0 to avoid undefined arguments. If **width** is less than or equal to zero, the VI returns an error. **width** defaults to 1.0.

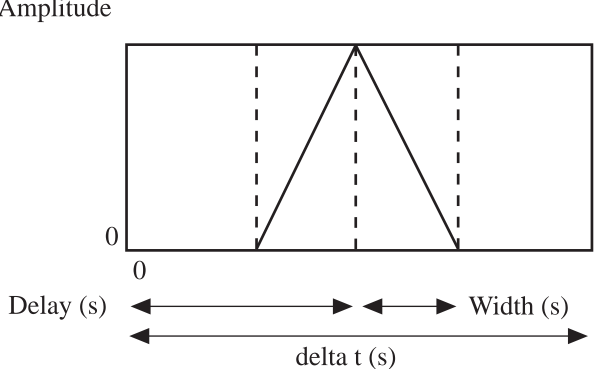
Triangle Pattern in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output triangular pattern.

Triangle Pattern out is a DSP Handle Cluster that is identical to **Triangle Pattern in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest triangle pattern that can be generated depends upon the amount of memory on your DSP board.

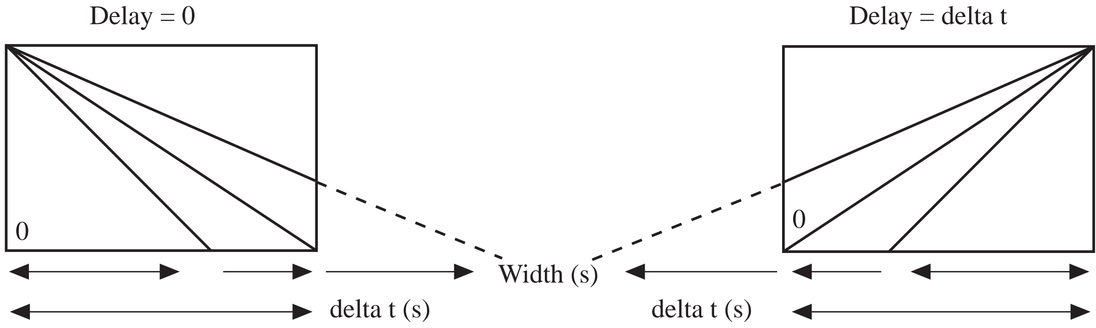
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

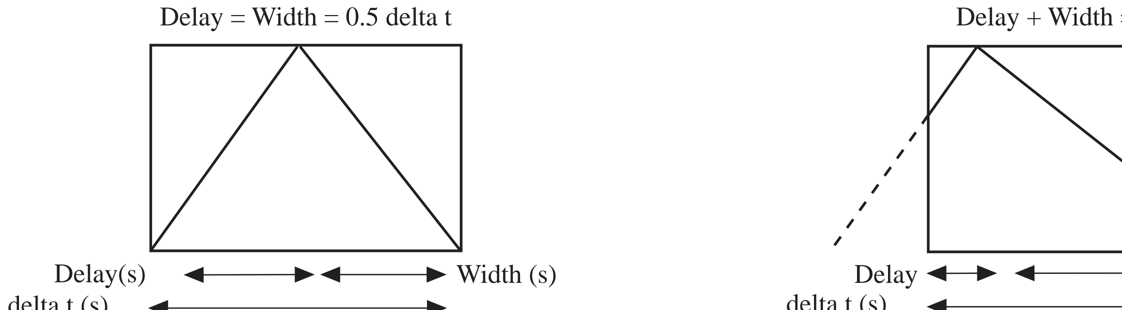
You use this VI to create patterns based on an isosceles triangle. The following figure shows how the VI parameters relate to the generated pattern.



The following figure illustrates how the pattern can vary with different values for the parameters.

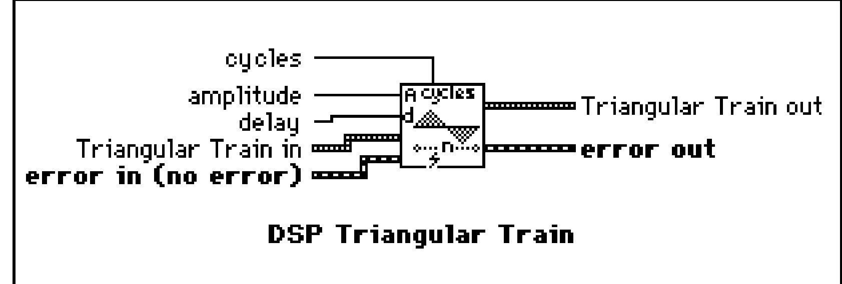


Notice that when the **Delay** is 0, the signal is a ramp with a slope of -**Amplitude**/**Width**, and when **Delay** equals **delta t**, the signal is a ramp with a slope of **Amplitude**/**Width**. Remember that the **Width** parameter is actually the width of the *ramp* rather than the width of the triangle. **delta t** is the duration of the pattern in seconds. In the previous and following illustrations, the box shows the pattern limits.



DSP Triangular Train

Generates a train of triangular pattern crossing value zero at **delay** with positive slope. If the triangular train is represented by the sequence Y, the pattern is generated according to the following formula:



$$y_i = \begin{cases} \frac{4a}{T}k & \text{if } 0 \le k \le \frac{T}{4} \\ \frac{-4a}{T}\left(k - \frac{T}{2}\right) & \text{if } \frac{T}{4} \le k \le \frac{3T}{4} \\ \frac{4a}{T}(k - T) & \text{if } \frac{3T}{4} \le k < T \end{cases} \quad \text{for } i = 0, 1, 2, \ldots, n-1$$

where $k = (i - d)$ modulo T,

$T = \frac{n}{\textbf{cycles}}$,

n is the number of elements in the triangular train, and

d is the delay.

SGL **amplitude** defaults to 1.0.

I32 **delay** defaults to 0.

SGL **cycle** defaults to 1.0.

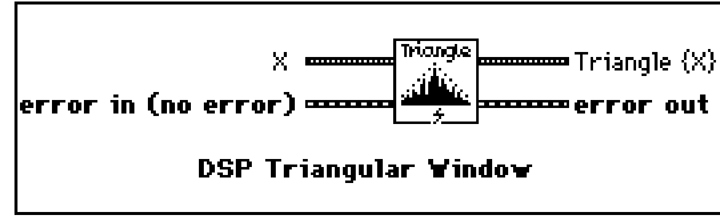
DSPHC **Triangular Train in** is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output triangular train.

DSPHC **Triangular Train out** is a DSP Handle Cluster that is identical to **Triangular Train in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest triangular train that can be generated depends upon the amount of memory on your DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Triangular Window



Applies a triangular window to the input sequence **X**. If Y represents the output sequence **Triangle{X}**, the elements of Y are obtained from the formula:

$$y_i = x_i \,\mathrm{tri}(w)$$

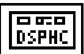
for $i = 0, 1, 2, \ldots, n-1$,

$$w = \frac{2i-n}{n},$$

where $\mathrm{tri}(w) = 1 - |w|$, and
n is the number of elements in **X**.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array **X** is overwritten by the output **Triangle {X}.**



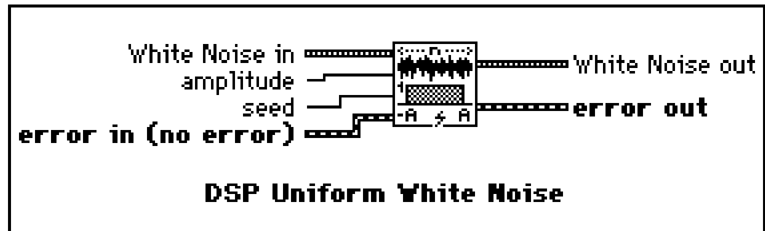
Triangle {X} is a DSP Handle Cluster that is identical to **X**, but with the results of **Triangle {X}** already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Uniform White Noise

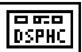
Generates a uniformly distributed pseudorandom pattern whose values are in the range [-a:a], where a is the absolute value of **amplitude**. The pseudorandom sequence is generated using a modified version of the Very-Long-Cycle random number generator algorithm.



amplitude defaults to 1.0.

seed < 65536.0. If **seed** > 0.0, the generated noise will be the same in repeated invocations if the **seed** value does not change. If **seed** ≤ 0.0, the VI generates a random value to use as the **seed**, and the noise differs in repeated invocations although the value in **seed** does not change. **seed** defaults to 0.0.

White Noise in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the output white noise.

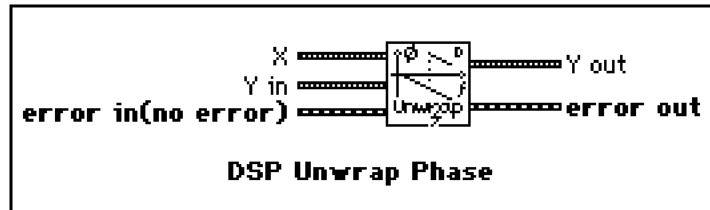
 **White Noise out** is a DSP Handle Cluster that is identical to **White Noise in**, but with the generated pattern already stored in the memory buffer on the DSP board. The largest white noise signal that can be generated depends upon the amount of memory on your DSP board.

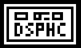
error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP Unwrap Phase

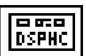
Unwraps the **Phase** array by eliminating discontinuities whose absolute values exceed π.





X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Y in is a DSP Handle Cluster that indicates the memory buffer on the DSP board that will contain the results of the integration of X.



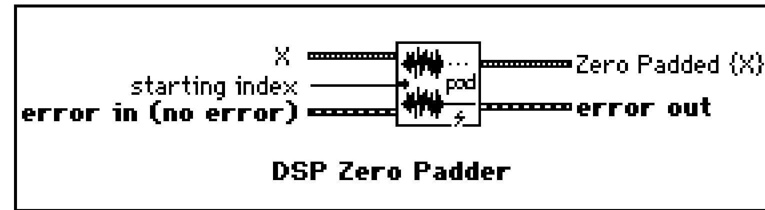
Y out is a DSP Handle Cluster that is identical to **Y in**, but with the unwrapped output array already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

Both the input array and the output array are in radians. The operation can be performed in place; that is, the input X and the output Y can be the same DSP Handle Cluster.

DSP Zero Padder



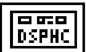
Pads the input array with zero from **starting index** to the end of the input array.

This VI is useful when the size of the acquired data buffers is not a power of two and you want to take advantage of fast processing algorithms in the analysis VIs. These algorithms include Fourier transforms, power spectrum, and fast Hartley transforms, which are extremely efficient for buffer sizes that are a power of two.

X is a DSP Handle Cluster that indicates the memory buffer on the DSP board that contains the input signal array.

Note: The operation is performed in place and the input array X is overwritten by the output **Zero Padded {X}**.

starting index is the index from which the input array is padded with zero. **starting index** defaults to 0.



Zero Padded {X} is a DSP Handle Cluster that is identical to **{X}**, but with the zero-padded array already stored in the memory buffer on the DSP board.

error in (no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

Chapter 1
Introduction to the NI-DSP Interface Utilities

This chapter contains an overview of the NI-DSP Interface Utilities, installation instructions, and explains how to use the NI-DSP Interface Utilities. You should familiarize yourself with the material in this chapter before beginning subsequent chapters in this part.

The National Instruments AT-DSP2200, a high-performance digital signal processing (DSP) plug-in board for the PC, is based on the AT&T WE DSP32C 32-bit, floating-point, digital signal processor. With the NI-DSP Interface Utilities, you can customize the DSP Library that resides on the board by adding or deleting functions to and from the common object file format (COFF) file that constitutes the onboard-resident software. This COFF file is made of the Kernel, memory management routines, execution control routines, data communication routines, interrupt handling routines for data acquisition, and a set of more than 60 analysis functions. Throughout the chapters in this part, the COFF file is referred to as the DSP Library. By customizing the DSP Library, you can change the number of analysis functions that are included. The other parts of the DSP Library always remain unchanged.

Note: To customize the DSP Library that resides on the board, you need to have the Developer ToolKit, available through National Instruments, which contains an AT&T C compiler, assembler, linker, and documentation.

Overview of the NI-DSP Interface Utilities

The NI-DSP Interface Utilities are on the NI-DSP for LabVIEW for Windows distribution disks.

Unless otherwise stated, all references to directories in this part of the manual are to subdirectories under the path you specified during installation of the NI-DSP Interface Utilities. If you chose the path `D:\DEVEL`, then the installer will create the directory `DEVEL` and copy all files and directories to it. Figure 1-1 shows directory structure created by `SETUP`.

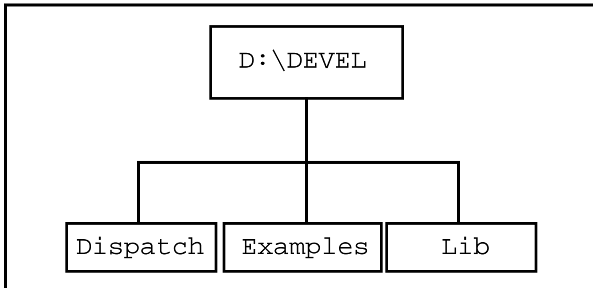


Figure 1-1. NI-DSP for DOS Directory Structure

The `Dispatch` directory contains the utilities necessary for creating a custom DSP Library dispatcher. The `Dispatch` directory contains the `Dispatch` application and the files `NIDSP.fnc`, `dispatch.s.`, and `NIESSEN.fnc`.

`Dispatch` is a PC application that uses the files `NIDSP.fnc` and `NIESSEN.fnc` and generates the WE DSP32C assembly code needed to properly dispatch to a custom DSP Library based on the grouped function names.

The `Examples` directory contains the files used in Part 4, Chapter 2, *Getting Started with the NI-DSP Interface Utilities*, to build a custom DSP Library.

The `LIB` directory contains `LIBLVDSP.a`, the archived library of object modules containing all of the analysis functions and Kernel routines used in building the downloadable DSP Library files, as well as the two DSP Library files `LV2200S.out` and `LV2200.out`.

Installing the NI-DSP Interface Utilities

For instructions on installing the NI-DSP Interface Utilities, refer to Part 1, *Getting Started with NI-DSP*, under the section titled *Installing NI-DSP for LabVIEW for Windows*.

For instructions on installing the AT&T WE DSP32C software, refer to the *WE DSP32C Support Software Library User Manual* and other related documentation.

Using the NI-DSP Interface Utilities

Use the NI-DSP Interface Utilities to customize the DSP Library and execute your own functions on the DSP board. Figure 1-2 shows a diagram of the interface layers used to access functions in your custom library.

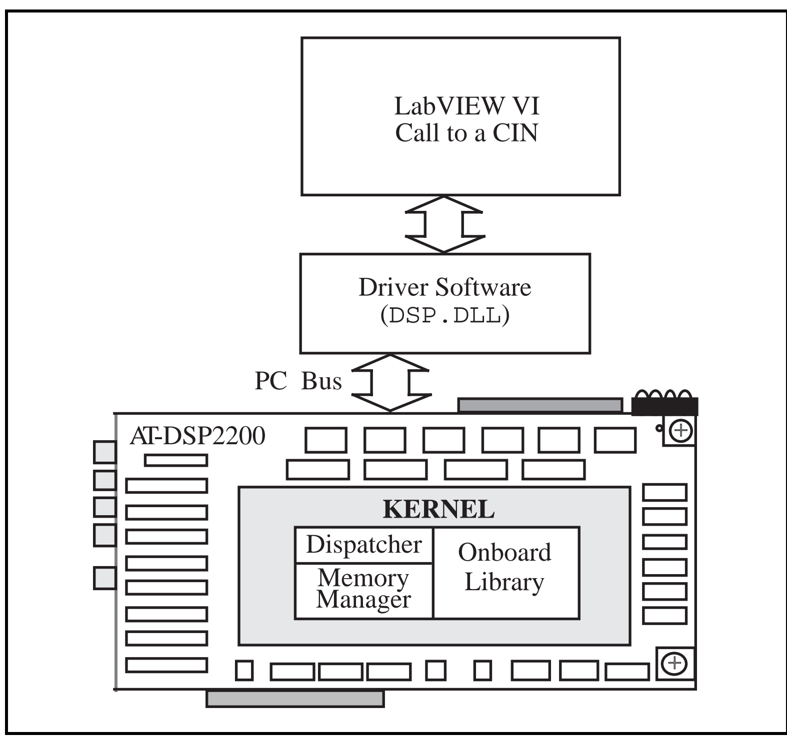


Figure 1-2. Interface Layers to Onboard Functions

When your library is downloaded and a LabVIEW interface to your function has been created, you can execute a function in your onboard custom library by making a call to the interface function in your program from LabVIEW. This call passes control to the interface software which calls the NI-DSP driver. The NI-DSP driver communicates with the DSP Library to pass parameters, indicates which onboard function to execute using the dispatcher, and returns results to LabVIEW.

Chapter 2
Getting Started with the NI-DSP Interface Utilities

This chapter contains a step-by-step example for building a custom DSP Library, creating a LabVIEW interface to a custom function, and executing the custom function from the LabVIEW environment. The chapter demonstrates this concept with an example of how to add a custom function. The custom function finds the maximum and minimum values in the input array, as well as the respective indices of the occurrence of the maximum and minimum value. This example custom function also returns an output array with the input array sorted.

Creating Your Custom NI-DSP Library

To build your own custom library, follow these steps:

1. Create your source code of C functions (or assembly functions).
2. Compile and/or assemble source code.
3. Add your object filenames to a linker file (`ifile`).
4. Add your new function names to a library function list file.
5. Run the `Dispatch` application to generate an assembly dispatch file.
6. Compile, assemble, and link your custom library.

These steps are detailed in the following pages.

Note: You can use the source files included in the `Examples` directory instead of creating the files in this example.

1. Create Your Source Code of C Functions

The first step in building a custom DSP Library is to create your WE DSP32C C-callable custom function(s). The `gmaxmin.c` example file in your `examples` directory contains the source code.

GMaxMin.c Example:

```
/* DSP_GMaxMin(z,y,max,min,n,imax,imin)
* This function returns the maximum and minimum of array z, as well as the
* respective indices of the occurrence of the maximum and minimum.  The output
* array y holds the input array sorted.
*/

#include "atdsp.h"
short DSP_GMaxMin(z,y,max,min,n,imax,imin)
register float *z,*max,*min;
float *y;
```

```
long *n,*imin,*imax;
{
long i,j;
float *x, small;
float localmin ,localmax;

if ((*n)<= 0) return(0);
x = z;
localmin = *x;
localmax = localmin;
*imax = 0;
*imin = 0;
for (i=0;i< *n;i++,*x++) {
    if (localmax < *x) {
         localmax = *x;
         *imax = i;
         }

    if (localmin > *x) {
         localmin = *x;
         *imin = i;
         }
    }

*min = localmin;
*max = localmax;

for(i=0; i< *n; i++) y[i] = z[i];
for(i=0; i< *n; i++){
   for(j=i+1; j< *n; j++)
      if(y[i] > y[j]) {
         small = y[j];
         y[j] =  y[i];
         y[i] = small;
      }
  }
return(noError);
}
```

Note: The file `ATDSP.h` is the header file containing all of the error codes used by the DSP Library. You can find it in the `LIB` directory. To efficiently and correctly compile WE DSP32C code that uses error codes defined in `ATDSP.h`, you may want to copy this header file to the `Include` directory of your WE DSP32C tools. Otherwise, you should specify the correct path of `ATDSP.h` in your program.

Guidelines for the Custom Functions

When adding functions to build your custom DSP Library, follow these guidelines:

- Pass all parameters by address–All input, output, and input/output parameters, whether arrays or scalars, must be passed by address (pointer). For instance, in the example `gmaxmin.c`, although the length **n** of an input array is only an input scalar, it is passed by address.
- You must pass parameters in a certain order–Pass all pointers to arrays, then pass all of the pointers to 32-bit floating-point scalars, and then pass all of the pointers to 32-bit long integer scalars. In `gmaxmin.c`, the parameters are passed in the following order–**z** (input array), **y** (output array), **max** (output floating-point scalar), **min** (output floating-point scalar), **n** (input integer scalar), **imax** (output integer scalar), and **imin** (output integer scalar).

- Parameters must be 32-bit floating-point, 32-bit integer scalars, or pointers to arrays of data (any type)– Supported parameter types are 32-bit floating-point scalars and multidimensional arrays, and 32-bit integer scalars and multidimensional arrays.

- Return a 16-bit short integer error code–Every function should return an integer error code. A list of error codes that the existing DSP Library returns is given in Appendix A, *Error Codes*. If any of these error codes are used, `ATDSP.h` should be included at the beginning of your source code. A function that completes with no error should return `noError` or zero.
- Use the memory management routines–Perform any dynamic onboard memory allocation using the functions `Alloc_Mem` and `Free_Mem`, described in the next section.

DSP Board Memory Management

To write your own customized DSP routines that become integrated with the DSP Library, you need some lower level memory management routines. Two onboard calls are included with the DSP Library to dynamically allocate and deallocate onboard memory–`Alloc_Mem` and `Free_Mem`. Use these functions instead of the standard C routines `malloc` and `free`.

Call the function `Alloc_Mem` whenever your routines require memory space. `Alloc_Mem` attempts to allocate memory of the requested size (in bytes) in any available memory bank. If the requested buffer size (in bytes) is not a multiple of four, the allocation routines ensure alignment to the nearest radix 4 boundary >= **NumBytes**. This is to guarantee memory alignment on 4-byte addresses. For example, if you request the allocation of a buffer of size 1,021, 1,022, 1,023 or 1,024 bytes, then the allocation routines allocate 1,024 bytes in all four cases. The allocation routines first attempt to allocate the buffer in on-chip memory (all DSP boards have 4 kilobytes of on-chip memory). If the routine fails, the allocation routine tries to allocate the buffer in onboard memory (DSP boards have between 256 kilobytes and 1.5 megabytes of onboard memory). The function and input parameters for `Alloc_Mem` are defined as follows:

```
void *   Alloc_Mem(size)
long     size;
```

where `size` is the requested block size in bytes.

`Alloc_Mem` returns a pointer to the allocated buffer or returns `NULL` if the allocation failed.

Call the function `Free_Mem` when you deallocate memory buffers. Use this function instead of the standard C function `free`. The function and input parameters for `Free_Mem` are defined as follows:

```
short    Free_Mem(ptr)
long     ptr;
```

where `ptr` is the address of the buffer to be deallocated.

`Free_Mem` deallocates the buffer pointed to by `ptr` if that buffer was allocated previously using `Alloc_Mem`. `Free_Mem` returns `NULL` if deallocation succeeded and returns an error code if deallocation did not succeed.

Each of the NI-DSP Analysis VIs call a function that is part of the DSP Library that resides on the board. Your own custom functions can call any of the DSP Library functions included in the `NIDSP.fnc` file described later in this chapter. The DSP functions that you can call from your own functions are prototyped in Part 4, Chapter 3, *DSP Board Function Overview*.

Note for Assembly Language Programmers: If you are using WE DSP32C assembly language, each function must accept and return C-style parameters and preserve all registers used in the WE DSP32C environment. For example, the function parameters are pushed onto the stack from right to left and the return value is placed in Register r1. Refer to the *WE DSP32C C Language Compiler Library Reference Manual* for details.

2. Compile and/or Assemble Source Code

Compile all new C source files and assemble all new assembly source files using the WE DSP32C C compiler and assembler. Remember to use only `float` and `long` data types for scalars. If you use any of the error codes from Appendix A, *Error Codes*, include `ATDSP.h` at the beginning of your source code. A function that completes with no error should return `noError` or zero. The `ATDSP.h` file is in the `Lib` directory. After you have installed the WE DSP32C tools, you may want to copy `ATDSP.h` to the `Include` directory in the root directory for those tools.

In this example, use `d3cc -c gmaxmin.c` to generate WE DSP32C object code.

3. Add Your Object Filenames to a Linker File (ifile)

The next step is to modify the file `NIDSPLNK` to add the names of your custom object files to the list of modules to be linked into your custom DSP Library. `NIDSPLNK` is a source file containing link editor directives for the WE DSP32C tools. This modification is illustrated in Figure 2-1.

```
essent.o
stackbld.o

/*** Add all your file names after this comment line ***/
gmaxmin.o

/*** Add all your file names above this comment line ***/
-lLVDSP
```

Figure 2-1. Linker File `NIDSPLNK`

Copy the `NIDSPLNK` file from the `LIB` directory to your working directory and then modify the copy. This ensures that you maintain the original `ifile`. Also, you need to copy the files `essent.o`, `dsp_glob.o`, and `stackbld.s`, from the `LIB` directory to your working directory.

Note: Add your function(s) name(s) to this file between the comments instructing you to do so. Do not modify this file in any other way.

4. Add Your New Function Names to a Library Function List File

Copy the files `NIDSP.fnc` and `NIESSEN.fnc` from the `Dispatch` directory to your working directory.

In Figure 2-2, the file `NIDSP.fnc` shows the addition of the function `DSP_GMaxMin` to the library function list. This is done by editing the file `NIDSP.fnc` in the current directory. Add this function to a new group, `My Functions`, at the end of this file.

```
  ** My Functions
DSP_GMaxMin
```

Figure 2-2. Library Function List File `NIDSP.fnc`

Notice that functions accepted by the `Dispatch` application should have acceptable C syntax, that is, names may contain letters, numbers, and the underscore character but must start with a letter or underscore. A function name can contain up to 31 characters. You must divide your functions into groups with a maximum of 64 (40 hex) functions per group. The `Dispatch` application returns an error message if function names are not acceptable or if you have more than 64 functions per group. You may have up to 24 groups of 64 functions each. The `Dispatch` application assigns a unique ID number to each function depending on how the functions are grouped. Groups start at zero and are consecutive (Group 0, 1, 2, ...). The ID number of the first function of a group is the group number multiplied by 64. The function IDs are then consecutive within that group. The function ID, therefore, has group information as well as index information for functions within each group. This information is necessary for the dispatcher to execute a particular function efficiently based on the function ID. The first group of functions are the Essential Functions group. The Essential Functions are always linked into the DSP Library. These functions are specified in `NIESSEN.fnc`. Do *not* modify this file. In the library function list file, `NIDSP.fnc`, group beginnings are marked by two asterisks (**) followed by a group name. Figure 2-3 shows a typical section of `NIDSP.fnc`.

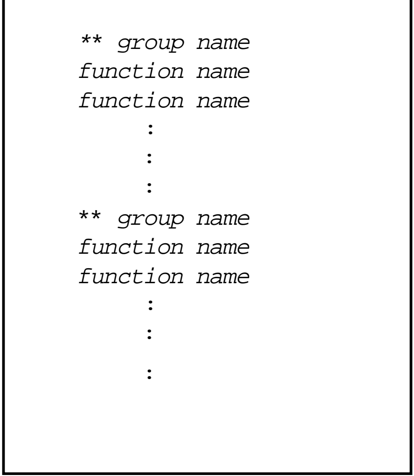


Figure 2-3. Typical Section of `NIDSP.fnc`

Customizing the DSP Library by Deleting Functions

Another way to customize the DSP Library is to delete functions from `NIDSP.fnc`. This decreases the size of the DSP Library because the object code for those deleted functions are not included. To customize the DSP Library by deleting functions, follow the instructions in this chapter, but delete functions rather than add functions. Follow these guidelines when deleting functions from the DSP Library:

- When modifying the `NIDSP.fnc`, replace the name of each function you want to delete with the function name `DSP_NOP`. This replacement ensures that the remaining functions in the library are addressable by the dispatcher using the same function ID. The Interface Library `DSP.DLL` is built using the organization of the original library function list file, and thus function IDs are hard coded into the Interface Library. If you reorganize the original functions and groups without replacing the function name with `DSP_NOP`, the interface to the function calls a different DSP Library function, which causes the DSP Library to fail. For example, if you had a function group with 10 functions and you wanted to delete the first and third functions of that group, if you replace their names with `DSP_NOP`, you can ensure that all of the other functions in the group maintain the same function ID. This ensures that the interface code for those functions remain valid. If you do not follow these guidelines, functions become disorganized, you could call the wrong function, and an unexpected error could happen.
- *Never* delete any of the function group names that form the original DSP Library that you received with the NI-DSP software package. The function group names ensure that the remaining functions in the DSP Library have the same function IDs. If you want to delete all functions of a group, replace each function name of that group in the `NIDSP.fnc` with `DSP_NOP`.

5. Run the Build Dispatch Application to Generate an Assembly Dispatch File

The next step is to generate an assembly dispatch file. In the current directory, run `dispatch.exe`. You can find the executable file under the `dispatch` directory. The `Dispatch` application automatically generates the WE DSP32C assembly code file `dispatch.s` and a header file `dspfncs.h` in your current directory.

The `dspfncs.h` file has define statements assigning function codes for all the functions in the library. Each function code is used by the LabVIEW interface when calling the DSP Library. The onboard Kernel uses these function codes (passed to it by the interface code) to execute the proper functions. Look at the file `dspfncs.h` to see the resulting function IDs assigned to your custom functions. The function codes for `DSP_GMaxMin` is 448. Remember this code. You will use the code later to build your LabVIEW interface VI. Figure 2-4 shows a section of `dspfncs.h` corresponding to the section given from `NIDSP.fnc`. Notice that in this example, `Signals` was the seventh group (Group 6) listed in `NIDSP.fnc` and hence the function IDs.

```
/** Signals
 **/
#define DSP_Sine_NUM   384
#define DSP_Square_NUM 385
#define DSP_Sinc_NUM   386
    :
    :
    :
    :
```

Figure 2-4. Signals Group Section in `dspfncs.h`

The file `dispatch.s` is an assembly file that the Kernel uses to execute the proper function. This file has information about the number of groups and the number of functions per group that are linked into the software that resides on the board. The `Dispatch` application uses the files `NIDSP.fnc` and `NIESSEN.fnc` to create `dispatch.s`. Figure 2-5 shows a section of `dispatch.s` corresponding to the section given from `NIDSP.fnc`. Notice that in this example, `Signals` was the seventh group (Group 6) listed in `NIDSP.fnc`. Group zero is the group of Essential Functions.

```
/** Signals
 **/
call DSP_Sine(r18)
nop
goto FNC_CALL_END
nop
call DSP_Square(r18)
nop
goto FNC_CALL_END
nop
call DSP_Sinc(r18)
nop
goto FNC_CALL_END
nop:
    :
    :
    :
```

Figure 2-5. Signals Group Section in `dispatch.s`

6. Compile, Assemble, and Link Your Custom Library

The last step in building a custom DSP Library is to compile, assemble, and link your custom library using the WE DSP32C tools. You must have installed those tools as described in the *WE DSP32C Support Software Library User Manual*. You must copy the file LIBLVDSP.a from the LIB directory in the stand-alone root directory to the LIB directory of the WE DSP32C tools as set by the environment variable recommended in the *WE DSP32C Support Software Library User Manual*. Table 2-1 lists the files you need to build the custom NI-DSP example.

Table 2-1. Files Required to Build the Custom DSP Library Example

File	Description
makelib.bat	Batch file to build the custom library
gmaxmin.c	C source file
dspfncs.h	C include file generated by Dispatch
dispatch.s	Assembly file generated by Dispatch
NIDSPLNK	Modified linker command file
LIBLVDSP.a*	Archived library of object modules for the DSP Library functions for LabVIEW
libc32c.a*	AT&T C library
libap32c.a*	AT&T Application library
libm32c.a*	AT&T Math library found in the LIB directory in the path set by the environment variable
* All these archives must be in the LIB directory in your WE DSP32C tools directory.	

Copy the file makelib.bat from the LIB directory. This is a batch file that rebuilds the DSP Library file to reflect the customization and changes needed. You must enter in the memory map file names to this .bat file to use in linking the DSP Library file, as well as the name of the output file for the DSP Library file. The two memory map files used for linking the DSP Library files that come with NI-DSP, dsp64.map and dsp.map, are located in the LIB directory. Copy the memory map file that you will use to your current directory. The memory map files are used when linking the DSP Library files LV2200S.out and LV2200.out, respectively. You can rename these files and move them to other directories. If you want the driver to load this new DSP Library file, make sure you run WDAQCONF as described in the section titled *Board Configuration* in Part 1, *Getting Started with NI-DSP*.

You should use one of the memory map files, dsp64.map or dsp.map, as input to makelib.bat. If the size of your library cannot be linked using the section sizes specified in these memory map files, you must edit these files accordingly. Refer to the *WE DSP32C Support Software Library User Manual* for more information.

To begin rebuilding the customized DSP Library, type the following command:

```
makelib memmap libfile
```

where

memmap is the name of the memory map file used
libfile is the name of the desired DSP Library file name

For example,

```
makelib dsp64 LV2200S
```

Do not use the extension names .map and .out for the fields *memmap* and *libfile*.

The `makelib.bat` batch file performs the following:

1. `makelib.bat` assembles the file `stackbld.s` in the current directory using the AT&T `d3as.exe` assembler. You need to assemble this file because it includes the file `stackbld.s`, created by the `Dispatch` program, and contains the assembly code that determines which function the DSP board executes. The file `dispatch.s` reflects the organization and content of the DSP Library file as is reflected by the library function list file, `NIDSP.fnc`, in the current directory.

2. `makelib.bat` rebuilds the DSP Library file with name *`libfile.out`* (the name you decided to give to the new DSP Library file). This new DSP Library file always has the Kernel, memory management routines, execution control routines, data communication routines, interrupt handling routines for data acquisition, and the functions listed in the library function list file `NIDSP.fnc.`

3. `makelib.bat` lists all undefined symbols from the DSP Library file. These symbols could be undefined because you did not link all the object modules.

Any other error messages that are encountered are WE DSP32C error messages. Refer to the *WE DSP32C Support Software Library User Manual* for information.

At this point, the WE DSP32C tools have generated a custom DSP Library file.

Creating Your LabVIEW Interface

Now that you have added a function to the DSP Library, you need to be able to call this function from LabVIEW using a VI. The NI-DSP Analysis VIs contains a VI called Custom VI in the Utility folder. You can use the Custom VI to call any function on the DSP board that follows the guidelines listed in the *Guidelines for the Custom Functions* section earlier in this chapter.

To create your LabVIEW interface, follow these steps:

1. Bundle all the input parameters to arrays.

2. Call the Custom VI.

3. Index the output arrays to get the results.

1. Bundle All of the Input Parameters to Arrays

Use the Custom VI in the Utility folder of the DSP2200 folder as the interface for calling custom functions on the DSP board from LabVIEW. You can use three types of data for input/output parameters in the Custom VI. The data types correspond to the different types of parameters that you can use in your custom functions on the DSP board.

Custom functions have three *groups* of parameters–pointers to arrays of data (any type; 32-bit floating point, 32-bit integer, and 16-bit integer), pointers to 32-bit floating-point scalars, or pointers to 32-bit integer scalars. All of the parameters in a group must be the same type. You must arrange the custom function parameters so that the group of all array pointers (if the function has any) are first, followed by the group of all pointers to 32-bit floating-point scalars (if the function has any), followed by the group of all pointers to 32-bit integer scalars. The LabVIEW interface to such a custom function, Custom VI, should reflect the same parameter order.

The number of parameters varies for different functions. To use the Custom VI as the interface for different custom functions, you must bundle the same types of parameters to an array. LabVIEW can check the array sizes in the CIN to determine how many parameters to pass to a custom function on the DSP board from LabVIEW. For this reason, Custom VI uses arrays as input/output parameter types. The number of parameters of a custom function is the sum of the sizes of three arrays of the Custom VI.

The array of DSP Handle Clusters holds all the references to arrays of data used by the custom DSP functions. You should bundle the DSP Handle Clusters together to an array in the order they appear in the first group of parameters in the custom function on the DSP board. For example, if the first three parameters of the custom function are the arrays X, Y, and Z, which correspond to the DSP Handle Clusters in LabVIEW, **hdl1**, **hdl2**, and **hdl3**, respectively, then you must bundle **hdl1**, **hdl2**, and **hdl3** to an array in that order. The same order requirements apply to the array of 32-bit floating-point scalars that holds the LabVIEW parameters corresponding to the second group of the parameters in the custom function. Likewise, the array of 32-bit integer scalars that holds the LabVIEW parameters corresponding to the third and final group of parameters in the custom function must also have the same order requirements as the other arrays.

Bundle all of the parameters of the same type to an array before you call Custom VI. Remember that the order in which you bundle the parameters should correspond to the order of the terminals in the corresponding group of the custom function. The first element in the array representing a group should correspond to the first parameter of the same group in the custom function, and so on.

The Custom VI is the interface from LabVIEW to all of the custom functions on the DSP board. Each parameter of the custom functions on the DSP board require two corresponding terminals in LabVIEW–an input control to reserve the space on the DSP board for the data, and an output indicator to hold the result. All of the output arrays in the Custom VI are internally connected to the corresponding input arrays.

The Custom VI treats all of the parameters in the custom functions as input/output parameters. The parameters that represent outputs of the custom function must also be bundled in LabVIEW in the array that holds all of the other input parameters of the same type. Although the output parameters, whether scalars or arrays, do not contain any valid or useful information on input, they will be overwritten by the return values. The parameters that represent inputs to the custom functions also exist in the corresponding output arrays, although the values of the output arrays are the same values as on input–the custom function will not change the parameter values.

Figure 2-6 shows how to bundle the parameters for `gmaxmin.c`. Allocate two DSP Handle Clusters corresponding to the input array **z** and the output array **y**. The bundling order is as follows–the DSP Handle Cluster representing **z** is first, followed by the DSP Handle Cluster representing **y**. **y** is the output array. The data in this array is not valid as an input, but you must still allocate it first and bundle it with other DSP Handle Clusters that represent input arrays. Notice that although the parameters **max value**, **min value**, **max index**, and **min index** are outputs only in `gmaxmin.c`, you must first bundle the additional four input scalars with the actual input parameters to reserve space in memory for the parameters on output. The initial values are not important. The parameter **n** (the number of elements) is only an input parameter in `gmaxmin.c`, but you will still have an output corresponding to this parameter, although the output and input values are the same.

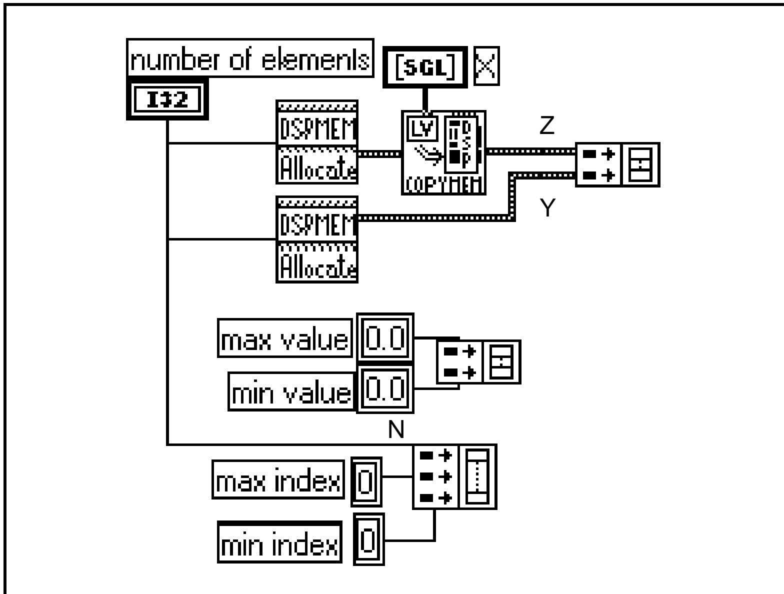


Figure 2-6. How to Bundle Parameters in LabVIEW to Call `gmaxmin.c`

2. Call the Custom VI

After you bundle all of the parameters to arrays, connect each array to the corresponding terminals of the Custom VI. Figure 2-7 shows how to connect to the Custom VI for the `gmaxmin.c`.

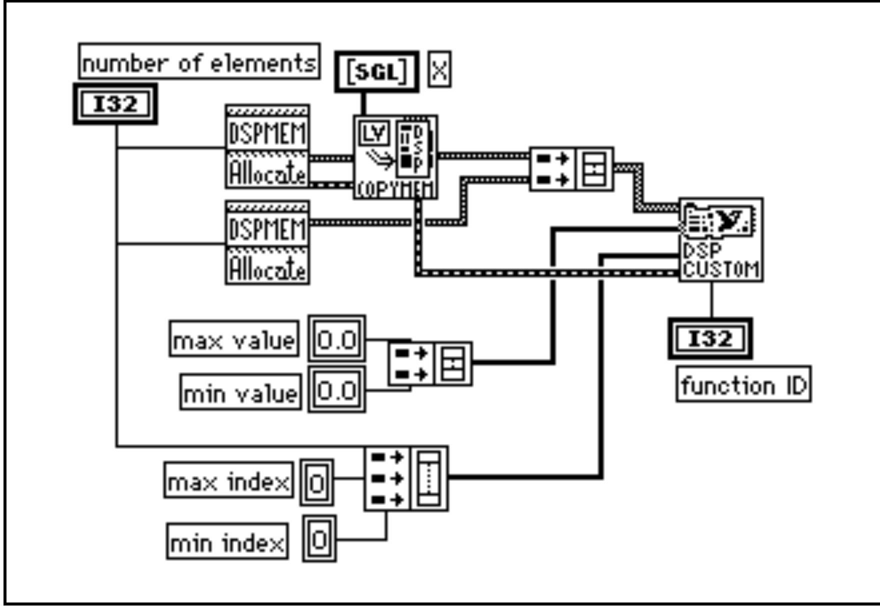


Figure 2-7. How to Connect to Custom VI to Call `gmaxmin.c`

If your custom function does not have a certain type of parameter, leave the corresponding terminal unwired. This is important for passing your parameters correctly from LabVIEW to the DSP board.

The value of the function ID is the custom function ID that you obtain from `dspfncs.h`. This ID determines which custom function on the DSP board the Custom VI calls. To call the correct function on the DSP board, you must supply the appropriate function ID. In this example, the function ID is 448, which you obtained in step 5 of creating your custom NI-DSP Library. Finally, you must indicate the slot number on which to execute the custom VI. In Figure 2-6, the slot parameter terminal is left unwired. This causes all VIs to execute on the default DSP board, which is set to slot 3.

3. Index the Output Arrays to Obtain the Results

All of the output results are in the output arrays of the Custom VI. You can read the output arrays to see the results. If you want to operate on some output parameters, use the method for indexing an array to extract an element from an array. The order of the output parameters in an output array is identical to the order in the corresponding input array.

Figure 2-8 shows how to index the output arrays of the Custom VI to obtain the results of `gmaxmin.c`. The Custom VI leaves the output data buffers on the board, and returns only the DSP Handle Clusters that indicate the locations of these buffers on the board. You must use the Copy Mem(DSP to LV) VI to copy data back to LabVIEW. The Custom VI always copies scalars back to LabVIEW.

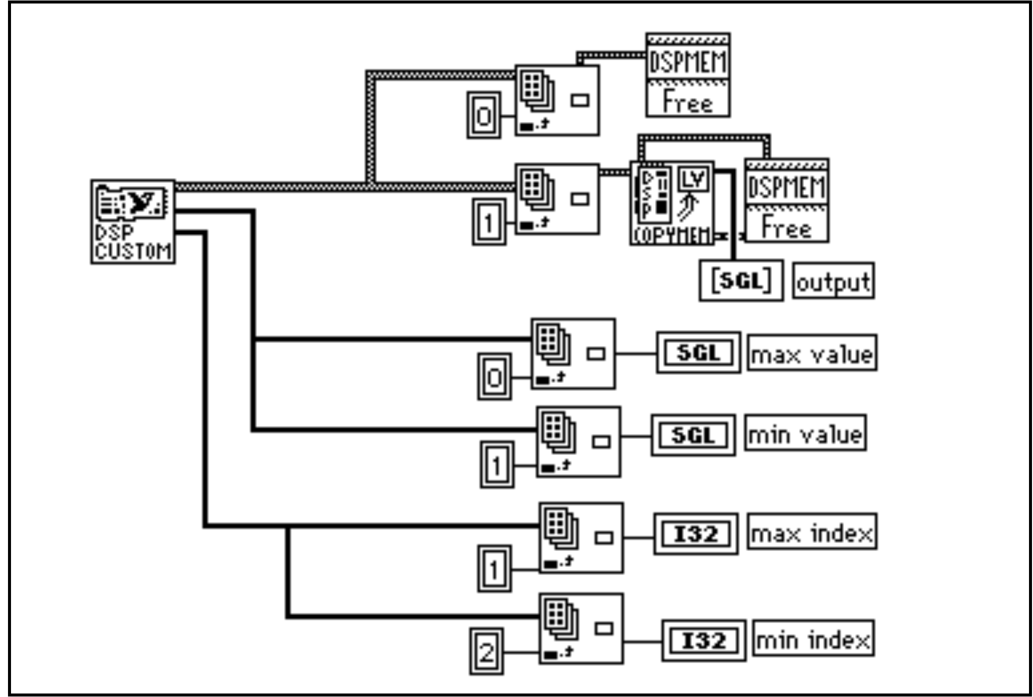


Figure 2-8. Block Diagram–How to Index the Output Arrays of the Custom VI to Obtain Results of `gmaxmin.c`

Figure 2-9 shows the whole block diagram that uses the Custom VI to call the custom function `gmaxmin.c` on the DSP board from LabVIEW. Figure 2-10 shows the front panel of the block diagram shown in Figure 2-9.

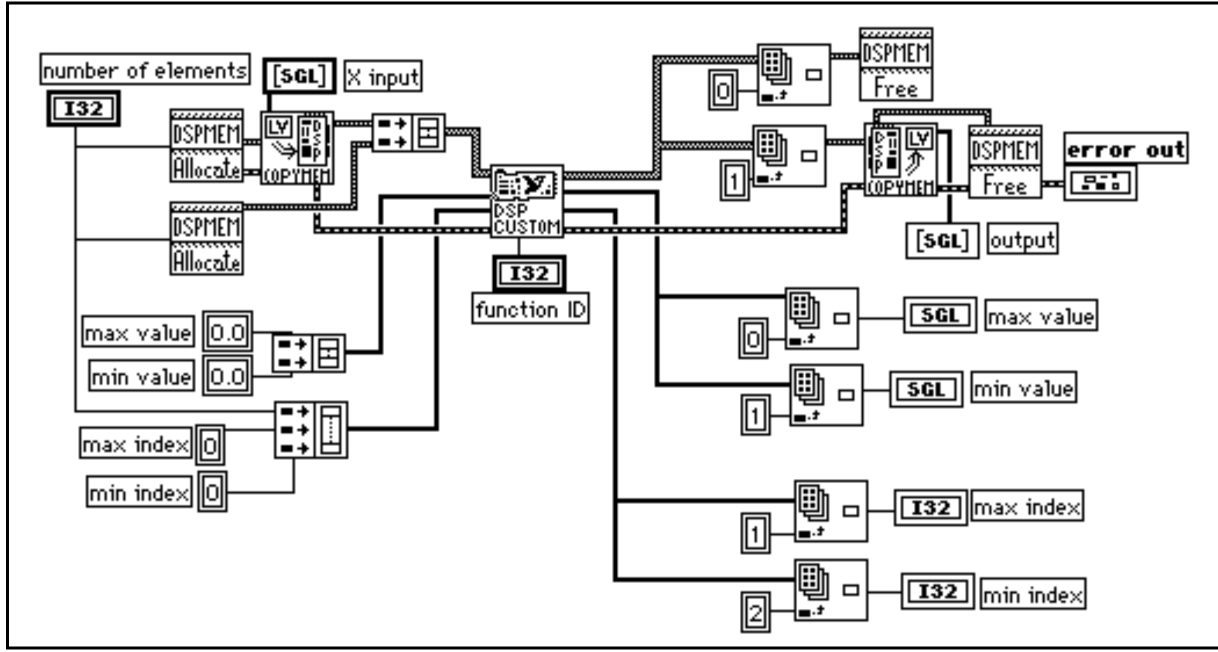


Figure 2-9. Block Diagram–Using the Custom VI to Call `gmaxmin.c` on the DSP Board from LabVIEW

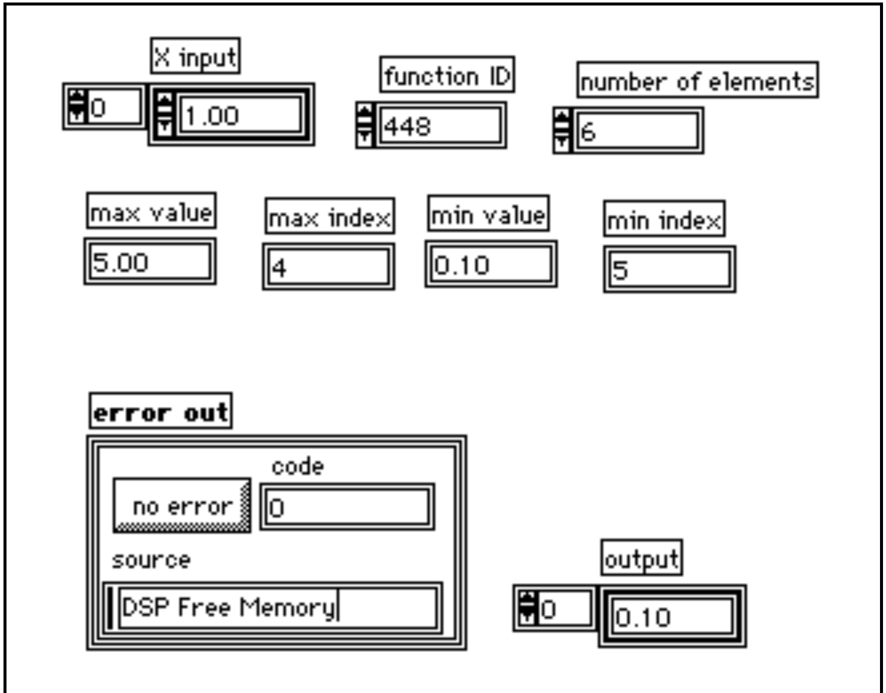


Figure 2-10. Front Panel–Using the Custom VI to Call `gmaxmin.c` on the DSP Board from LabVIEW

At this point, finish creating the VI interface to call your custom function on the DSP board from LabVIEW. You can customize this VI as a subVI using the method described in the *LabVIEW User Manual* if you want to use this VI from your other applications in LabVIEW.

Executing the Custom Function from LabVIEW

Run this VI from LabVIEW. Change the input array and display the results.

Chapter 3
DSP Board Function Overview

This chapter contains an overview of the prototypes of the C-callable NI-DSP Analysis functions on the DSP board that you can use in your custom programs.

Every NI-DSP Analysis VI calls a function on the DSP board. When you write your own custom functions, you can call these functions from your program. These C functions include numerical analysis, signal generation, digital signal processing, digital filtering, windowing, and memory management.

For more information about these functions, refer to the *NI-DSP Software Reference Manual for DOS/LabWindows.*

Note: The functions listed in this chapter are the same functions described in the *NI-DSP Software Reference Manual for DOS/LabWindows*, except for the following differences:

- For all of the functions, ignore the slot number parameter specified in the tables of parameters. The slot number is not meaningful when you call these functions from the DSP board.
- The integer parameters are all 32 bits long. The corresponding parameter type in the *NI-DSP Software Reference Manual for DOS/LabWindows* is 16-bit integer. When you call these functions in your custom program, use the 32-bit long type defined in this manual. Use the data types defined in this manual rather than the data types for similar functions listed in any other manual. The parameter order and meaning and the purpose of the functions, however, are equivalent with the functions described in the *NI-DSP Software Reference Manual for DOS/LabWindows.*
- The functions listed are all of the functions that you can call from your custom program. Not all the functions described in the *NI-DSP Software Reference Manual for DOS/LabWindows* are available in NI-DSP for LabVIEW for Windows.

The following is a list of the prototypes of each function that you can customize for use on the DSP board.

Signals

```
short DSP_Gaussian (long n, float sDev, float* noise, float seed)
short DSP_ImpTrain (long n, float amp, long delay, long period, float * x)
short DSP_Impulse (long n, float amp, long index, float * x)
short DSP_Pulse (long n, float amp, long delay, long width, float *x)
short DSP_Ramp (long n, float  first, float last, float * x)
short DSP_Sawtooth (long n, float amp, long delay, float cycles, float * x)
short DSP_Sinc (long n, float amp, float delay, float dt, float *x)
short DSP_Sine (long n, float amp, float phase, float cycles, float * x)
short DSP_Square (long n, float amp, float duty, float cycles, float * y)
short DSP_Triangle   (long n, float amp, float delay, float width,
                     float dt, float * x)
short DSP_TriTrain (long n, float amp, long delay, float cycles, float * x)
short DSP_Uniform (long n, float * noise, float seed)
short DSP_WhiteNoise (long n, float amp, float * noise, float seed)
```

Frequency Domain

```
short DSP_CrossPower (float * x, float * y, long n)
short DSP_CxFFT (float * x, float * y, long n)
short DSP_FHT (float * x, long n)
short DSP_InvFFT (float * x, float * y, long n)
short DSP_InvFHT (float * x, long n)
short DSP_ReFFT (float * x, float * y, long n)
short DSP_Spectrum (float * x, long n)
short DSP_ZeroPad (float * x, long n, long size)
```

Time Domain

```
short DSP_Convolution (float * x, long n, float * y, long m, float * cxy)
short DSP_Correlation (float * x, long n, float * y, long m, float * rxy)
short DSP_Decimate (float * x, long n, long decFact, long ave, float * y)
short DSP_Deconvolution (float * cxy, long n, float * y, long m, float * x)
short DSP_Difference (float * x, long n, float dt, float xInit,
                      float xFinal, float * y)
short DSP_Integrate  (float * x, long n, float dt, float xInit,
                      float xFinal, float * y)
```

Filters

```
short DSP_Bw_Coef    (float fs, float f_low, float f_hi, long order,
                      float * a, float * b, long type)
short DSP_Ch_Coef    (float fs, float f_low, float f_hi, long order,
                      float ripple, float * a, float * b, long type)
short DSP_Elp_Coef   (float fs, float f_low, float f_hi, long order, float
                      ripple, float atten, float * a, float * b, long type)
short DSP_EqRip_BPF  (float * x, long n, long taps, float fs, float  fh1,
                      float fl2, float fh2, float fl3, float * y)
short DSP_EqRip_BSF  (float * x, long n,  long taps, float fs, float fh1,
                      float fl2, float fh2, float fl3, float * y)
short DSP_EqRip_HPF  (float * x, long n, long taps, float fs, float fh1,
                      float fl2, float * y)
short DSP_EqRip_LPF  (float * x, long n, long taps, float fs, float fh1,
                      float fl2, float * y)
short DSP_IIR_Filter (float * x, long n, float * a, float * Condx, long sza,
                      float * b, float * Condy, long szb, float * y)
short DSP_InvCh_Coef (float fs, float f_low, float f_hi, long order, float
                      ripple, float *a, float * b, long type)
short DSP_Median_Filter (float * x, long n, long rank, float * y)
short DSP_Parks_McClellan  (long n, float fs, long bands, float * Amp, float
                            * fLow, float * fHi, float * wRipple, long
                            filterType, float * h, float * delta)
```

Windows

```
short DSP_CosWin(float * x, long n, long type)
short DSP_ExpWin (float * x, long n, float finalval)
short DSP_ForceWin (float * x, long n, float duty)
short DSP_GenCosineWin (float * x, long n, float * coeff, long m)
short DSP_KsrWin (float * x, long n, float beta)
short DSP_TriWin (float * x, long n)
```

Array Functions

```
short DSP_Abs (float * x, long n, float * y)
short DSP_Add (float * x, float * y, long n, float * z)
short DSP_Clip (float * x, long n, float upper, float lower, float * y)
short DSP_Div (float * x, float * y, long n, float * z)
short DSP_LinEv (float * x, long n, float a, float b, float * y)
short DSP_Log (float * x, long n, float mult, float * y)
short DSP_MaxMin     (float * x, long n, float * max, long * imax, float *
                      min, long * imin)
short DSP_Mul (float * x, float * y, long n, float * z)
short DSP_PolyEv (float * x, long n, float * coeff,long k, float * y)
short DSP_Prod (float * x, long n, float * prod)
short DSP_Reverse (float * x, long n)
short DSP_Set (float * x, long n, float a)
short DSP_Shift (float * x, long n, long shift, float * y)
short DSP_Sort (float * x, long n, long direction, float * y)
```

```
short DSP_Sqrt (float * x, long n, float * y)
short DSP_Sub (float * x, float * y, long n, flaot * z)
short DSP_Subset (float * x, long n, long index, long length, float * y)
short DSP_Sum (float * x, long n, float * sum)
short DSP_Unwrap (float * x, long n, float * y)
short DSP_To Polar (float * x, float * y, long n, float * mag,
                    float * phase)
short DSP_To Rect (float * mag, float * phase, long n, float * x,
                    float * y)
```

Memory Management Data Transfer

```
void * Alloc_Mem (long numbytes)
short DSP_CopyMem    (void * Source, void * Destination, long type, long
                     count)
short Free_Mem (long ptr)
```

Data Acquisition Functions

Some low-level data acquisition NI-DSP functions are also available for you to call in an application on the DSP board. You can use these low-level data acquisition functions to develop acquisition and waveform generation applications that occur on the DSP board, such as acquiring and sending data over the input and output channels. If you use any of the low-level data acquisition functions in your custom functions, please include the `dspdaq.h` file in your program. `dspdaq.h` contains all the prototypes of functions and the data structures that the data acquisition functions use. You can find this header file in the `LIB` directory. For more information about these functions, refer to Part 4, Chapter 5, *Data Acquisition Functions for the AT-DSP2200*, in the *NI-DSP Software Reference Manual for DOS/LabWindows*. All of the information in that chapter applies to the NI-DSP software for LabVIEW for Windows.

Chapter 4
Using the DMA VIs

This chapter describes two special VIs that transfer data between the host computer and the DSP board without interfering with the DSP board.

Most VIs in the NI-DSP library run sequentially on the DSP board. In other words, if you run a VI from LabVIEW, even though the actual function is running on the board, LabVIEW must wait for the function to finish before it can do anything else. If you have a custom function that runs indefinitely on the board, LabVIEW will continue to wait. However, every VI has the ability to check for timeout errors, and you can use the DSP Timeout VI to resolve this problem by setting a timeout limit. The host will wait for the DSP board to finish an operation for only the amount of time you designate; if the DSP board does not complete the function in the specified amount of time, the VI will return a timeout error. If your custom function runs a long time or indefinitely, you can set a very short timeout period and ignore the timeout error in your LabVIEW program. However, under these conditions you normally lose communication with the DSP board until you reset it with the DSP Reset VI.

Two VIs–DSP DMA Copy(LV to DSP) and DSP DMA Copy(DSP to LV)–solve this communication problem. These two VIs can transfer data between the host and the DSP board using onboard DMA without interfering with the DSP board. You can use the DSP DMA Copy(LV to DSP) VI to send data to a designated memory location on the DSP board from LabVIEW when the DSP board is still running. The function running on the board can read data from that location and update operations that depend on that data. LabVIEW can also read data back from the DSP board using the DSP DMA Copy(DSP to LV) VI to check the status of the board or read results while the board is still running. In other words, you can still communicate with the board.

When you use the DSP DMA Copy(DSP to LV) and DSP DMA Copy(LV to DSP) VIs, remember the following:

- These two VIs require a real DSP address to transfer data. You can determine this address in two ways:
    - Give a valid DSP address, such as FFF000 hex, if you know the exact DSP memory address of a buffer or variable that your DSP program will use.
    - Allocate a normal DSP Handle Cluster and use the DSP Handle To Address VI to obtain the valid DSP address for this DSP Handle Cluster. You can use this DSP address in your program on the DSP board as you would use any pointer. Remember to allocate the DSP Handle Cluster before you call your custom function through the DSP Custom Function VI.
- The data format will not change when you transfer data. If the data is floating point, you need to convert the data format in your code between DSP format and IEEE format. When you send floating-point data from LabVIEW to the DSP board, convert the data to DSP format before using it in your program on the DSP board. When you copy floating-point data back from the DSP board to LabVIEW, convert the data to IEEE format before you save it to the DSP memory location from which you will copy data.
- Do not use data acquisition VIs to acquire data to PC memory or send data from PC to DSP memory while using these two VIs, because these VIs use the same DSP board register, PDR, to transfer data.

Two examples shipped with this package show you how to make use of these VIs. The examples are installed in the `Examples` subdirectory of your NI-DSP package.

The first example is a simple spectral analyzer. The main VI is called Analyzer VI. It uses the DSP Custom Function VI to call a custom function running on the board. The scheme is as follows:

- Custom function on the board:

  1. Set the data acquisition parameter and initialize the board.

  2. Start the data acquisition.

```
while(TRUE) {
   when data is ready, copy it to the process buffer
   read processing information from DSP memory location 1
   depending on the processing information
      apply the selected window to the data
      apply the selected filter to the data
      compute the FFT magnitude or Power Spectrum
   Transfer the data format to IEEE format.
   Save processed data in DSP memory location 2.
}
```

- In LabVIEW:

  1. Initialize the board and set parameters.

  2. Allocate DSP Handle Cluster 1; get the address for this handle (this is DSP memory location 1).

  3. Allocate DSP Handle Cluster 2; get the address for this handle (this is DSP memory location 2).

  4. Set a timeout limit to a very small number (2 ms).

  5. Pass DSP Handle Cluster 1 and 2 with the other necessary parameters to the DSP Custom Function VI and call the Analyzer VI.

  6. Ignore the timeout error returned from the DSP Custom Function VI.

```
While(STOP button not pressed) {
   send new processing information (you can change certain processing
   controls from the front panel) to DSP memory location 1 using the
   DSP DMA Copy(LV to DSP) VI.
   copy the processed data from DSP memory location 2 to LabVIEW and
   plot the data.
}
```

7. Call the DSP Reset VI to reset the DSP board.

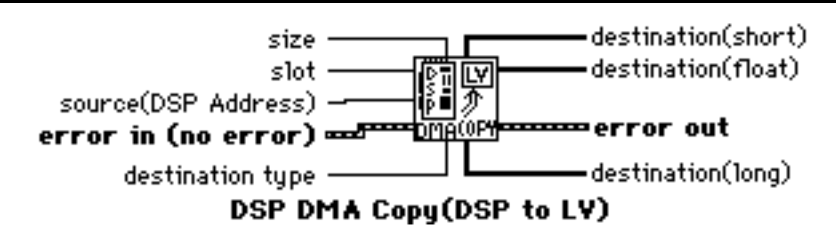
The second example is the audio equalizer example. The VI loads the audio equalizer kernel called `audio.out` using the DSP Load VI. It uses the DSP DMA Copy(LV to DSP) VI to send the new gains for each channel to the DSP board and uses the DSP DMA Copy(DSP to LV) VI to copy the processed audio data back to LabVIEW and then plots the data.

You can find all of the source codes and VIs for these two examples in the `Examples` subdirectory of your NI-DSP package.

These examples show you how to use the DSP DMA Copy(DSP to LV) and DSP DMA Copy(LV to DSP) VIs to communicate between the DSP board and the host computer. They also show you how to run parallel DSP and LabVIEW applications, which makes the board more flexible and powerful. With a well-designed system, you can gain both speed and flexibility.

DSP DMA Copy(DSP to LV)

Copies a buffer of **size** elements from **source(DSP Address)** on the DSP board to one of the **destination** arrays in LabVIEW using the onboard DMA transfer method.

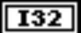


This VI uses only the DMA controller on the DSP board to transfer data. Therefore, it does not interfere with the other program that is running on the DSP board.

Note: This VI does not convert floating-point data to the IEEE format while copying. When you copy floating data, convert to IEEE format in your program on the DSP board. However, the Copy Mem(DSP to LV) VI automatically converts data format during the copying procedure.

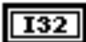
To copy data correctly from the DSP board to LabVIEW, you must indicate what type of data is stored in the **source(DSP Address)** buffer on the DSP board, set the **destination type** to the appropriate type, and wire to the corresponding **destination** terminal. This VI has three destination types–float (32-bit), short (16-bit), and long (32-bit). Remember, you must wire only one destination terminal.

slot is the board ID number. **slot** defaults to 3.



source(DSP Address) is the actual DSP address from which you copy your data. It should be a 24-bit integer (such as FFF000 hex) less than or equal to FFFFFF hex. If you know the exact DSP address where your data is stored, use this address directly. Alternatively, you can use the DSP Handle To Address VI to convert a DSP Handle Cluster to an actual DSP address.

size indicates the amount of data you want to copy back from the **source(DSP Address)** buffer on the DSP board to the **destination** buffer in the LabVIEW. **size** defaults to 0.



destination type indicates the type of the data in the **source(DSP Address)** buffer on the DSP board. It has three options:

0: 32-bit floating-point.

1: 16-bit short integer.

3: 32-bit long integer.

destination type defaults to 32-bit floating-point.

[SGL] **destination(float)** is the terminal to which you must wire the destination buffer if the data you want to copy is a 32-bit floating-point data array. Remember, the program on the DSP board must convert this data to IEEE format before it is copied.

[I16] **destination(short)** is the terminal to which you must wire the destination buffer if the data you want to copy is a 16-bit short data array.

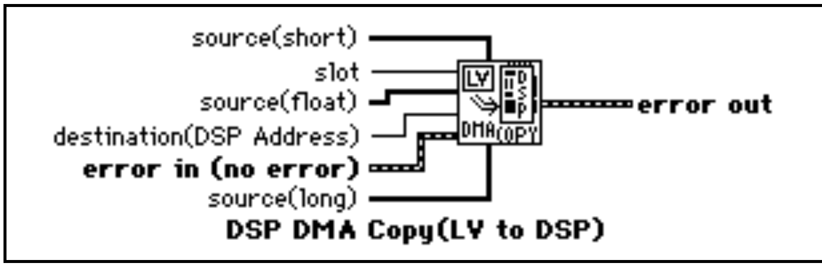
[I32] **destination(long)** is the terminal to which you must wire the destination buffer if the data you want to copy is a 32-bit long data array.

error in(no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

DSP DMA Copy(LV to DSP)

Copies the data in the LabVIEW array **source** to the **destination(DSP Address)** on the DSP board using the onboard DMA method.



This VI uses only the DMA controller on the DSP board to transfer data. Therefore, it does not interfere with the other program that is running on the DSP board unless you copy the data to the DSP memory location that is being used by the other program.

Note: This VI does not convert floating-point data to DSP format while copying. When you copy floating data, convert to DSP format in your program on the DSP board before using this data. However, the Copy Mem(LV to DSP) VI automatically converts data format during the copying procedure.

The **source** buffer can contain one of three kinds of data–float (32-bit), short (16-bit), and long (32-bit). To copy different types of data, you must wire the source data buffer to the appropriate source terminal. For example, if you want to copy floating data, you must wire the data buffer to the terminal **source(float)**. You must wire only one source terminal. The **destination** buffer must be large enough to contain all of the data from the **source** buffer.

I16 **slot** is the board ID number. **slot** defaults to 3.

[SGL] **source(float)** is the terminal to which you must wire the source buffer if the data you want to copy is a 32-bit floating-point data array. Remember, the program on the DSP board must convert this data to DSP format before using it.

[I32] **source(long)** is the terminal to which you must wire the source buffer if the data you want to copy is a 32-bit long data array.

[I16] **source(short)** is the terminal to which you must wire the source buffer if the data you want to copy is a 16-bit long data array.

I32 **destination(DSP Address)** is the actual DSP address to which you want to copy your data. It should be a 24-bit integer (such as FFF000 hex) less than or equal to FFFFFF hex. If you know the exact DSP address where you want to store the data, you can use this address directly. Alternatively, you can use the DSP Handle To Address VI to convert a DSP Handle Cluster to an actual DSP address.

error in(no error) contains the error information from a previous VI. If an error occurs, it is passed out **error out** and no other calls are made.

error out contains the error information for this call.

Appendix A
Error Codes

This appendix contains a list of the error codes returned by the NI-DSP Analysis VIs and the corresponding error messages.

Error Conditions

If an error condition occurs during execution of any of the VIs in the NI-DSP Analysis, the VI returns an error code. This code is a value which indicates the type of error that occurred. The currently defined error codes and their associated meanings are given in Table A-1. For error codes other than those listed, refer to the *NI-DAQ Software Reference Manual for DOS/Windows/LabWindows*.

Table A-1. NI-DSP Analysis Library Error Codes

Error Number	Error Name	Description
0	`noError`	No error; the call was successful.
-10401	`unknownDeviceErr`	The board specified is not a National Instruments DSP board.
-10005	`badDeviceErr`	The board number used in function call should be $1 \leq board \leq 8$.
-10403	`deviceSupportErr`	Function cannot be executed by specified board.
-10444	`memFullErr`	Insufficient memory or disk space.
-10243	`configFileErr`	Board configuration file not found.
-10247	`cmosConfigErr`	EISA system configuration invalid.
-10459	`DLLInterfaceErr`	The DLL could not be called due to an interface error.
-21204	`SamplesGTZero`	The number of samples must be greater than zero.
-21205	`SamplesGEZero`	The number of samples must be greater than or equal to zero.
-21206	`SamplesGETwo`	The number of samples must be greater than or equal to two.
-21207	`SizesGTZero`	The sizes of the input sequences must be greater than zero.
-21208	`xSizeGEySize`	The size of x must be greater than or equal to the size of y.
-21209	`OutSizeGEInSize`	The output array size must be greater than or equal to the input array size.
-21210	`OutOfMem`	There is not enough space left on the DSP board for onboard processing.
-21212	`DecFactErr`	The decimating factor must meet: $0 < \text{decimating factor} \leq \text{samples}$.
-21213	`WidthLTSamples`	The width must meet: $0 < \text{width} < \text{samples}$.
-21214	`IndexLTSamples`	The index must meet: $0 \leq \text{index} < \text{samples}$.
-21215	`DelayGEZero`	The delay must be greater than or equal to zero.
-21216	`WidthGEZero`	The width must be greater than or equal to zero.
-21217	`DelayWidthErr`	The following condition must be met: $0 \leq (\text{delay} + \text{width}) < \text{samples}$.

(continues)

Table A-1. NI-DSP Analysis Library Error Codes (Continued)

Error Number	Error Name	Description
-21218	WinDutyCyclesErr	The window duty cycle value must be between 0.0 and 100.0.
-21219	dtGTZero	dt must be greater than zero.
-21220	DutyCycleErr	The duty cycle must meet: 0 ≤ duty cycle ≤ 100.
-21221	CyclesErr	The number of cycles must be greater than zero and less than or equal to the number of samples.
-21223	UpperGELower	The upper value must be greater than or equal to the lower value.
-21225	IntervalNumErr	The number of intervals must be greater than zero.
-21226	MixedSignErr	The sign of y in Ax=y values must be all positive or negative.
-21227	SizeGTOrder	The array size must be greater than the order.
-21228	OrderGTZero	The order must be greater than zero.
-21229	ConvSizeErr	The number of elements of convolved arrays should be greater than zero.
-21233	NyquistErr	The cutoff frequency, fc, must meet: 0 ≤ fc≤ fs/2.
-21234	Nyquist2Err	The following conditions must be met: 0 ≤ f_low ≤ f_high ≤ fs/2.
-21235	RippleGTZero	The ripple amplitude must be greater than zero.
-21236	AttenGTZero	The attenuation must be greater than zero.
-21237	WidthGTZero	The width must be greater than zero.
-21242	SizeGTZero	The size of the input sequence must be greater than zero.
-21243	NullVectorErr	The input vector is null. The unit vector does not exist.
-21245	AttenGTRipple	The attenuation must be greater than the ripple amplitude.
-21246	StepSizeErr	The step-size parameter μ must meet: 0 ≤ μ ≤ 0.1.
-21248	LeakErr	The leakage coefficient, Leak, must meet: 0 ≤ Leak ≤ μ.
-21249	FilterDesignErr	The filter cannot be designed with the specified input parameters.
-21250	LenGEOne	The number of coefficients must be greater than or equal to one.
-21251	RankErr	The rank of the filter must meet: 1 ≤ (2 rank +1) ≤ size.
-21297	NotPowerOfTwo	The size of the input array must be a valid power of two: size = 2^m, 0 < m < 14.
-21301	MEM_HRDWARE_ERR	The onboard hardware diagnostic software found a memory hardware error. Running functions on the board is unreliable.
-21302	ZERO_ALLOCATION	There are no allocated buffers on the DSP board.
-21303	TooBigAnOffset	The indexing requested falls outside of the buffer into which indexing is requested.
-21304	HandleNotFound	The DSP Handle specified is not found on this board: it has either been freed or never allocated.
-21305	TooManyAllocs	The Memory Look Up Table is full: no more allocations allowed before freeing up some DSP handles (maximum allowed = 128).
-21306	PtrNotFound	The DSP address pointer could not be found to free the buffer.
-21307	ParamSizeErr	One of the parameters does not have enough space allocated for it on the board.

(continues)

Table A-1. NI-DSP Analysis Library Error Codes (Continued)

Error Number	Error Name	Description
-21308	`TransferSizeErr`	The size of requested block transfer does not have enough space allocated for it on the board.
-21309	`NotDSPHandle`	The DSP Handle specified is not valid for this board: it has either been freed or never allocated.
-21310	`HandleSlotErr`	The DSP Handle used does not belong to the same board on which the function request is made.
-21311	`HandleAllocErr`	Failed to allocate a handle in the interface code.
-21312	`TimeOutErr`	The Kernel did not respond or finish function execution in the selected timeout.
-21313	`COFF_FilePathErr`	Could not find the DSP Library file in the directory set by the `DAQCONF` utility.
-21314	`CoffHdrErr`	The file you are attempting to load to the DSP board is not created using the WE DSP32C tools (not an acceptable COFF file).
-21315	`CoffSectErr`	Found an unimplemented section flag in the COFF file.
-21317	`AddrSpaceErr`	The COFF file you are trying to load is linked with a memory map that exceeds the maximum memory space allowed by the WE DSP32C chip (2^{24}).
-21318	`NoBrdRespToTransfer`	The DSP board on which you are trying to load the COFF file is not responding to the transfer of the file. Check if there is a board in the slot desired and verify with the `DAQCONF` utility.
-21324	`FNC_ERR`	The function ID specified does not correspond to a function that is part of the DSP Library currently running on the board.
-21325	`GROUP_ERR`	The function ID specified does not belong to any group that is part of the DSP Library currently running on the board.
-21331	`NotFindFunctionInDLL`	An error occurs when loading `dsp.dll` library in CIN.
-21332	`LoadDLLLibErr`	An error occurs when loading a function in `dsp.dll` in CIN.
-21333	`ErrInGetFunctionHandle`	An error occurs when calling the `GetIndirectFunctionHandle` function in CIN.
-21334	`InvalidBytesSelect`	bytes/element selector can only select between 4 bytes(0) and 2 bytes(1).
-21336	`AllocArrayFailed`	Failed to allocate a LabVIEW array.
-21337	`InvalidArrayType`	The array type can only be 32-bit floating point data(0), 16-bit long integer data(1) or 32-bit short integer data(2).
-21338	`PathnameTooLong`	The path name should be less than 256 characters.
-21339	`InvalidCopyType`	The entire/partial copy selector can only select between entire copy (0) and partial copy(1).
-21340	`CopyNumGTZero`	The number of data to copy should be greater than zero.
-21341	`OffsetGEZero`	The offset should be greater than or equal to zero.
-21342	`IndexSizeGTZero`	The size of the New DSP Handle Cluster should be greater than zero.

(continues)

Table A-1. NI-DSP Analysis Library Error Codes (Continued)

Error Number	Error Name	Description
-21343	`IndexSizeOffsetErr`	The size+offset should be less than or equal to the size of the DSP Handle Cluster that you index into.
-21344	`InvalidDataType`	The data type can only be 4 bytes long(0) or 2 bytes long(1) .
-21345	`NumBytesGTZero`	The number of bytes to allocate should be greater than zero.

Appendix B
Customer Communication

For your convenience, this appendix contains forms to help you gather the information necessary to help us solve technical problems you might have as well as a form you can use to comment on the product documentation. Filling out a copy of the *Technical Support Form* before contacting National Instruments helps us help you better and faster.

National Instruments provides comprehensive technical assistance around the world. In the U.S. and Canada, applications engineers are available Monday through Friday from 8:00 a.m. to 6:00 p.m. (central time). In other countries, contact the nearest branch office. You may fax questions to us at any time.

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(512) 795-8248
Technical support fax: (800) 328-2203
(512) 794-5678

Branch Offices	Phone Number	Fax Number
Australia	(03) 879 9422	(03) 879 9179
Austria	(0662) 435986	(0662) 437010-19
Belgium	02/757.00.20	02/757.03.11
Denmark	45 76 26 00	45 76 71 11
Finland	(90) 527 2321	(90) 502 2930
France	(1) 48 14 24 00	(1) 48 14 24 14
Germany	089/741 31 30	089/714 60 35
Italy	02/48301892	02/48301915
Japan	(03) 3788-1921	(03) 3788-1923
Netherlands	03480-33466	03480-30673
Norway	32-848400	32-848600
Spain	(91) 640 0085	(91) 640 0533
Sweden	08-730 49 70	08-730 43 70
Switzerland	056/20 51 51	056/27 00 25
U.K.	0635 523545	0635 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 ____

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 will reproduce the problem _________________________________

____

____

NI-DSP for LabVIEW for Windows Hardware and Software Configuration Form

Record the settings and revisions of your hardware and software on the line located to the right of each item. Complete 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

- DSP Hardware ______________________________
- Interrupt Level of Hardware ______________________________
- DMA Channels of Hardware ______________________________
- Base I/O Address of Hardware ______________________________
- NI-DSP Version ______________________________
- NI-DAQ Version ______________________________
- LabVIEW Version ______________________________

Other Products

- Computer Make and Model ______________________________
- Computer Bus (XT/AT/ISA or EISA) ______________________________
- Microprocessor ______________________________
- Clock Frequency ______________________________
- Type of Video Board Installed ______________________________
- DOS Version ______________________________
- 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: **NI-DSP™ Software Reference Manual for LabVIEW® for Windows**

Edition Date: **December 1993**

Part Number: **320571-01**

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.

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Glossary

Prefix	Meaning	Value
n-	nano-	10^{-9}
µ-	micro-	10^{-6}
m-	milli-	10^{-3}
k-	kilo-	10^{3}
M-	mega-	10^{6}

1D one-dimensional
2D two-dimensional
Cx complex
dB decibels
DFT Discrete Fourier Transform
DMA direct memory access
DSP digital signal processing
FFT Fast Fourier Transform
FHT Fast Hartley Transform
FIR finite impulse response (filter)
hex hexadecimal
IDFT Inverse Discrete Fourier Transform
IEEE Institute of Electrical and Electronic Engineers
IFFT Inverse Fast Fourier Transform
IFHT Inverse Fast Hartley Transform
IIR infinite impulse response (filter)
I/O input/output
Kwords 1,024 words of memory
LMS least mean square
LSB least significant bit (of a word)
LV LabVIEW
MB megabytes of memory
MLUT Memory Look Up Table
MSE mean squared error
Mwords 1,024 x 1,024 words of memory
pt point
RTSI Real-Time System Integration
SCXI Signal Conditioning eXtensions for Instrumentation
s seconds
word 32-bits of memory or four bytes, unless otherwise stated

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