dsPICworks™ Version 1.0 Data Analysis and Digital Signal Processing Software User's Guide

Developed for
Microchip Technology Inc.

by
Momentum Data Systems Inc.

Table of Contents

CHAPTER 1 *Introduction*

dsPICworks^{TM} software[1] for Windows is an easy-to-use application for general digital signal processing. It combines functionality and sophistication to allow the DSP professional to accomplish complex tasks without having to go through the arduous and highly technical mathematics inherent to these applications.

In addition, dsPICworks software provides features that allow it to interface with the Microchip development tool suite, MPLAB® IDE (Integrated Development Environment), and the dsPIC30F assembler - MPLAB ASM30.

This manual is a reference guide to dsPICworks software. It is not intended to be a tutorial on digital signal processing since several excellent texts on the subject exist and it is assumed that the user has had a certain amount of academic or professional exposure to the subject.

System operation is controlled via the standard Windows interface of a main menu bar with pull-down menus and dialog boxes. The main menu bar consists of the following:

TABLE 1-1 Main Menu Selections

Selection	Description
File	Standard file operations including, Record/ Play Script, Import/Export Files, Print and Exit functions and information on dsPICworks software
View	Toolbar and Status bar selections
Edit	Waveform graphical editing functions
Generator	Waveform generation for standard waveforms

1. dsPICworks and dsPIC are trademarks, and MPLAB is a registered trademark, of Microchip Technology Inc.

Selection	Description
Operation	Mathematical operations on time domain waveframe, graphical and real time displays
DSP	Additional DSP operations on time domain & Frequency domain
Display	Waveform displays
Utilities	System utility functions such as file format conversion, number type conversion, graph control
Window	Selection of Windows for display and order of windows, Font & Color Selection

1.1 Digital Signal Processing Capabilities

dsPICworks software is a general purpose signal processing system. Signals can be generated from one of the signal generators. All operations or commands work on the entire signal. Thus, the Add command of two signals adds corresponding samples in the first file to the second file to give a new output file. Other commands cannot be issued until this command has completed. An extensive variety of operations or commands are available. Script files are available for repetitive operations and a file import/export capability can be used to facilitate interfacing with other systems. All commands use a typical Windows interface with pull down menus and popup dialog boxes. Menus and commands have been organized for very intuitive use. The following sections provide an overview of dsPICworks software capability.

1.1.1 Generators

Waveform synthesis takes place in the Generators section of the program which contains a large variety of functions for generating discrete data sequences.

1. Sinusoidal
2. Square
3. Triangular
4. Exponential
5. Unit Sample
6. Unit Step
7. Swept Sine
8. Windowing Functions
9. Sinc
10. Ramp
11. Noise

Noise can be added to waveform based on probability density functions. Signal length is limited only by disk capacity of the system disk. Signals are real valued only. Signals can be generated as 32-bit floating point values or 16-bit fractional fixed point values.

1.1.2 Waveform Editing

Graphical waveform editing using the mouse to select graph segments is available. *Cut, Copy, Paste* and *Delete* of waveform segments can be used to easily manipulate signals.

1.1.3 Operations on Generated or Acquired Signals

TABLE 1-2 Summary of options under the Operations menu in dsPICworks software

Operation	Function
Arithmetic	$y(n) = b_0x(n) + b_1x(n-1) + b_2x(n-2) - a_1y(n-1) - a_2y(n-2)$ $y(n) = ax_1(n) + bx_2(n) + c$ $y(n) = ax_1(n)x_2(n)$
Reciprocal	$y(n) = \frac{a}{x(n)}$
Square	$y(n) = a[x(n)]^2$
Square Root	$y(n) = a\,\mathrm{sgn}\,x\sqrt{\lvert x\rvert}$
Shift	$y(n) = x(n-N)$
Flip	Reverses the order of samples in a given sequence y(n) = x(N-1-n) for n=0,1,...,N-1
Join	Concatenates two sequences, the second sequence is appended to the end of the first sequence.
Trigonometric	y(n)=sin[x(n)]
	y(n) = cos[x(n)]
	y(n) = tan[x(n)]
Exponential	$y(n) = Ae^{ax(n)}$
Extract	Allows the user to extract a segment of data from one file to another file.
Smooth	$y(n) = \frac{1}{d}\sum_{i=0}^{d-1} x(n-i)$
Sample & Hold	Samples on a periodic basis and holds the value until the next sample period.
Difference	$y(n) = x(n) - x(n-d)$
Quantize Fixed Point	0 - 16 bit quantization
Signal Statistics	Mean,max,variance,min, SD etc.

1.1.4 DSP Operations

TABLE 1-3 Summary of options under the DSP menu in dsPICworks software

Operation	Function
Signal Filtering	Apply filter designed by QEDesign to a time domain sequence
Convolution	Convolve any two time domain signals
Autocorrelation	Computes the autocorrelation of a given sequence $r(k) = \sum_n x(n+k)x(n)$
Crosscorrelation	Computes the crosscorrelation for two given sequences $r(k) = \sum_n y(n+k)x(n)$
Decimation	Sample rate reduction. Generates a sequence by decimating the input sequence with a user specified decimation factor y(n) = x(Dn) n=0,1,...int[(N-1)/D]
Interpolation	Sample rate increase. Generates a new sequence by inserting zeros between samples with a user specified interpolating factor y(n) =x(n/U) n=0,U,2U,...
Fast Fourier Transform	Fast Fourier Transform Operation
Inverse FFT	Inverse Fast Fourier Transform Operation
Average FFT	Average the magnitude of a set of FFT frames

1.1.5 Display Capabilities

TABLE 1-4 Summary of options under the Display menu in dsPICworks software

Operation	Function
Time Domain Displays	Single and multi-channel waveform displays
Power Display	One dimensional Power display available for stored waveforms only
Phase Display	Phase display for stored waveforms.
Spectral Display	One-dimensional Magnitude display. Normally used for real-time display. Can be used to display stored waveform.

Operation	Function
Two-dimensional Spectral Display	Two-dimensional magnitude or power display. Classic spectogram or sonogram normally used for real-time display. Can be used to display stored waveform.
Three-dimensional Spectral Display	Three dimensional magnitude or power display. Classic waterfall display normally used for real-time display. Can be used to display stored waveform.

1.1.6 Miscellaneous Features

TABLE 1-5 **Options under the Utilities menu in dsPICworks software**

Operation	Function
ASCII to Binary Conversion	Converts format from ASCII to binary and vice versa
Integer to Float Conversion	Converts numeric data types
Demultiplex/Multiplex Operations	For separating and combining multi-channel signals.

1.2 Hardware Requirements

dsPICworks software for Windows requires Windows 98 or greater and a minimum of 2 Mbytes of RAM. Note, Windows ME is not a recommended platform for running dsPICworks software.

1.3 Installation Procedure

1.3.1 Software Installation

Invoke the ***Windows***/Run command and type in

```
<drive>:setup

e.g. E:\setup
```

Note: This starts Windows and executes SETUP on the dsPICworks software CD which decompresses the files and installs dsPICworks software in a program group.

1.4 System Operation

dsPICworks software was designed to be intuitively easy for the user to operate.

The menus and dialogs are largely self-explanatory thus allowing the system to be used with the minimum of difficulty. The user merely needs to select the desired option/s by following the menu and dialog prompts. All dialogs have a cancel box which will cause an escape to the main menu.

Note that enabled menu items are shown in black with disabled options appearing in gray.

1.4.1 Filenames

dsPICworks software is file oriented. All signals whether generated or acquired are stored in files. All operations on signals generally require the specification of one or more input files and an output file.

To simplify the operation as much as possible while retaining flexibility, filename fields are entered by clicking on the filename field. A standard file open box then pops up. For input files, simply select the desired field by clicking on it. For output files, simply enter the desired filename. If the required suffix is not part of the filename, the system will automatically append it. In all cases, the desired filename will appear in the filename field of the dialog box.

1.4.2 Computer Arithmetic

In computer systems, there is only a finite set of number representations available. How these numbers are interpreted, depends on whether the numbers are fractions, integers, or mixed numbers; the formats of numbers are floating point or fixed point; how negative numbers are represented and how many digits are provided for number representations. The following discussion assumes that numbers are represented in the base-2 or binary system, and 2’s complement notation is used to represent numbers.

Consider the following number:

$$x_a = a_0a_1a_2...a_{n-1}$$

where a_0 is the sign bit taking value 0 or 1. The remaining digits in x_a specify either the true magnitude when $a_0 = 0$ or two’s complemented magnitude when $a_0 = 1$.

In fixed point notation, the binary point is regarded as fixed at the same location within the number and can be divided into three categories. In integer notation, the radix point is to the immediate right of the least significant digit a_{n-1} which ensures that the magnitude of the number is always an integer. In fractional notation, the binary point is positioned between the sign digit a_0 and most significant magnitude digit a_1. This ensures that any fraction is always less than one. The advantage of fractional fixed point is that multiplication of two fractional numbers results in another fractional number and no overflow occurs, but this is not true in the case of addition. In mixed notation the binary point is positioned somewhere in between. In summary, fixed point notation can be represented as follows:

TABLE 1-6 **Fixed point notation representation**

Format	Representation
Integer format	$a_0a_1a_2...a_{n-1}\ \Delta$
Fraction format	$a_0\ \Delta\ a_1\ a_2...a_{n-1}$
Mixed format	$a_0\ a_1...a_{i-\Delta}a_i...a_{n-1}$

where ∆ is the binary point. Keeping track of the binary point is not always a trivial matter. The necessity of always knowing the location of the radix point is a major drawback of the fixed point format.

dsPICworks software supports two types of number types: 16-bit fractional fixed point and 32-bit floating point which are described in detail in the following sections.

1.4.3 16 bit fractional fixed point

A 16 bit fractional number representation is shown as:

$$x_b=b_0\Delta b_1b_2b_3...b_{15}$$

where ∆ represents the binary point and b0 is the sign bit representation of positive numbers if zero or negative numbers if one. The fractional representation is limited to numbers between -1 and $1-2^{-15}$. Fractional fixed point is usually used instead of integer fixed point because multiplying two fractional numbers together yields another fractional number, thus no overflow problems can occur in multiplication operations. Overflow can occur in addition, however, this is acceptable if the final value of a summation is less than 1 in absolute value. In contrast, multiplying two integers can form a product that cannot be represented as a valid storage format integer. Fractional arithmetic can be done even if a processor has only integer arithmetic. Multiplying two numbers scaled (1,15) gives a result in a double length accumulator scaled (2,30). Thus to get proper value saved in a memory location, shift the double length accumulator 1 bit to the left and use the 16 most significant (upper) bits.

To illustrate this point assuming the following two numbers reside in registers of a microprocessor which only handles integer arithmetic.

TABLE 1-7 **Integer Arithmetic Example 1**

	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
R0	1	0	1	1	0	0	0	0	0	0	0	0	0	0	0	0
R1	0	0	1	1	1	0	0	0	0	0	0	0	0	0	0	0

R0 contains the value of the multiplicand and R1 has the value of the multiplier.Since fractional fixed point is assumed, the actual value of R0 and R1 are -0.625_{10} and 0.4375_{10} respectively. The processor would carry this multiplication and store the result in 32 bit accumulators as follows:

TABLE 1-8 Integer Arithmetic Example 2

	0	1	2	3	4	5	6	7	8	9	...	31
AC	1	1	1	0	1	1	1	0	1	0	...	0

Note, there is one additional sign bit that must be eliminated by a left shift.

Shifting the accumulator 1 bit to the left and storing the upper 16 bits in the memory location results in:

TABLE 1-9 Integer Arithmetic Example 3

	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
MEM	1	1	0	1	1	1	0	1	0	0	0	0	0	0	0	0

The actual value of this memory location is -0.2734375_{10} which is the correct result of multiplying -0.625_{10} and 0.4375_{10}.

Multiplication in integer or mixed point format can lead to overflow problems.

A problem arises when exceeding the limit $[-1, 1-2^{-15}]$. This problem is well known as two's complement arithmetic overflow. There are two alternatives to handle such problems: wrap-around and saturation overflow or clipping. In the wrap-around method, the resulting error can be very large when overflow occurs. For example, when `0.110...0(0.7500)` and `0.100.0 (0.500)` is added, the carry propagates all the way to the sign bit so that the result is `1.010...0` (-0.75_{10}) not (1.25_{10}). Thus the resulting error can be very large when overflow occurs. The clipping method, will give a result of -1 if the number is less than -1(the smallest negative number). The clipping method will give a result of $1-2^{-15}$ if the number is greater than $1-2^{-15}$ (the largest positive number). With this approach, the size of the error does not increase abruptly when overflow occurs. However, it destroys the useful property of two's complement arithmetic, i.e. if several two's complement numbers whose sum would not overflow are added, then the result is still correct. dsPICworks software provides both options. Depending on the application, each method has its own advantages. For example, when generating a waveform with peak amplitude equal to one using 16 bit fractional fixed point and adding random noise option to that waveform, due to fluctuation of noise, the value at some point would exceed the range $[-1, 1)$, the user would be better off to use the clipping option.

1.4.4 32-bit Floating point

Alternatively, numbers can be represented in floating point format. A floating point number consists of two parts: a fraction f and an exponent e. The two parts represent a number that is obtained by multiplying f times two raised to the power e, that is, the floating point number x_a can be expressed as:

$$x_a = f \times 2^e \qquad \text{(EQ 1.1)}$$

where f and e are both signed, fixed point numbers. dsPICworks software implements the following 32-bit binary floating point format which complies with the IEEE floating point

TABLE 1-10 **Bit allocation**

0	8	31
Sign	Exponent	Fraction

TABLE 1-11 **Floating Point Format**

Bits	Name	Content
0	Sign	1 iff number is negative
1-8	Exponent	8-bit exponent, biased by 127. Values of all zeros and all ones reserved
9-31	Fraction	23-bit fraction component of normalized significant. The “one” bit is “hidden”

The format consists of a 1-bit sign s, an 8-bit biased exponent e, and a 23- bit fraction f. Normalizing format is used to acquire one more bit of precision. The magnitude of the normalized fraction has an absolute value within the range [0.5,1).

The only exception is a floating point number equal to 0. If a number cannot be normalized because it does not have a non-zero digit, it is represented in floating point by an all-zero fraction and an all-zero exponent, i.e. e=0 and f=0. The exponent e is biased by 127, i.e. they are represented by:

$$e_{\text{biased}} = e+127$$

Consider now the problem of fraction alignment using biased exponents and normalized operands to add two numbers.

Let $x_a = f_a \times 2^{e_a}$ and $x_b = f_b \times 2^{e_b}$ then

$$x_a + x_b = \begin{cases} (f_a \times 2^{e_a}) + (f_b \times 2^{e_b}) & \text{for } e_a = e_b \\ [f_a + 2^{-(e_a - e_b)}] \times 2^{e_a} & \text{for } e_a > e_b \\ [f_a \times 2^{-(e_a - e_b)} + f_b] \times 2^{e_b} & \text{for } e_a < e_b \end{cases} \qquad \text{(EQ 1.2)}$$

is a shifting factor that is multiplied by the fraction with the smaller exponent. Thus for $e_a > e_b$, f_a is added to the aligned (right shifted) f_b fraction, and the resulting fraction is characterized with the larger exponent. From the above discussion, the binary point of the two operands x_a and x_b must be aligned before addition can be performed. This is accomplished by comparing the relative magnitudes of the two exponents and shifting the fraction with the smaller exponent $|e_a - e_b|$ bit positions to the right. The addition of the fractions then proceeds with the larger exponent used as the exponent for the resulting fractional sum, and the resulting fraction has a value in the range $0 \le |f| < 2$.

1.4.5 File Formats

dsPICworks software stores all signal waveforms in disk files. Almost all signal processing operations of dsPICworks software involve reading or writing of signal values from and to disk files.

1.4.6 Basic File Types

Basic file types are identified by a suffix added to the filenames. There are four basic file types:

TABLE 1-12 File Types and Suffixes

File Type	Suffix
Time domain data	.TIM
Frequency domain data	.FRE
Filter coefficients	.FLT
Script File	.SCR

Time domain data fields are all real-valued signals with time values implicit. The signal starts at time 0 and the nth value in the file represents time nT where T is the time between samples (i.e. the reciprocal of the sampling frequency). Important information about the file such as the sampling frequency is maintained in a header record.

Frequency domain data files are all complex valued for a specified frequency value. The interval between frequency values is not stored in the data values, but is determined implicitly from the FFT length stored in the header record. The complex valued data are stored in rectangular form representing real and imaginary parts.

Filter coefficient files have several formats depending on the type of arithmetic, the realization method and whether the filter is an FIR or an IIR filter. See the dsPIC Filter Design manual for details.

1.4.7 Storage Format

*.TIM and *.FRE files can be created, stored and operated upon in binary or ASCII modes by dsPICworks software. However, ASCII files in signal processing applications require data conversion for any arithmetic operations. In contrast, data values within binary files

do not require conversion for arithmetic operations, but data values in these files cannot be displayed in editor windows. Processing of binary files is significantly faster than ASCII files. Data acquired from external sources is always in binary format.

It should be noted that although ASCII files can be displayed in editor windows they should not be edited. This is due to the fact that these files are not free format text files. Record alignment is carefully maintained to allow random access of any record in the files.

Irrespective of the format in which dsPICworks software creates *.TIM or *.FRE files, both file types may be exported to the external world in a variety of ASCII or Binary formats using the ***FILE**/EXPORT* option.

1.4.8 Header Record

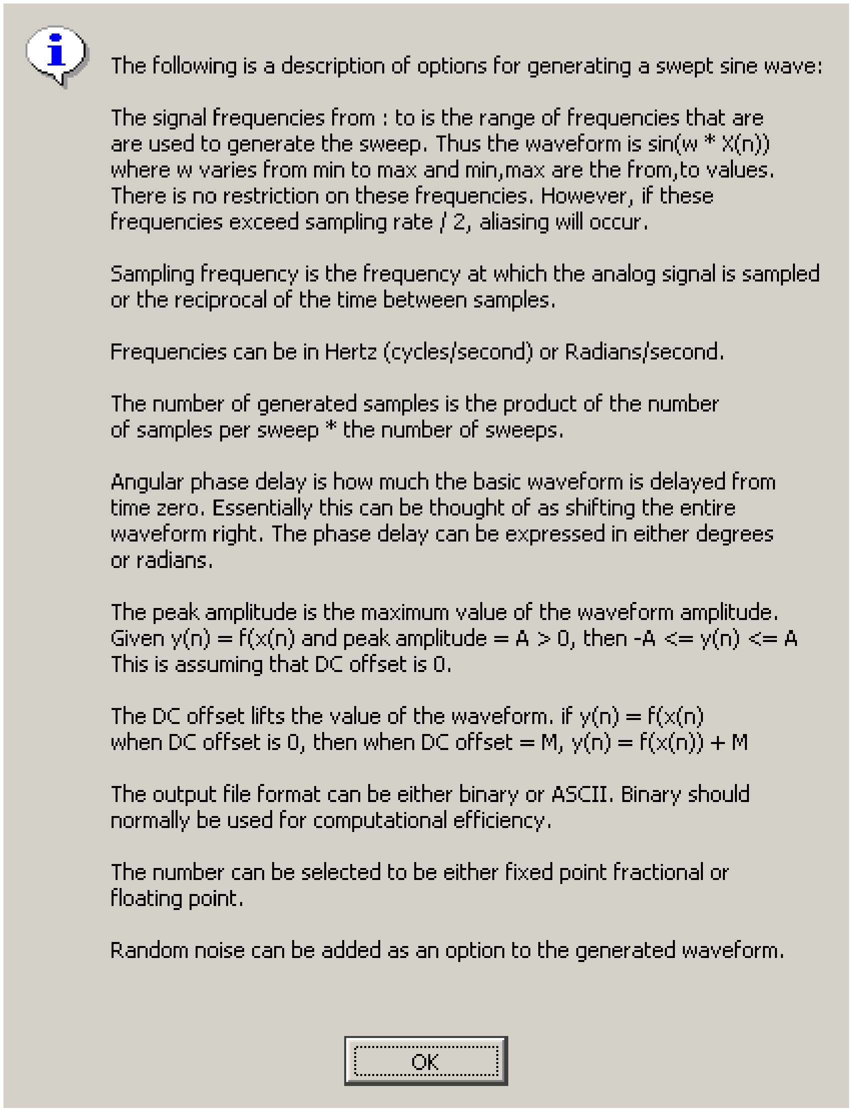
Every file has a header record which describes the file. The header record is written in ASCII even for a binary file. Information in the header record includes: storage format, number type, last operation, etc.

1.5 Help

Most dialog boxes have HELP facilities. There are two types of help available - a HELP button in the dialog box and balloon style help accessible via specific fields.

Clicking the HELP button will result in a HELP dialog box being displayed as shown in Figure 1.1 on page 12. This will describe the general function of the operation.

FIGURE 1.1 **Example of HELP Dialog Box**



The following is a description of options for generating a swept sine wave:

The signal frequencies from : to is the range of frequencies that are are used to generate the sweep. Thus the waveform is sin(w * X(n)) where w varies from min to max and min,max are the from,to values. There is no restriction on these frequencies. However, if these frequencies exceed sampling rate / 2, aliasing will occur.

Sampling frequency is the frequency at which the analog signal is sampled or the reciprocal of the time between samples.

Frequencies can be in Hertz (cycles/second) or Radians/second.

The number of generated samples is the product of the number of samples per sweep * the number of sweeps.

Angular phase delay is how much the basic waveform is delayed from time zero. Essentially this can be thought of as shifting the entire waveform right. The phase delay can be expressed in either degrees or radians.

The peak amplitude is the maximum value of the waveform amplitude. Given y(n) = f(x(n) and peak amplitude = A > 0, then -A <= y(n) <= A This is assuming that DC offset is 0.

The DC offset lifts the value of the waveform. if y(n) = f(x(n) when DC offset is 0, then when DC offset = M, y(n) = f(x(n)) + M

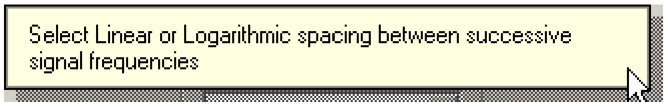
The output file format can be either binary or ASCII. Binary should normally be used for computational efficiency.

The number can be selected to be either fixed point fractional or floating point.

Random noise can be added as an option to the generated waveform.

If the Dialog Box features a question mark in the upper left corner, then the balloon type help is implemented and can be implemented by clicking on the question mark and dragging the help cursor to the area of interest and clicking. If balloon help is available for that specific area, it will then be displayed as shown in Figure 1.2 on page 12.

FIGURE 1.2 **Balloon Type Help**



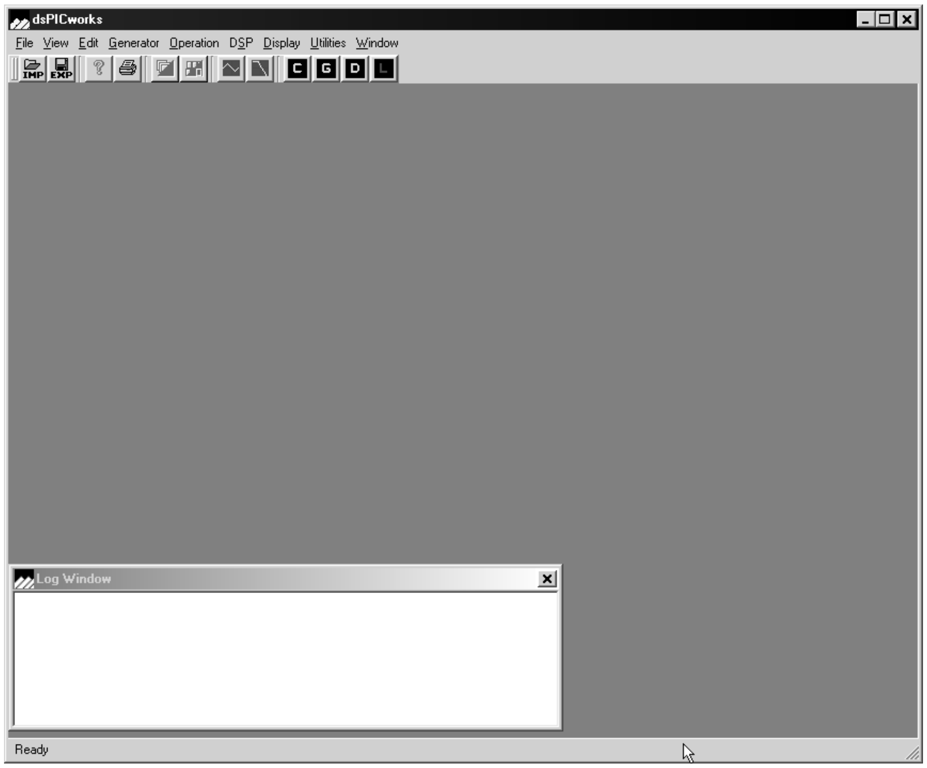
Select Linear or Logarithmic spacing between successive signal frequencies

1.6 Starting the system

Simply double-click on the dsPICworks icon on your Windows desktop to launch the system. Alternatively, choose “dsPICworks” from “Start Menu>>Programs>>MDS”.

The screen as shown in Figure 1.3 displays all the menu options.

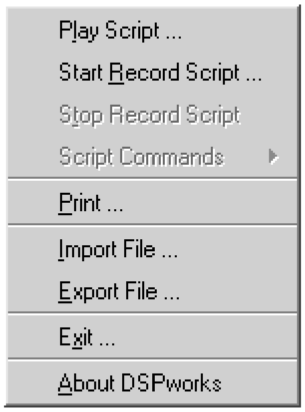
FIGURE 1.3 **dsPICworks software**


dsPICworks
File View Edit Generator Operation DSP Display Utilities Window
IMP
EXP
C
G
D
L
Log Window
Ready

CHAPTER 2 *File Menu*

The *FILE* menu as displayed in Figure 2.1 contains the scripting options which allow scripts to be recorded for repetitive operations, ***IMPORT*** and ***EXPORt*** file options as well as the standard ***PRINT, ABOUT*** and ***EXIT*** Functions.

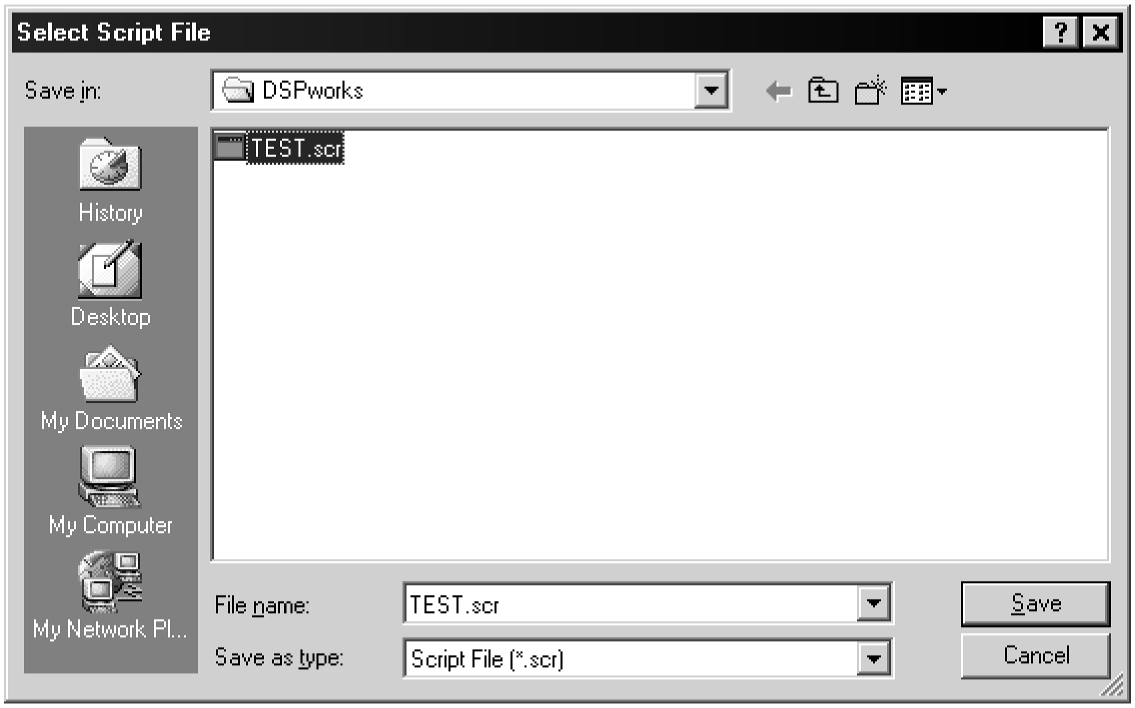
FIGURE 2.1 **File Menu**



2.1 Creating/Using Script Options

The ***FILE****/START RECORD SCRIPT/* option allows the user to record frequently used specifications in a script file suffixed by .SCR which may then be recalled as required by invoking the ***Play Script*** option. To start recording a script, select ***Start Record Script***. A standard file dialog box as shown in Figure 2.2 will appear with all existing Script extensions (.SCR).

FIGURE 2.2 Start Record Script



TEST.scr

File name: TEST.scr

Save as type: Script File (*.scr)

Enter the new name or select an existing filename to be overwritten. Upon selecting SAVE, dsPICworks software will enter script recording mode and the ***Start Record Script*** menu item will be grayed out to prevent nested scripts. Script recording will write all parameters to the output file in the correct format suitable for Play script in the following dsPICworks software functions: all menu items under ***GENERATOR, OPERATION, DSP, DISPLAY***; most items under *UTILITIES*.

Stop Record Script - terminates the script recording process. This button will be grayed out when the recording has been stopped.

Play Script - To play a previously recorded script, a standard open file dialog box will appear with all existing script extensions (.SCR). Select the appropriate filename and open it. When a script is in the process of being played, no user input is allowed until it has been completed. Note that to cause a file to be displayed while executing a script, explicitly use the ***Display*** menu item, or toolbar shortcut.

To cancel script playback, hold down the Escape key until the playback has stopped. If a long operation such as the cross-correlation is in progress, click on the cancel button in the progress window to cancel the operation, then hold down the Escape key to stop the script playback.

See SCRIPTS.DOC text file for documentation of file format for script files.

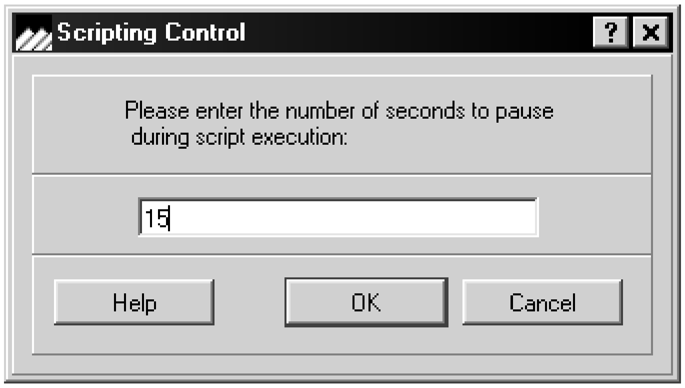
2.2 Script Commands

The following sub-menu selects script commands which can modify the control of execution when a script file is being executed.

2.2.1 Pause

This feature as shown in Figure 2.3 allows the user to enter the number of seconds for the system to pause during script execution.

FIGURE 2.3 Scripting Pause Feature


Scripting Control

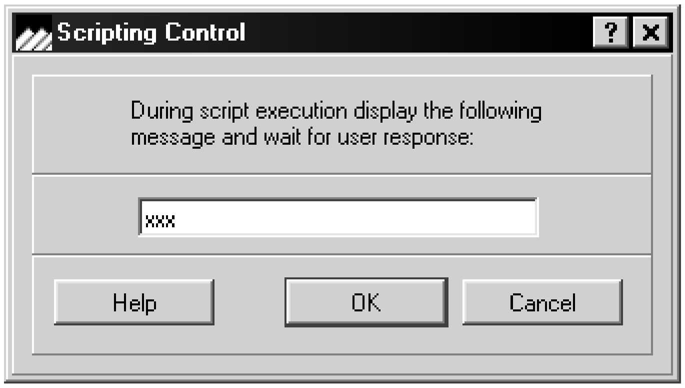
Please enter the number of seconds to pause during script execution:

15

2.2.2 Message

This option as shown in Figure 2.4 allows the display of a specific message which also requires a user response during execution.

FIGURE 2.4 Message Display during Scripting


Scripting Control

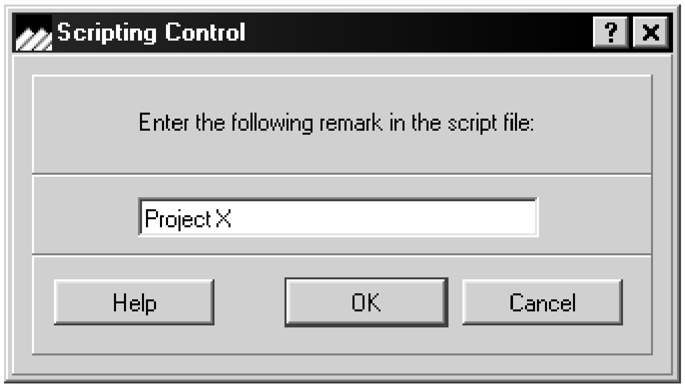
During script execution display the following message and wait for user response:

xxx

2.2.3 Remark

This feature as displayed in Figure 2.5 allows remarks to be entered in the script file. Remark lines in the script files are ignored during execution.

FIGURE 2.5 Script File Remarks Entry



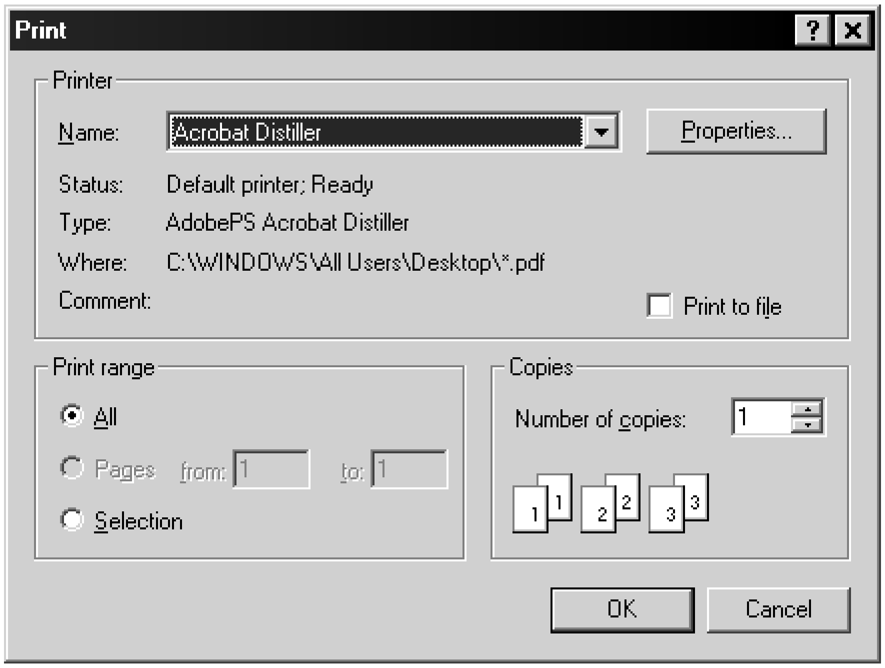
Enter the following remark in the script file:

Project X

2.3 Print Option

The ***PRINT*** option shown in Figure 2.6 option will cause the currently selected plot to be printed on the default printer. Note this dialog box may be Operating System dependent.

FIGURE 2.6 **Select Printer**



Print

Printer

Name: Acrobat Distiller

Status: Default printer; Ready

Type: AdobePS Acrobat Distiller

Where: C:\WINDOWS\All Users\Desktop\*.pdf

Comment:

☐ Print to file

Print range

[✓ All]

Pages from: 1 to: 1

Selection

Copies

Number of copies: 1

Printing options for 2-D and 3-D graphs are not available for this release of the application.

2.4 dsPICworks Software File Import and Export Features

dsPICworks software features a wide range of options to import and export data from and to the outside world. This makes it a very useful data analysis tool. This section will take a closer look at the featured import and export utilities, relative to the following topics:

- Time and Frequency-domain files
- Supported File Formats and Extensions
- Microchip MPLAB Compatibility
- Setting up dsPICworks software for File Import
- Setting up dsPICworks software for File Export

2.4.1 Time and Frequency-domain files

Since dsPICworks software features a variety of functions that operate on time and frequency domain data, the import/export utilities expect the user to specify whether the chosen data file is a time-domain or a frequency-domain file.

When importing data files that are specified as frequency-domain files by the user, dsPICworks software will assume that the file contains complex data samples that are stored as interleaved real and imaginary points. For both time and frequency-domain files, the user needs to specify the sampling rate while importing data. The Import utility also allows the user to import multiplexed / multi-channel data into the time-domain.Time-domain data files are imported into a dsPICworks file with the extension "*.TIM", whereas frequency-domain files are imported into a dsPICworks file with the extension "*.FRE".

2.4.2 Supported File Formats and Extensions

Data files that need to be imported or exported may have any file extension. The files that dsPICworks will recognize as being data files by default, have extensions - *.DAT or *.MCH. A drop-down list in the "Browse" dialog boxes, allows the user to choose "*.*" instead of the default file extensions. Time-domain data files are imported into a dsPICworks software waveform file with the extension "*.TIM", whereas frequency-domain files are imported into a dsPICworks waveform file with the extension "*.FRE".

Data in the file to be imported may be in many different formats. Likewise, data can be exported to files in many different formats.These depend on whether the data represent time or frequency domain samples. Table 2-1 on page 22 shows a list of valid combinations for import and export operations performed by dsPICworks software.

TABLE 2-1 **Supported Import / Export Operations: File Format v/s File Type**

	File Type	
File Format	**Time Files**	**Frequency Files**
Fractional / Integer Binary	Both	-
Fractional / Integer ASCII Decimal	Both	Import Only
Fractional / Integer ASCII Hexadecimal	Both	Import Only
Fractional / Integer ASCII Hex Multicolumn	Import Only	Import Only
Floating Point 32-bit ASCII Decimal	Both	Both
Floating Point 32-bit ASCII Hexadecimal	Both	Both
Floating Point 64-bit ASCII Decimal	Both	Both
Floating Point 64-bit ASCII Hexadecimal	Both	Both
Windows WAV file [8 and 16-bit support]	Both	-
8-bit Integer	Import Only	-
16-bit Integer	Import Only	-
16-bit Offset Integer	Import Only	-

1 An additional file type - QED filter file is supported on the export operation

2 Hexadecimal files can alternatively have dsPIC assembler directives prefixed to them in the export operation. In such a case, the export file is created with a *.s file extension and can be added into the user's MPLAB® IDE project or workspace.

The various file formats are described now, in further detail. The descriptions explicitly call out import and export operations for these file formats:

2.4.2.1 Fractional / Integer Binary

When used as the source file for an import operation, this file format is considered a byte stream with the least significant byte coming before the most significant byte. dsPICworks software will read the binary file and interpret the data as a stream of 1.15 fractional fixed-point samples. The number of samples to be imported is equal to the half the size of file.

When used as the destination file in an export operation, dsPICworks software will create a binary file from the time-domain waveform file. The file is a byte stream with the least significant byte coming before the most significant byte. If the time-domain source file comprised of floating-point samples, these are converted to appropriate integer values such that any floating point values larger than 1.0 in magnitude are saturated or wrapped around according to current settings in the Control Center. For multi-channel time-domain source files comprising of integer data, all samples at time nT are exported consecutively with channel 0 first.

2.4.2.2 Fractional / Integer ASCII Decimal

When this file format is used as a source file for an import operation, the file must contain samples in the 16-bit range [-32768, 32767]. The individual samples in the file are in decimal notation and delimited by a Carriage Return + Line Feed (CR+LF) combination. The number of samples to be imported is equal to the number of lines delimited by the CR+LF combination. Data from this file is actually read into a 32-bit integer data-type and wrapped or saturated according to the current settings in the Control Center, and eventually stored into a dsPICworks waveform file. The word "ASCII" refers to the fact that the file is of ASCII-text file type, as opposed to a binary file-type. The term "Fractional / Integer" refers to the fact that though the data samples are signed decimal integers, dsPICworks will treat them as 1.15 fractional numbers. An example of the contents of such a file is shown in Figure 2.7 on page 23.

Special support exists for importing multi-channel time-domain data files. In multi-channel data files, the number of lines should equal the number of samples per channel, i.e. all samples for time instant, nT, should be on the same line. These samples on the same line are delimited by ' ' (space), '" (single-quote), '"' (double quote) or ',' (commas) and each sample represents data from a different channel. Missing samples for multi-channel files are set to 0.

When this file format is used as a destination file for an export operation, dsPICworks software generates a file containing samples that are 6-character right aligned decimal. Floating point numbers are converted to appropriate integer values. Floating point values larger than 1.0 in magnitude are saturated or wrapped according to the settings in the Control Center. For multi-channel time files, all samples at the time nT are exported to the same line delimited by a comma.

FIGURE 2.7 **Fractional / Integer ASCII Decimal Data file Forma**

```
32000
12000
15000
7000
-12000
-32767
```

2.4.2.3 Fractional / Integer ASCII Hexadecimal

When this file format is used as a source file for an import operation, the file must contain samples in the 16-bit range [0x8000, 0x7FFF]. The samples in the file are in hexadecimal notation and delimited by a CR+LF combination. The number of samples to be imported equals number of lines delimited by the CR+LF combination. Data from this file is actually imported into a 32-bit integer data-type and wrapped or saturated according to the current settings in the Control Center and eventually stored into a dsPICworks waveform file. The word "ASCII" refers to the fact that the file is of ASCII-text file type, as opposed to a binary file-type. The term "Fractional / Integer" refers to the fact that though the samples are stored as 16-bit hexadecimal numbers, dsPICworks software will treat them as 1.15 fractional numbers. An example of the contents of such a file is shown in Figure 2.8 on page 24.

Special support exists for importing multi-channel data files. In multi-channel data files, the number of lines should equal the number of samples per channel, i.e. all samples for time instant, nT, should be on the same line. These samples on the same line are delimited by ‘ ‘ (space), ‘’’ (single-quote), ‘”’ (double quote) or ‘,’ (commas) and each sample represents data from a different channel. Missing samples for multi-channel files are set to 0.

When this file format is used as a destination file for an export operation, dsPICworks software will create a file containing 4-character hexadecimal samples. Floating point numbers are converted to appropriate integer values. Floating point values larger than 1.0 in magnitude are saturated or wrapped according to current settings in the Control Center. For multi-channel files, all samples at the time nT are exported to the same line delimited by a comma.

FIGURE 2.8 **Fractional / Integer ASCII Hexadecimal Data file Format**

```
AAAA
BBBB
CCCC
DDDD
EEEE
FFFF
. . . .
4567
5678
6789
789A
89AB
```

2.4.2.4 Fractional / Integer ASCII Hexadecimal Multi-column

When this file format is used as a source file for an import operation, the file must contain samples in the 16-bit range [0x8000, 0x7FFF]. The samples in the file are arranged 8 in a line and are in hexadecimal notation. Each line is delimited by a CR+LF combination. The number of samples to be imported therefore equals 8 times the number of lines delimited by the CR+LF combination. Data from this file is actually imported into a 32-bit integer data-type and wrapped or saturated according to the current system settings for wrap-around/saturation and eventually stored into a dsPICworks waveform file. The word “ASCII” refers to the fact that the file is of ASCII-text file type, as opposed to a binary file-type. The term “Fractional / Integer” refers to the fact that though the samples are stored as 16-bit hexadecimal numbers, dsPICworks software will treat them as 1.15 fractional numbers. The term “Multi-column” refers to the fact that the file contains rows of 8 samples each. Successive samples are tab or space-delimited. An example of the contents of such a file is shown in which shows the same samples shown in Figure 2.9 on page 25 except in multi-column format.

Special support exists for importing multi-channel data files. In multi-channel data files, the number of lines should equal the number of samples per channel, i.e. all samples for time instant, nT, should be on the same line. These samples on the same line are delimited by ‘ ‘ (space), ‘” (single-quote), ‘”’ (double quote) or ‘,’ (commas) and each sample represents data from a different channel. Missing samples for multi-channel files are set to 0.

This file format has been provided to allow importing any file exported from MPLAB IDE into dsPICworks software. The dsPICworks export utility does not support creating multi-column files.

FIGURE 2.9 **Fractional / Integer ASCII Hexadecimal Multi-Column Data file Format**

AAAA	BBBB	CCCC	DDDD	EEEE	FFFF	1111	2222
3333	4444	5555	6666	7777	8888	9999	0000
1234	2345	3456	4567	5678	6789	789A	89AB

2.4.2.5 Floating Point 32-bit ASCII Decimal

When this file format is used as a source file for an import operation, the file must contain CR+LF delimited samples in the 32-bit IEEE floating-point range [-3.402823466e+38, 3.402823466e+38] with a resolution of 1.175494351e-38. Data from this file is actually imported into a 64-bit floating-point data-type, clipped back to a 32-bit floating-point data type and eventually stored into a dsPICworks waveform file. Denormalized numbers are set to 0. The samples in the file are in decimal notation, for e.g., 1.0000000e+000. Tabs and spaces are allowed before the number and any characters are allowed after the number. The word “ASCII” refers to the fact that the file is of ASCII-text file type, as opposed to a binary file-type. Missing samples for multi-channel files are set to 0. An example of the contents of such a file is shown in Figure 2.10 on page 25.

When this file format is used as a destination file for an export operation, dsPICworks software will create a file identical in format to the source file used in the import operation. A few special cases exist when creating files of this format on an export operation:

- 16-bit integer single channel file: the integer number is converted to a 32-bit floating point number by dividing by 32768. Thus all output values x are in the range: $-1 <= x < 1$
- 16-bit integer multichannel file: The integer number is converted to a 32-bit floating-point number by dividing by 32678. Thus all output values x are in the range $-1 <= x < 1$ and all samples at time nT are exported to the same output line delimited by a comma.
- For frequency files, the real and imaginary values are shown on the same line.

FIGURE 2.10 **Floating Point 32-bit ASCII Decimal Data file Format**

```
2.5E+10
-2.3E-2
2.3789E+10
1.0E+1
9.876E-1
2.2E-2
```

2.4.2.6 Floating Point 32-bit ASCII Hexadecimal

When this file format is used as a source file for an import operation, the file must contain CR+LF delimited samples in the 32-bit IEEE floating-point range [-3.402823466e+38, 3.402823466e+38] with a resolution of 1.175494351e-38. Data from this file is actually imported into a 64-bit floating-point data-type, clipped back to a 32-bit floating-point data type and eventually stored into a dsPICworks waveform file. Denormalized numbers are set to 0. The samples in the file are in hexadecimal notation, so the range as expressed in hexadecimal would be [0xFF7FFFFF, 0x7F7FFFFF]. It should be noted that the 32-bit hexadecimal samples may be specified in the data file as one entire 32-bit sample per line or one sample for each of two consecutive lines, i.e. 4 hexadecimal digits per line and 8 hex digits per sample. In either case, the sample must be specified as follows: Consider the example of the 32-bit IEEE floating point sample of +1.5925 (=0x3FCBD70A). It may be written in the data file as "D70A3FCB" in one line or as "D70A" and "3FCB" in successive lines, in that order. This order is compatible with files exported from Microchip's MPLAB Integrated Development Environment. The word "ASCII" refers to the fact that the file is of ASCII-text file type, as opposed to a binary file-type. An example of the contents of such a file are shown in Figure 2.11 on page 26. The values used in Figure 2.11 are hexadecimal equivalents of the values used in Figure 2.10 on page 25.

When this file format is used as a destination file for an export operation, dsPICworks software will create a file identical in format to the source file used in the import operation. Each floating point value in hexadecimal is represented by two entries of 4 hexadecimal digits, each with the least significant entry first i.e. the 4 hexadecimal digits of the lower 16-bits is the first entry. The hexadecimal digits within each 16-bit word are in big-endian order i.e. most significant bit (digit) first.

If multi-channel output is specified, each channel has two entries of 4 hexadecimal digits each, with channel 1 following channel 0 and so on. A few special cases exist when creating files of this format on an export operation:

- 16-bit integer single channel file: the integer number is converted to a 32-bit floating point number by dividing by 32768. Thus all output values x are in the range: -1 <= x < 1
- 16-bit integer multichannel file: The integer number is converted to a 32-bit floating-point number by dividing by 32678. Thus all output values x are in the range -1 <= x < 1 and all samples at time nT are exported to the same output line delimited by a comma.
- For frequency files, the real and imaginary values are shown on the same line.

FIGURE 2.11 Floating Point 32-bit ASCII Hexadecimal Data file Format

```
43B7
50BA
6A7F
BCBC
3DEB
50B1
0000
4120
D35B
3F7C
```

FIGURE 2.11 **Floating Point 32-bit ASCII Hexadecimal Data file Format**

```
3958
3CB4
```

2.4.2.7 Floating Point 64-bit ASCII Decimal

dsPICworks software provides some limited support to import and export data from and to the 64-bit IEEE floating-point format.

If this file format is specified as the source for an import operation, dsPICworks software will read the 64-bit data and clip it to the 32-bit IEEE floating-point format prior to storing in the waveform file. Denormalized numbers are set to 0. The samples in the file are in decimal notation, for e.g., 1.00000000000000e+000. The word "ASCII" refers to the fact that the file is of ASCII-text file type, as opposed to a binary file-type. Missing samples for multi-channel files are set to 0.

If this file format is specified for the destination file in an export operation, dsPICworks software will store data in the 64-bit IEEE floating-point format. However, the magnitude of the data will not exceed the range of 32-bit IEEE floating-point range. A few special cases exist when creating files of this format on an export operation:

- 16-bit integer single channel file: the integer number is converted to a 64-bit floating point number by dividing by 32768. Thus all output values x are in the range: $-1 <= x < 1$
- 16-bit integer multichannel file: The integer number is converted to a 64-bit floating-point number by dividing by 32678. Thus all output values x are in the range $-1 <= x < 1$ and all samples at time nT are exported to the same output line delimited by a comma.
- For frequency files, the real and imaginary values are shown on the same line.

2.4.2.8 Floating Point 64-bit ASCII Hexadecimal

dsPICworks software provides some limited support to import and export data from and to the 64-bit IEEE floating-point format.

If this file format is specified as the source for an import operation, dsPICworks software will read the 64-bit data and clip it to the 32-bit IEEE floating-point format prior to storing in the waveform file. Denormalized numbers are set to 0. The samples in the file are in hexadecimal notation. Each double precision value in hexadecimal, should be represented by four entries of 4 hexadecimal digits each with the least significant entry first, i.e. the 4 hexadecimal digits or the lowest 16-bits is the first entry. The hexadecimal digits in each 16-bit word should be in big-endian order i.e. most significant bit (digit) first. The word "ASCII" refers to the fact that the file is of ASCII-text file type, as opposed to a binary file-type. Missing samples for multi-channel files are set to 0.

If this file format is specified for the destination file in an export operation, dsPICworks software will store data in the 64-bit IEEE floating-point format. However, the magnitude of the data will not exceed the range of 32-bit IEEE floating-point range. Each double precision value in hexadecimal, is represented by four entries of 4 hexadecimal digits each with the least significant entry first, i.e. the 4 hexadecimal digits or the lowest 16-bits is the first entry. The hexadecimal digits in each 16-bit word are in big-endian order i.e. most significant bit (digit) first. If multi-channel output is specified, each channel has four entries of 4 hexadecimal digits each with channel 1 following channel 0 and so on. A few special cases exist when creating files of this format on an export operation:

- 16-bit integer single channel file: the integer number is converted to a 64-bit floating point number by dividing by 32768. Thus all output values x are in the range: $-1 <= x < 1$
- 16-bit integer multichannel file: The integer number is converted to a 64-bit floating-point number by dividing by 32678. Thus all output values x are in the range $-1 <= x < 1$ and all samples at time nT are exported to the same output line delimited by a comma.
- For frequency files, the real and imaginary values are shown on the same line.

2.4.2.9 Windows Wave File

When importing data from a file of this format, dsPICworks software will read the number of bits/sample, number of samples, and sampling rate from the WAV file. The Sampling rate may be overwritten by user. Exporting data (multi-channel data included) using this file format will create a Windows *.WAV file. Both 8-bit and 16-bit WAV files may be created.

2.4.2.10 8-bit Integer

This is a signed 8-bit value which is converted to a 16-bit signed integer on import.

2.4.2.11 Offset 8 and 16-bit Offset Integer

Offset binary is defined as number range $[0,2^8-1\}$ or $[0,2^{16}-1]$ for 8-bit binary offset and 16-bit binary offset respectively. The zero point is midway between the range. Thus the negative numbers appear as positive numbers in this number reprsentation. The system converts both 8-bit binary offset and 16-bit binary offset to 16-bit signed integers.

2.4.3 Microchip MPLAB® IDE Compatibility

Microchip’s MPLAB IDE (versions 6.32 and higher) provides a mechanism to import data from external files to the dsPIC device data memory and a mechanism to export data from the dsPIC device data memory into external files. These mechanisms are known as the "Import Table" and "Export Table" options, which are available on right-clicking on the File-Register window.

2.4.3.1 Export Table Option

Right-click on the File-Register window after selecting a range of addresses and choose the "Export Table" option. An "Export As" dialog box pops up on the screen allowing the user to (optionally) specify the address range, the file format and the file extension.

The default file format is "ASCII hexadecimal single-column". An alternative that is allowed is the "ASCII hexadecimal Multi-column" format. To obtain the Multi-column format file, simply un-check the box named “Single-Column Output”

The default extension for the created file is *.MCH and can be over-ridden to allow any other extension, for example - *.dat, *.foo, *.myextension etc.

The default address range is that selected on the File-Register window but this can be over-ridden by the user, as well.

2.4.3.2 Import Table Option

Right-click on the File-Register window after selecting a starting address and choose the "Import Table" option. An "Import" dialog box pops up on the screen allowing the user to (optionally) specify the starting address and the file extension.

Both "ASCII hexadecimal single-column" and the "ASCII hexadecimal Multi-column" format files may be imported without having to specify the file format explicitly.

The default extension for the file to be imported is *.MCH and can be over-ridden to allow any other extension, for example - *.dat, *.foo, *.myextension etc.

The default starting address is that selected on the File-Register window but this can be over-ridden by the user, as well.

2.4.3.3 Data Arrangement in Memory

Assume Figure 2.8 on page 24 and Figure 2.9 on page 25 represent data exported out of the MPLAB IDE.

Note that in the set of samples shown in Figure 2.8 on page 24, the 16-bit data word "AAAA" resides in the lower-most address in data memory (RAM or EEPROM), while the 16-bit data word "89AB" resides in the highest address in data memory. In this file listing above, the word "AAAA" is followed in memory by the word "BBBB" and so on until "789A" and eventually "89AB".

Also note that though data is stored in "Little-Endian" format within the dsPIC™ device, the files exported from MPLAB IDE show each individual data word in "Big-Endian" format. For instance, in the case of data word "89AB", "89" is the Most Significant Byte while "AB" is the least significant byte within the 16-bit data word. The data byte "89" occupies the higher byte-address while the byte "AB" occupies the lower byte address. Consecutive 16-bit words are however shown in Little-Endian format.

Both complex-valued data and IEEE floating point data can be exported from MPLAB IDE in the same manner as exporting real-valued 16-bit integer/fractional data.

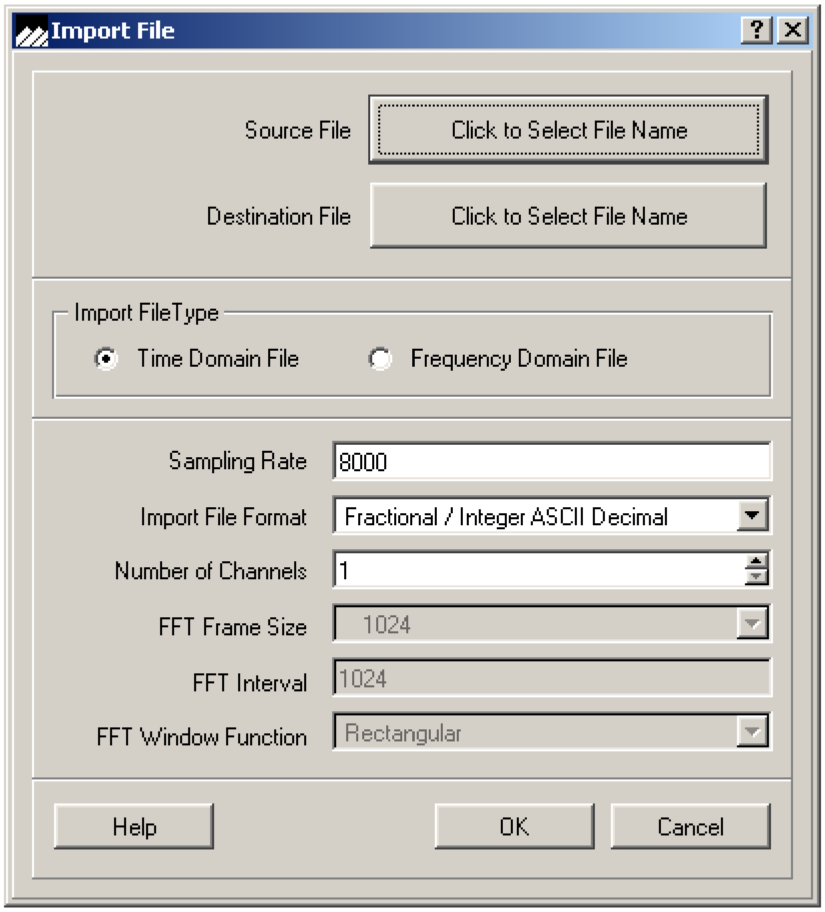
For instance, if the file-listing in Figure 2.8 on page 24 was the output data from a complex FFT routine, then the first complex sample would be "AAAA + jBBBB", where "BBBB" was the imaginary part of the first frequency bin.

Similarly, if the file-listing in Figure 2.8 on page 24 was an array of IEEE 32-bit floating point numbers, then the first 32-bit floating-point sample would be "BBBBAAAA" (MSBit to LSBit) and that would represent the floating-point number "5.7271319e-3".

Further, if the file-listing in Figure 2.8 on page 24 was an array of IEEE 64-bit floating point numbers, then the first 64-bit floating-point sample would be "DDDDCCCCBBBBAAAA" (MSBit to LSBit) and that would represent the floating-point number "-1.4535641191212697e+144".

2.4.4 Setting up dsPICworks Software for File Import

FIGURE 2.12 **Import File**



Import FileType: [✓ Time Domain File] Frequency Domain File

Sampling Rate	8000
Import File Format	Fractional / Integer ASCII Decimal
Number of Channels	1
FFT Frame Size	1024
FFT Interval	1024
FFT Window Function	Rectangular

The user should follow the steps below in sequence while importing files into dsPICworks software:

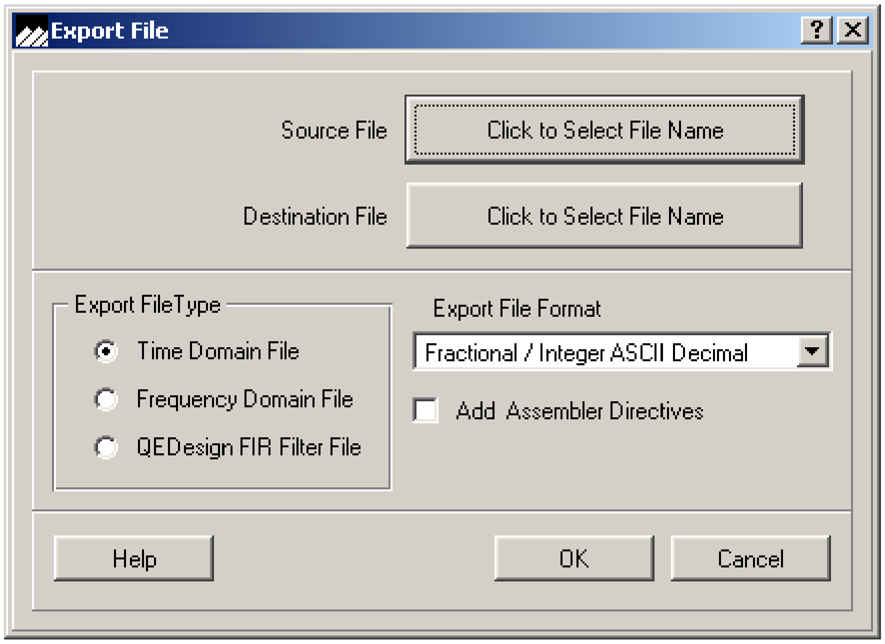
1. Click on the ***FILE/****IMPORT* option or the "Import" icon on the toolbar.
2. In the Import dialog box, select the file type - ***Time or Frequency***.
3. If time files were selected, then choose the ***Sampling Rate*** and the ***Number of Channels***.
4. If frequency files were selected, then choose the ***Sampling Rate, FFT Frame Size, Interva***l and F***FT Window Function***.
5. Select the appropriate file format - for example, "Fractional / Integer ASCII Hexadecimal", from the drop-down list of files.
6. Select the source file (*.MCH, *.DAT, *.*) from the source file browse window dialog.
7. Select the destination file name (*.TIM or *.FRE) and path from the destination file browse window dialog.
8. Finally, click on OK.

Note that selecting the source and destination files must be performed only after the file type (time or frequency) and the file format (e.g. Fractional/Integer ASCII Decimal) have been selected.

The imported file should display on a window within dsPICworks software.

2.4.5 Setting up dsPICworks Software for File Export

FIGURE 2.13 **Export File**



Export FileType
- (●) Time Domain File
- ( ) Frequency Domain File
- ( ) QEDesign FIR Filter File

Export File Format: Fractional / Integer ASCII Decimal

[ ] Add Assembler Directives

The user should follow the steps below in sequence while exporting files from dsPICworks:

1. Click on the ***FILE/EXPORT*** option or the “Export” icon on the toolbar.
2. In the Export dialog box, select the file type - ***Time, Frequency*** or ***QEDesign FIR Filter File***.
3. Select whether or not the exported file needs to be a dsPIC assembler file.
4. Select the appropriate file format - for example, “Fractional / Integer ASCII Hexadecimal”, from the drop-down list of files.
5. Select the source file (*.TIM, *.FRE) from the source file browse window dialog.
6. Select the destination file name (*.DAT, *.MCH, *.*) and path from the destination file browse window dialog.
7. Finally, click on OK.

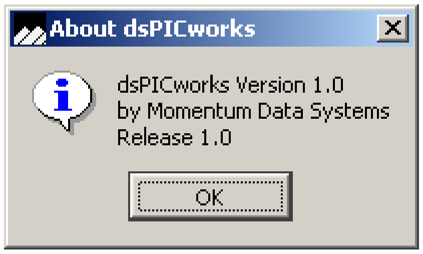
Note that selecting the source and destination files must be performed only after the file type (time or frequency) and the file format (e.g. Fractional/Integer ASCII Decimal) have been selected.

The exported file should be available for use in the selected folder.

2.5 About

This feature as displayed in Figure 2.14 provides information basic system information including the Version number which will be required in the event of any support being required.

FIGURE 2.14 About dsPICworks



dsPICworks Version 1.0
by Momentum Data Systems
Release 1.0

CHAPTER 3 *View Menu*

The ***VIEW*** menu allows the user the option to view the toolbar and status bars as shown in Figure 3.1.

FIGURE 3.1 **View Menu Options**

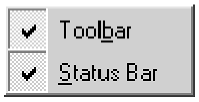
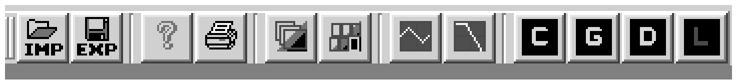


FIGURE 3.2 **Toolbar**



The toolbar as shown in Figure 3.2 contains the shortcuts to the following menu options (in order of appearance on toolbar):

- Import - for further information please refer to Section 2.4.4 on page 30
- Export- for further information please refer to Section 2.4.5 on page 31
- Help- for further information please refer to Section 1.5 on page 11
- Print- for further information please refer to Section 2.3 on page 20
- Cascade- for further information please refer to Section 10.1.1 on page 128

- Tile- for further information please refer to Section 10.1.1 on page 128
- Display Time File- for further information please refer to Section 8.1 on page 111
- Display 1D Frequency File- for further information please refer to Section 8.2 on page 112
- Control Center- for further information please refer to Section 9.1 on page 118
- Graph Control Center- for further information please refer to Section 10.1.4 on page 131
- Display Control- for further information please refer to Section 10.1.3 on page 129
- Log Window- for further information please refer to Section 10.1.2 on page 128

FIGURE 3.3 **Status Bar**

The system's status is indicated as shown in Figure 3.3.

CHAPTER 4 *Edit Menu*

The Edit menu allows the user the ability to graphically cut, delete, copy and paste segments of a time domain signal. The left mouse button is used to mark a segment of a graph. Single points are indicated by a crosshair.

Segments with more than one point are indicated with highlighting. Holding the left button down and dragging the mouse across the graph will highlight the desired segment. To extend a segment, hold the shift key down and drag mouse to desired position. To extend a segment which exceeds the graph window, advance the plot via the scrollbar, hold the shift key down, and then drag the mouse to the desired position. Note that the highlighted region X and Y coordinates and deltas in these coordinates will be displayed in the Tracking cursor co-ordinate dialog box.

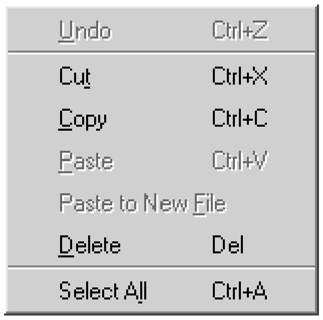
The right mouse button can also be used to mark segments for graphical editing. The right mouse button displays a resizable rectangle which when released will highlight the graph segment within the rectangle. The right mouse button also reads out tracking values in the Tracking cursor co-ordinate dialog box.

Segments which are copied or cut are placed in the file CLPBOARD.TIM. Segments which are pasted into files come from the CLPBOARD.TIM file.

4.1 Edit Menu

The *EDIT* Menu as shown in Figure 4.1 features standard edit menu options.

FIGURE 4.1 **Edit Menu Options**



For destructive operations such as Cut, Delete and Paste, the original file is renamed with a BAK extension. It is possible to rename the backup file to a TIM extension using either DOS commands or the Windows Environment. In case there is not sufficient disk space for these edit operations to complete, a dialog box will display a message stating that there is insufficient disk space. In this case the original file will have a BAK extension and the new signal file will not be written.

4.1.1 Undo

This function undoes the last graphical editing operation. This operation deletes the existing file and renames the backup file with the BAK extension to the TIM extension.

4.1.2 Cut

This function cuts the highlighted region of the waveform of the active graph window and places it in the CLPBOARD.TIM file. The original file is saved in a file with the same name but with the BAK extension.

Note: you cannot cut an entire waveform. If the intention is to delete a file, simply do this externally to the system by deleting the file. The system requires at least three points be retained in the file.

4.1.3 Copy

This function copies the highlighted region of a waveform of the active graph window and places it in the CLPBOARD.TIM file.

4.1.4 Paste

This function pastes the contents of the CLPBOARD.TIM file into the marked position of the of the active graph window. The original file is saved in a file with the same name but with the BAK extension.

4.1.5 Paste to a New File

This function pastes the contents of the CLPBOARD.TIM file into a new file.

4.1.6 Delete

This function deletes the highlighted segment of the active graph window. The original file is saved in a file with the same name but with the BAK extension.

Note: You cannot delete an entire waveform. If the intention is to delete a file, simply do this externally to the system by deleting the file. The system requires **at least three points** be retained in the file.

4.1.7 Examples of Highlighting Graph Windows

FIGURE 4.2 Waveform Display Prior to Highlighting

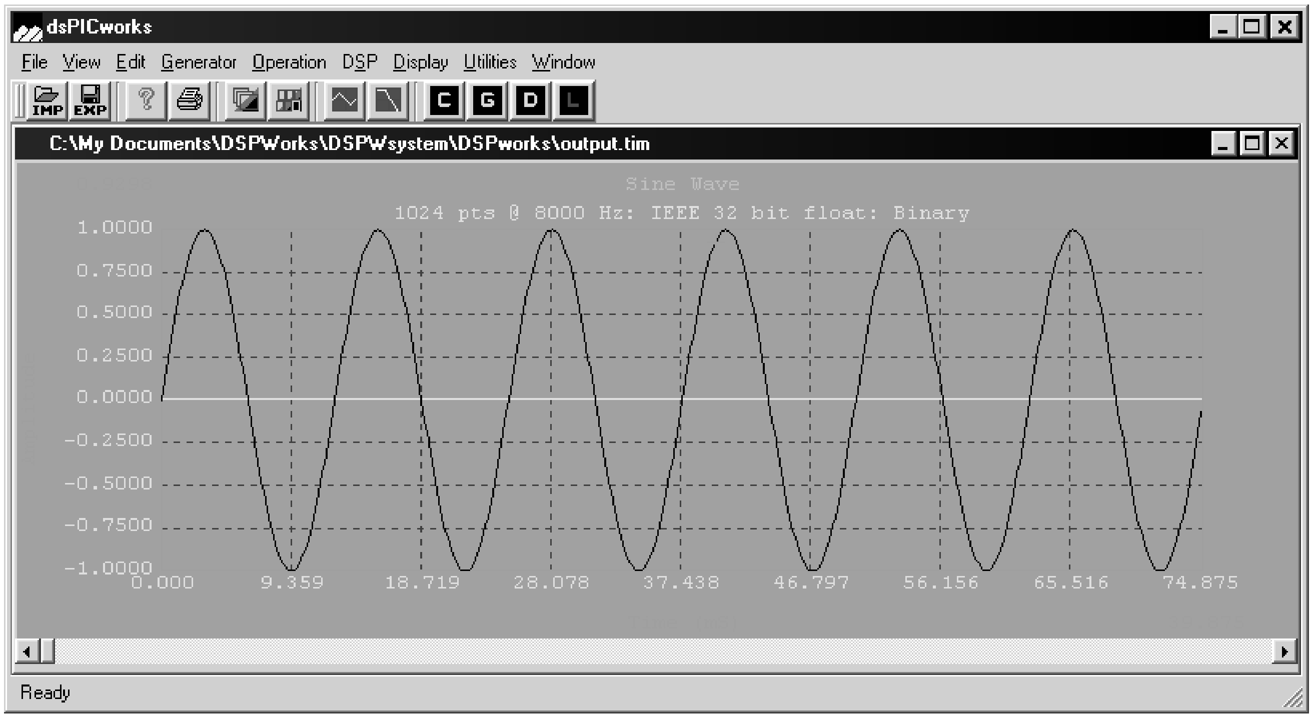

dsPICworks
File View Edit Generator Operation DSP Display Utilities Window
IMP EXP
C G D L
C:\My Documents\DSPWorks\DSPWsystem\DSPworks\output.tim
Sine Wave
1024 pts @ 8000 Hz: IEEE 32 bit float: Binary
1.0000
0.7500
0.5000
0.2500
0.0000
-0.2500
-0.5000
-0.7500
-1.0000
0.000
9.359
18.719
28.078
37.438
46.797
56.156
65.516
74.875
Ready

FIGURE 4.3 Waveform Display After Highlighting

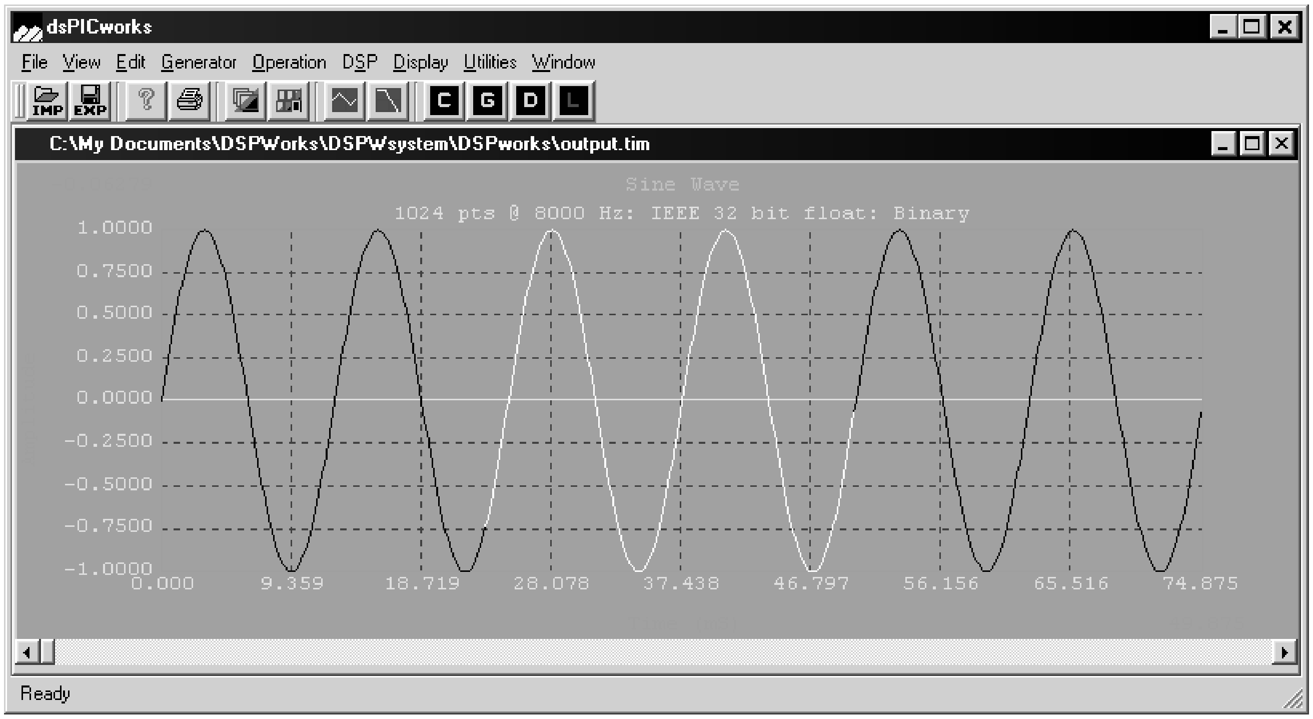
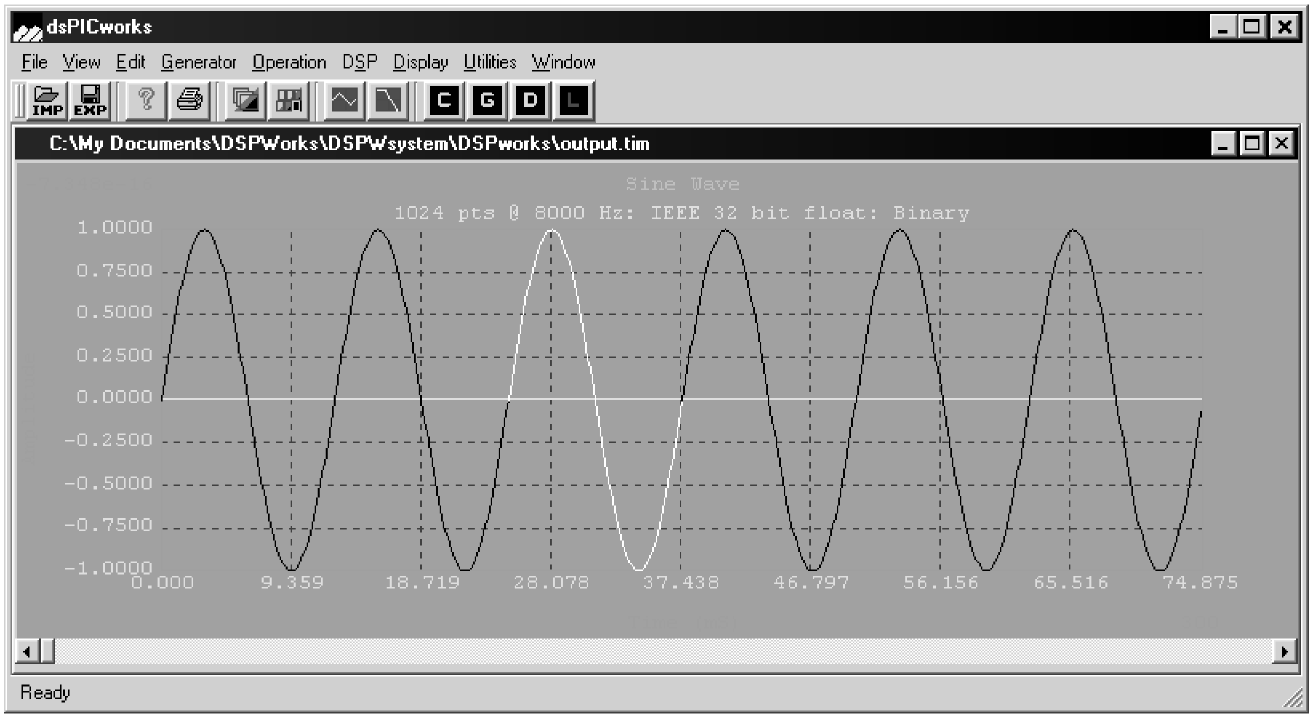

dsPICworks
File View Edit Generator Operation DSP Display Utilities Window
IMP EXP C G D L
C:\My Documents\DSPWorks\DSPWsystem\DSPworks\output.tim
Sine Wave
1024 pts @ 8000 Hz: IEEE 32 bit float: Binary
1.0000
0.7500
0.5000
0.2500
0.0000
-0.2500
-0.5000
-0.7500
-1.0000
0.000
9.359
18.719
28.078
37.438
46.797
56.156
65.516
74.875
Ready

FIGURE 4.4 Example of a Sine wave with one cycle highlighted

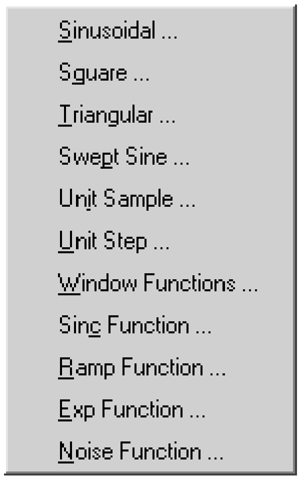

dsPICworks
File View Edit Generator Operation DSP Display Utilities Window
IMP EXP C G D L
C:\My Documents\DSPWorks\DSPWsystem\DSPworks\output.tim
Sine Wave
1024 pts @ 8000 Hz: IEEE 32 bit float: Binary
1.0000
0.7500
0.5000
0.2500
0.0000
-0.2500
-0.5000
-0.7500
-1.0000
0.000
9.359
18.719
28.078
37.438
46.797
56.156
65.516
74.875
Ready

CHAPTER 5 *Generator menu*

The ***GENERATOR*** menu provides the waveform synthesis section capability of dsPICworks software.

The dsPICworks ***GENERATOR*** menu as shown in Figure 5.1 has the capability of synthesizing various waveforms. Samples from the synthesized waveform are stored in files specified by the user. The sections that follow will describe waveform generation in greater detail.

FIGURE 5.1 **Generator menu**



The sinusoidal dialog box is described in considerable detail. The fields contained in this dialog box are common to all the generator functions and the user is referred to sinusoidal for more detailed explanations.

5.1 Sinusoidal

The *SINUSOIDAL* menu option allows the generation of a discrete time sinusoidal signal with quantized amplitude values.

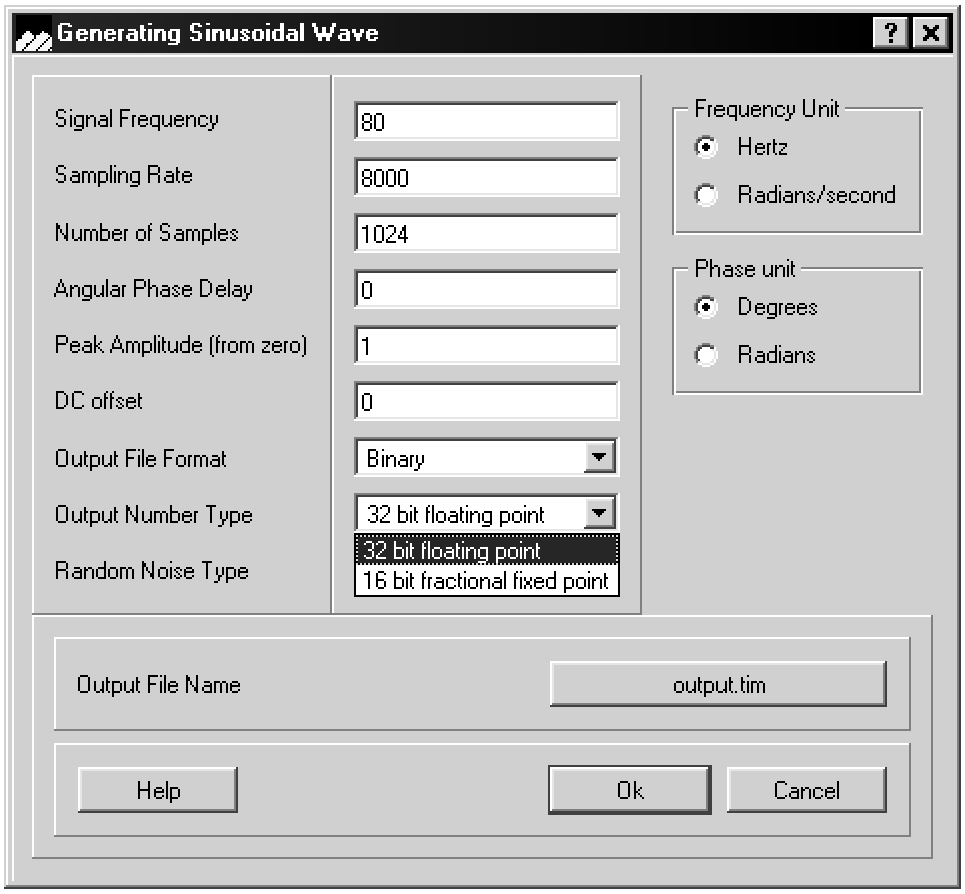
A discrete-time sinusoidal signal is expressed as $y(n) = A \times \sin(2\pi fnT + \Theta)$ where A is the amplitude, f is the frequency in Hertz (cycles per second), n is the sample number, T is the time between samples and Θ is the phase delay expressed in degrees or radians. Note that the sampling period T is related to the sampling frequency f_S by $T = \frac{1}{f_s}$.

The formula for y(n) can be easily derived from the continuous time formula $y(t) = \sin(2\pi ft + \theta)$ where t is replaced by nT. Frequently the T is omitted in the equations for discrete time formula. Thus T is understood in y(n) but to make the formula explicit, the T is present in $\sin(2\pi fnT + \theta)$ as this is the formula used for calculating y(n).

If frequencies are expressed in radians per second, then $2\pi f$ is replaced by ω.

The dialog box as shown in Figure 5.2 appears for the parameters of the sinusoidal waveform.

FIGURE 5.2 Sinusoidal Generator



Generating Sinusoidal Wave

Signal Frequency: 80
Sampling Rate: 8000
Number of Samples: 1024
Angular Phase Delay: 0
Peak Amplitude (from zero): 1
DC offset: 0
Output File Format: Binary
Output Number Type: 32 bit floating point
Random Noise Type: 32 bit floating point / 16 bit fractional fixed point

Frequency Unit: [✓ Hertz] Radians/second

Phase unit: [✓ Degrees] Radians

Output File Name: output.tim

Signal frequency and ***sampling rate*** are entered either in Hertz or Radians/second, depending on the frequency selection. The sampling frequency should be at least twice the signal frequency, however, the system does not enforce this. Click on the Frequency unit field to display the Hertz or Radians/second options and select the desired option.

The ***number of sample*** points is arbitrary. However, if a large number is specified, the system disk may fill up.

The ***output file name*** specifies the name of the file in which the sinewave will be written.

Angular phase delay is the offset Θ in degrees or radians. To generate a cosine wave, specify the phase delay as $\frac{\pi}{2}$ radians or 90^{o}.

Zero to peak amplitude specifies the maximum value that the sinusoidal wave can attain. For 32-bit floating point, this is any real positive number. For 16-bit fractional fixed point, this value must be > 0 and <1. Note that the largest number that can be represented in 16-bit fractional fixed point format is $1 - 2^{-15}$. The hexadecimal equivalent of this number is 7FFF.

DC offset is a number that shifts the waveform up or down in the magnitude direction. Note that fractional fixed point numbers cannot be larger than 1 in absolute value. Thus the sum of DC offset and the zero to peak amplitude must not exceed 1.0 in absolute value for fractional fixed point.

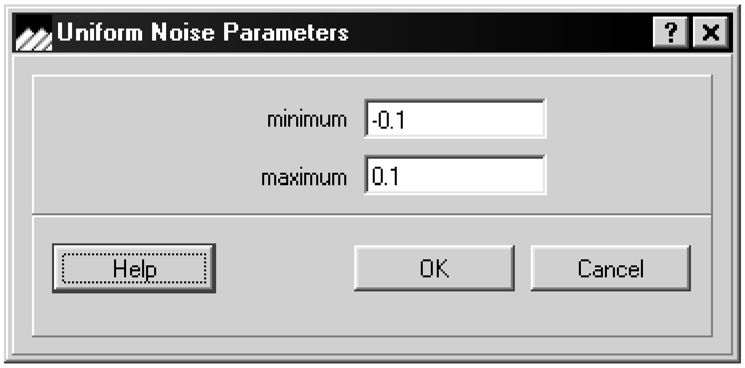
Output files can be generated in either ASCII or binary format. dsPICworks software will process data in binary files much faster than ASCII files as binary files are considerably more compact than ASCII files. However, the data values for the binary format cannot be viewed in any editor window.

The ***Output number types*** field specifies whether the samples recorded in the generated file are in 16-bit fixed-point fractional format or in 32-bit IEEE floating point format. For 16-bit fractional fixed point format, all samples will be in the range [-1,+1).

The ***Random Noise*** field provides users the ability to optionally add either Normal (Gaussian) noise or Uniform noise to the Sinusoidal waveform.

If ***Uniform Noise*** was selected, the dialog box shown in Figure 5.3 is displayed. The min and max fields specify the minimum and maximum noise values. All values between these extremes have equal probability of occurring. Uniform noise is specified by a maximum and minimum value from which its mean and variance can easily be determined. Click on OK to return to the Generating Sinusoidal Waveform window.

FIGURE 5.3 **Uniform Noise Parameters**

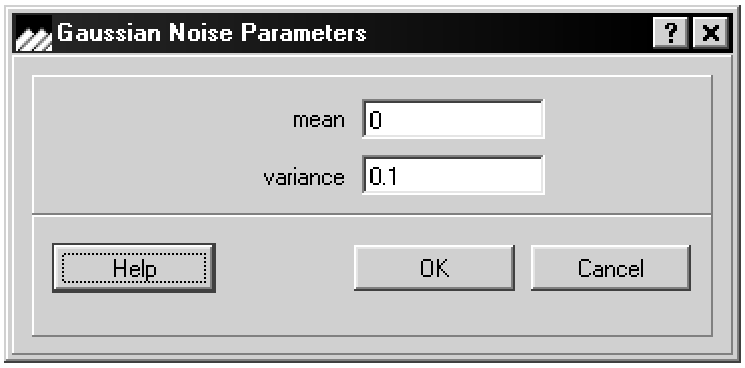

Uniform Noise Parameters

minimum -0.1

maximum 0.1

If ***Gaussian Noise*** was selected, the dialog box shown in Figure 5.4 is displayed. Enter appropriate values for Variance and Mean fields. Gaussian noise is completely defined by the mean and variance. Click OK to return to the Generating Sinusoidal Waveform window.

FIGURE 5.4 **Gaussian Noise Parameters**


Gaussian Noise Parameters

mean 0

variance 0.1

When adding noise content to the sinusoidal signal, the User must note a special property of the fixed-point fractional data type. It is possible that when noise is added to the sinusoidal waveform, the amplitude exceeds the bounds of fixed-point fractional numbers, i.e. [-1,+1) If this occurs, for 16-bit fractional data type, the waveform samples will either be saturated or wrapped-around depending on the selection in the **Utilities** menu.

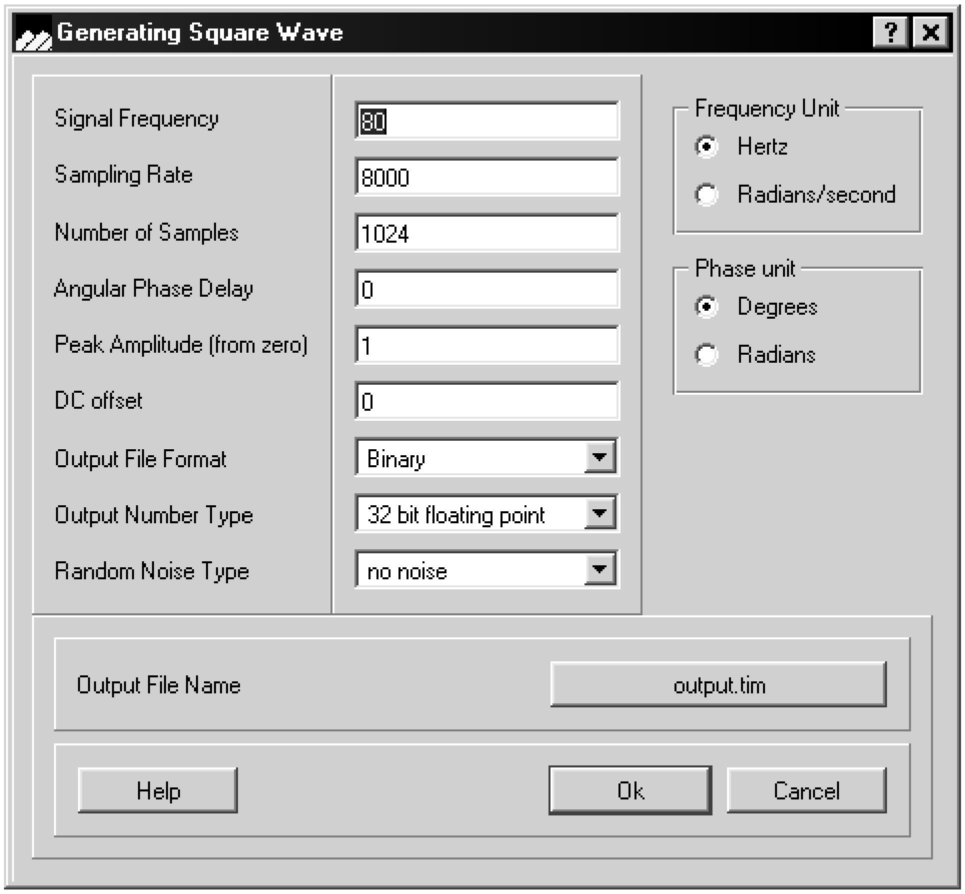
Once all the required values have been entered in the sinusoidal generator dialog box, click on ‘OK’. The Waveform will automatically be displayed in the **Utilities** menu.

Once all the required values have been entered in the Sinusoidal generator dialog box, click on ‘OK’. The waveform will automatically be displayed upon generation completion.

5.2 Square

The dialog box as shown in Figure 5.5 appears if *SQUARE* was selected on the ***GENERATOR*** menu. The *SQUARE* wave generator dialog box has the same parameters as the *SINUSOIDAL* Waveform dialog box. Please refer to Section 5.1 on page 43 for an explanation of each field.

FIGURE 5.5 Square Wave Generator



Generating Square Wave

Signal Frequency	80
Sampling Rate	8000
Number of Samples	1024
Angular Phase Delay	0
Peak Amplitude (from zero)	1
DC offset	0
Output File Format	Binary
Output Number Type	32 bit floating point
Random Noise Type	no noise

Frequency Unit: [✓ Hertz] Radians/second

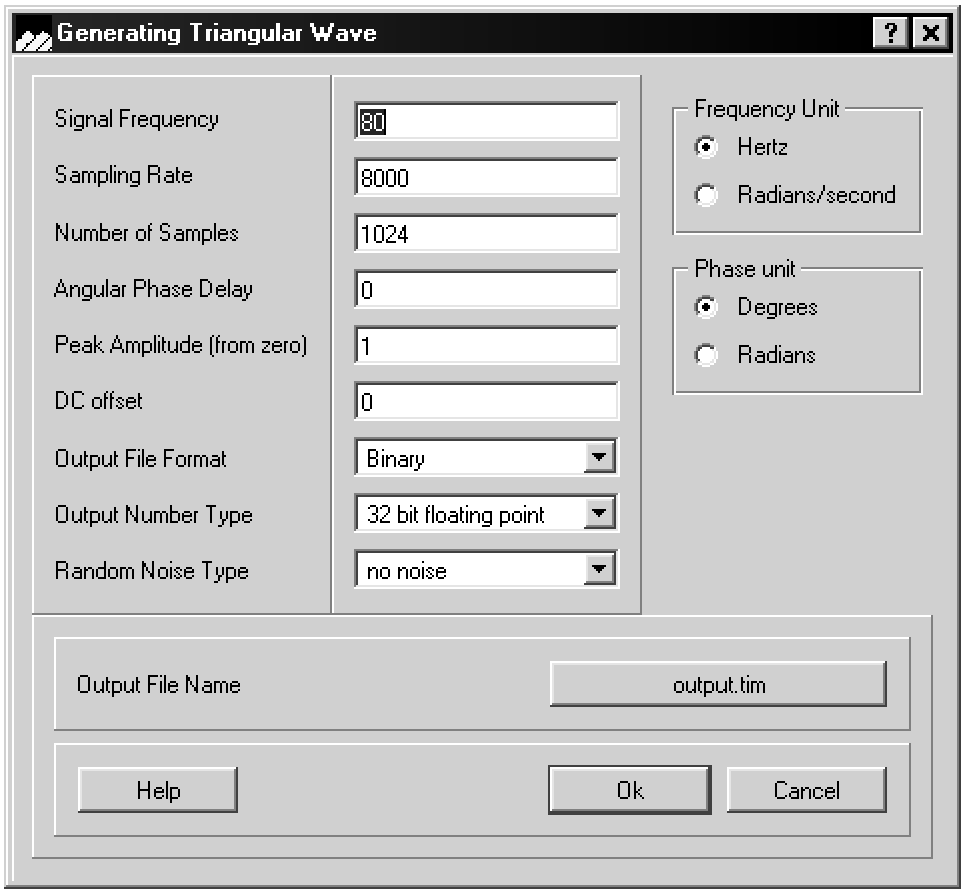
Phase unit: [✓ Degrees] Radians

Output File Name: output.tim

5.3 Triangular

The dialog box as shown in appears if *TRIANGULAR* was selected on the ***GENERATOR*** menu. The Triangular wave generator dialog box has the same parameters as the *SINUSOIDAL* Waveform dialog box. Please refer to Section 5.1 on page 43 for an explanation of each field.

FIGURE 5.6 **Triangular Wave Generator**



Generating Triangular Wave

Signal Frequency	80
Sampling Rate	8000
Number of Samples	1024
Angular Phase Delay	0
Peak Amplitude (from zero)	1
DC offset	0
Output File Format	Binary
Output Number Type	32 bit floating point
Random Noise Type	no noise
Output File Name	output.tim

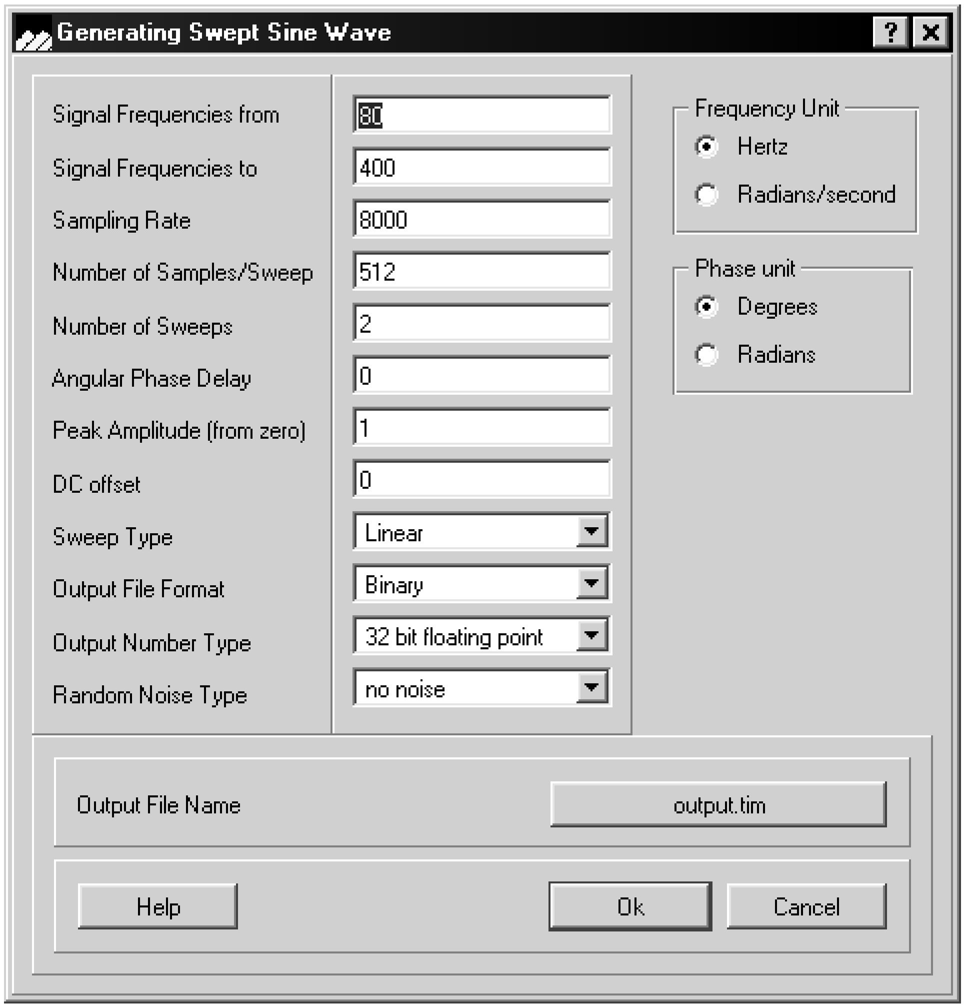
Frequency Unit: [✓ Hertz] Radians/second

Phase unit: [✓ Degrees] Radians

5.4 Swept Sine Function

If the *SWEPT SINE FUNCTION* was selected on the ***GENERATOR*** menu, Figure 5.7 is displayed:

FIGURE 5.7 **Swept Sine Wave Generator**



Generating Swept Sine Wave

Signal Frequencies from	80
Signal Frequencies to	400
Sampling Rate	8000
Number of Samples/Sweep	512
Number of Sweeps	2
Angular Phase Delay	0
Peak Amplitude (from zero)	1
DC offset	0
Sweep Type	Linear
Output File Format	Binary
Output Number Type	32 bit floating point
Random Noise Type	no noise
Output File Name	output.tim

Frequency Unit: [✓ Hertz] Radians/second

Phase unit: [✓ Degrees] Radians

The Swept Since Function records samples in an output file by sampling continuous time sine waves in a range of frequencies.

The fields, ***Signal Frequencies From*** and ***Signal Frequencies To***, specify the range of frequencies that will be swept by the generator.

The field, ***Number of Samples/Sweep***, specifies the number of samples that will be recorded in the output file for each sweep of the specified frequency range.

The field, ***Number of Sweeps***, specifies the number of times the specified frequency range will be swept while generating the waveform.

The total number of points in the output file is the product of the number of sweeps and the number of points per sweep.

The field, ***Sweep type***, controls the distribution of frequencies in the frequency range of the sweep. A linear distribution implies the frequency of the sine waves within a sweep changes linearly. Similarly, a Log distribution implies the frequency of the sine waves within a sweep changes logarithmically

Assume the sweep starting frequency is 1 and the sweep ending frequency is 100 with the number of samples per sweep being 200. Then for a linear sweep, the frequencies will be:

$$\boldsymbol{i}\left[\frac{100-1}{199}\right]+1 \quad \text{for i=0,...,199} \tag{EQ 5.1}$$

A more generalized form of this is formula can be expressed as follows:

S = Starting frequency

E = Ending frequency

N = Number of points/sweep

then the frequencies in the sweep are:

$$\left(\boldsymbol{f_i} = \boldsymbol{i}\left[\frac{\boldsymbol{E}-\boldsymbol{S}}{\boldsymbol{N}-1}\right]+\boldsymbol{S} \quad \text{for i=1,...,N-1}\right)\ldots \tag{EQ 5.2}$$

and the function is:

$$\sin(2\pi \boldsymbol{f_i}\boldsymbol{x}(\boldsymbol{n})) \tag{EQ 5.3}$$

where x(n) is the time at the nth sample.

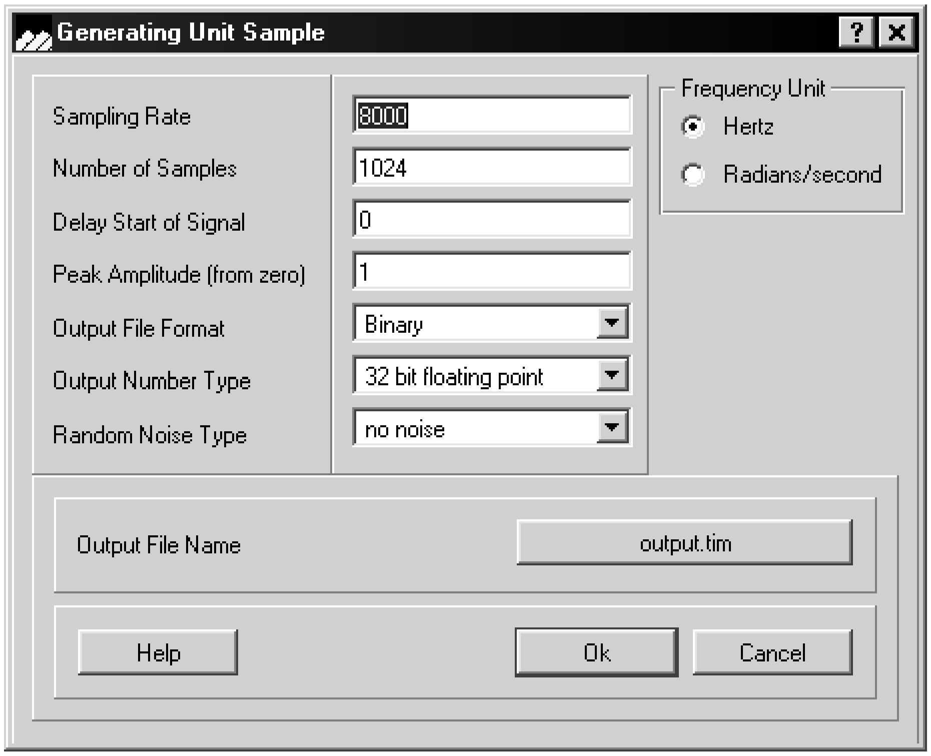
For logarithmic interpolation:

$$\log 10\boldsymbol{f_i} = \left[\frac{\boldsymbol{i}[\log 10(\boldsymbol{E})-\log 10(\boldsymbol{S})]}{\boldsymbol{N}-1}\right]+\log 10(\boldsymbol{S}) \quad \text{""""} \quad \text{for i=1,...,N-1} \tag{EQ 5.4}$$

5.5 Unit Sample

If *UNIT SAMPLE* was selected on the ***GENERATOR*** menu, Figure 5.8 is displayed. Please refer to Section 5.1 on page 43 for an explanation of number of sample points, output file name, output file format, output number type and random noise. The unit sample can have a delay specified in terms of number of samples. The peak amplitude can be set to any value greater than zero for floating point data but the maximum value for fixed point fractional must be less than 1.0. Thus, the maximum value for fixed point fractional is 7FFF in hexadecimal. Note that adding random noise allows the generation of a noisy unit pulse.

FIGURE 5.8 **Unit Sample Generator**



Generating Unit Sample

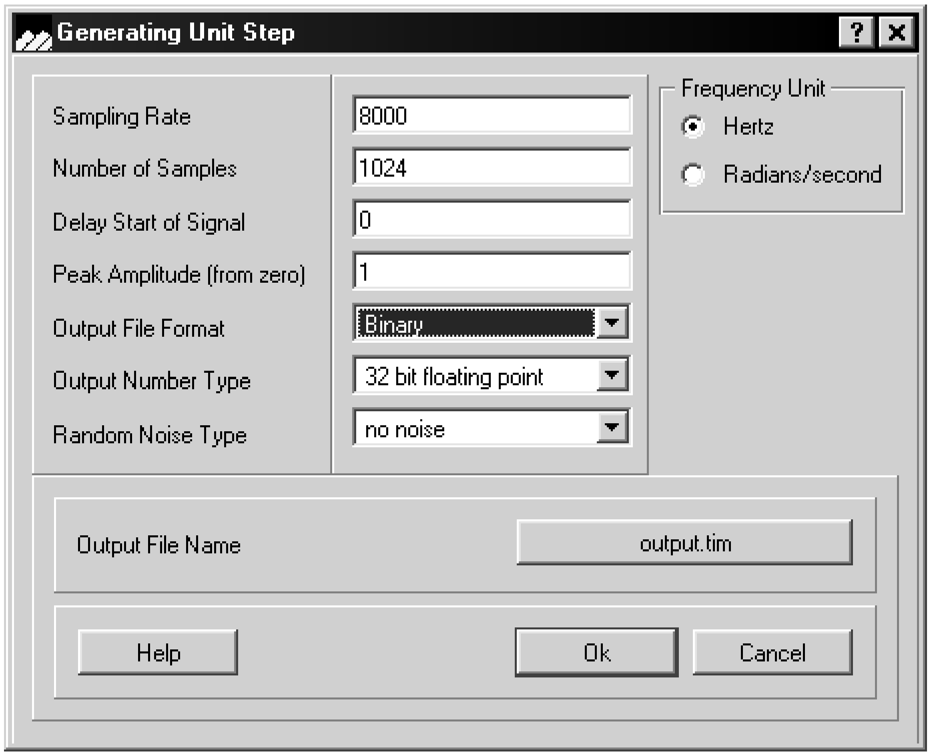
Sampling Rate	8000
Number of Samples	1024
Delay Start of Signal	0
Peak Amplitude (from zero)	1
Output File Format	Binary
Output Number Type	32 bit floating point
Random Noise Type	no noise
Output File Name	output.tim

Frequency Unit: [✓ Hertz] Radians/second

5.6 Unit Step

If *UNIT STEP* was selected on the ***GENERATOR*** menu, Figure 5.9 is displayed

FIGURE 5.9 Unit Step Generator


Generating Unit Step

Sampling Rate: 8000
Number of Samples: 1024
Delay Start of Signal: 0
Peak Amplitude (from zero): 1
Output File Format: Binary
Output Number Type: 32 bit floating point
Random Noise Type: no noise

Frequency Unit: [✓ Hertz] Radians/second

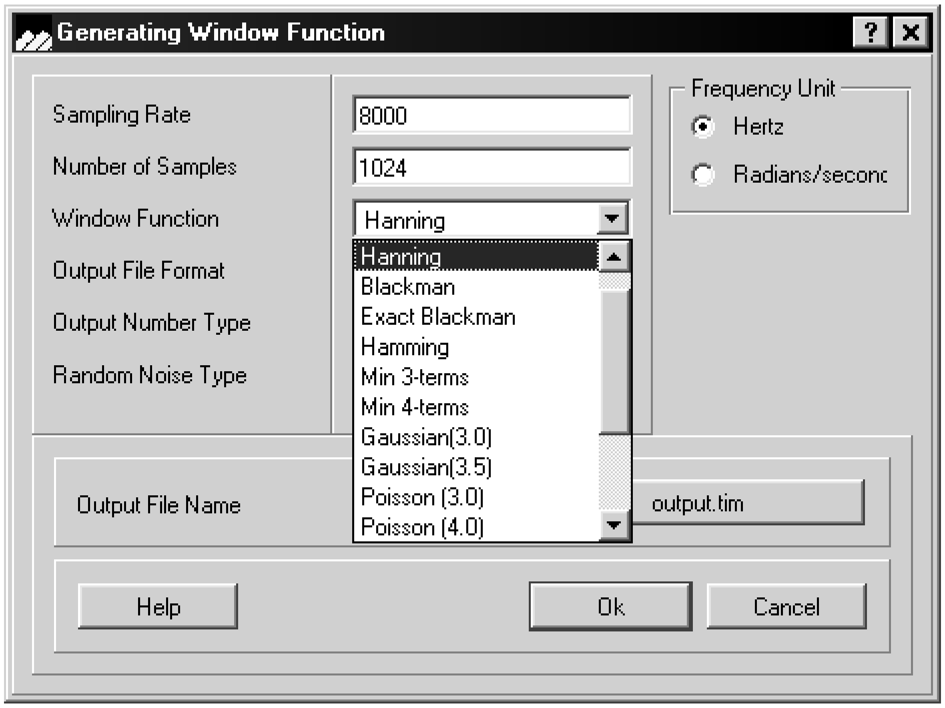
Output File Name: output.tim

The *UNIT STEP* waveform dialog box has the same parameters as the *UNIT SAMPLE* waveform dialog box. Please refer to Section 5.1 on page 43 for an explanation of number of sample points, output file name, output file format, output number type or random noise. See Section 5.5 on page 50 section for a description of delay, start of signal and peak amplitude.

5.7 Window Functions

Use the pull-down list under *WINDOW* function to select the desired window function.

FIGURE 5.10 Window Function Generator


Generating Window Function

Sampling Rate: 8000
Number of Samples: 1024
Window Function: Hanning
Output File Format
Output Number Type
Random Noise Type

Frequency Unit: (●) Hertz ( ) Radians/seconc

Hanning
Blackman
Exact Blackman
Hamming
Min 3-terms
Min 4-terms
Gaussian(3.0)
Gaussian(3.5)
Poisson (3.0)
Poisson (4.0)

Output File Name: output.tim

The available window functions are shown in Table 5-1 on page 53

TABLE 5-1 **Window Functions**

Window Function	Height of First Sidelobe db	Decay of Function db/Octave
Rectangular	-13.00	6
Hanning	-31.47	18
Blackman	-58.11	18
Exact Blackman	-68.20	6
Hamming	-43.19	6
Min 3-terms	-71.48	6
Min 4-terms	-98.17	6
Gaussian (3.0)	-55.00	6
Gaussian (3.5)	-69.00	6
Poisson (3.0)	-24.00	6
Poisson (4.0)	-31.00	6
Cauchy (4.0)	-35.00	6
Cauchy (5.0)	-30.00	6
Kaiser-Bessel (3.0)	-69.00	6
Kaiser-Bessel (3.5)	-82.00	6
Triangle	-27.00	6

Note: For more detailed explanations in Window functions, please refer to the following excellent texts recommended in the Reference:

- Reference No. 5, Harris F.J., On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform. Proc. IEEE, Vol 66, No. 1, pp. 51-83, Jan 1978

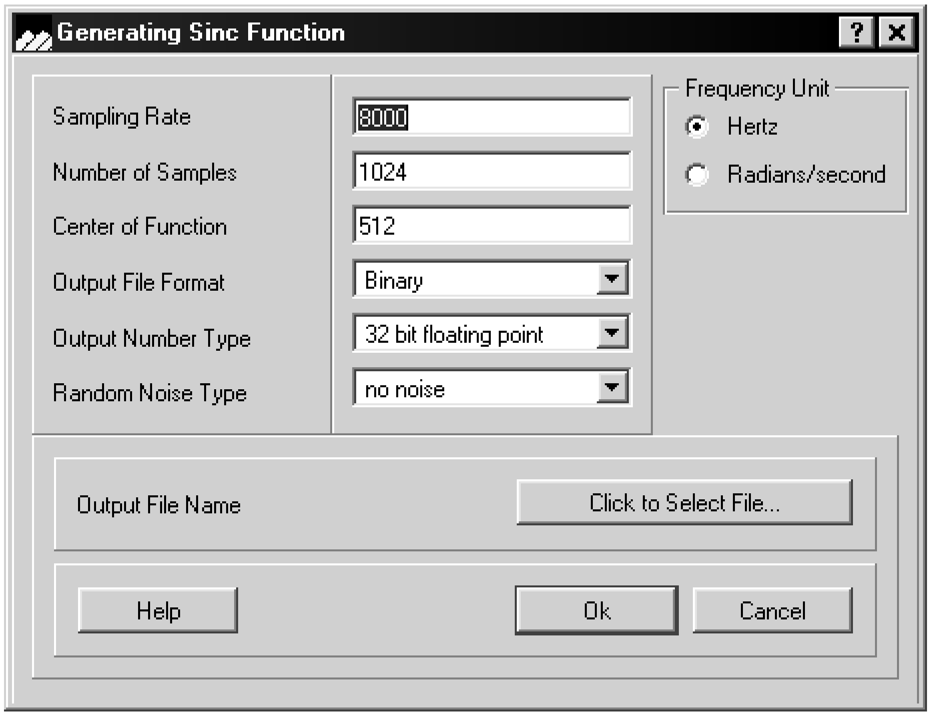
and

- Reference No. 10, Nuttal A. H. Some Windows with Very Good Sidelobe Behavior, IEEE Trans. Acoust., Speech and Signal Processing, vol ASSP-29, No. 1, pp 84-91, Feb. 1981

5.8 Sinc Function

The dialog box as shown in appears if the *SINC FUNCTION* was selected on the ***GENERATOR*** menu.

FIGURE 5.11 Sinc Function Generator



Sampling Rate	8000
Number of Samples	1024
Center of Function	512
Output File Format	Binary
Output Number Type	32 bit floating point
Random Noise Type	no noise
Output File Name	

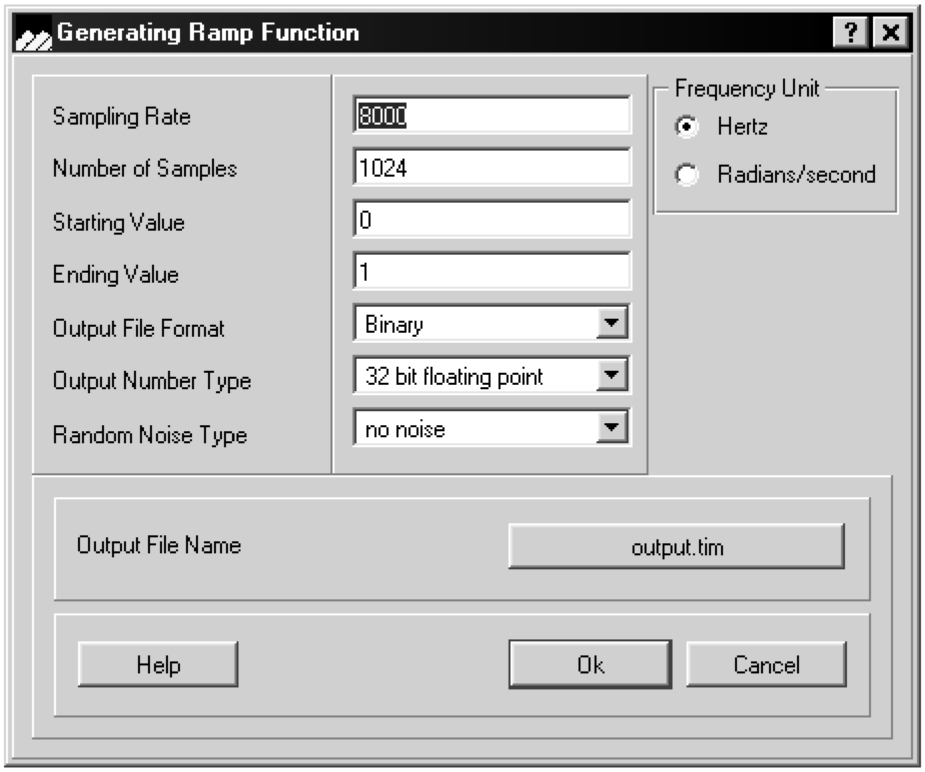
Frequency Unit
[✓ Hertz]
Radians/second

This allows the creation of a simple Sinc pulse with a specified center and duration.

5.9 Ramp Function

The dialog box as shown in Figure 5.12 appears if the *RAMP* function was selected on the ***GENERATOR*** menu.

FIGURE 5.12 **Ramp Function Generator**


Generating Ramp Function

Sampling Rate	8000
Number of Samples	1024
Starting Value	0
Ending Value	1
Output File Format	Binary
Output Number Type	32 bit floating point
Random Noise Type	no noise

Frequency Unit: [✓ Hertz] Radians/second

Output File Name: output.tim

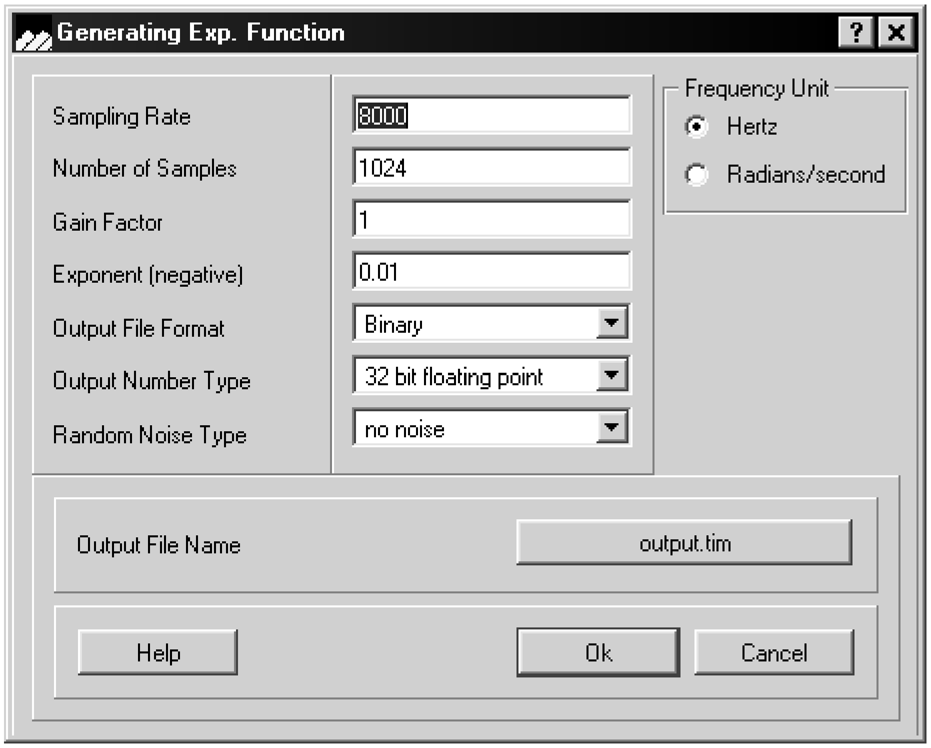
The starting value is used for the first sample and the end value is used for the last sample. Thus the slope of the ramp function is:

$$\frac{\text{end value - starting value}}{((\text{number of samples - 1}) \times \text{sampling rate})} \quad \text{(EQ 5.5)}$$

5.10 Exponential Function

The dialog box as shown in Figure 5.13 appears if the *EXPONENTIAL* Function was selected on the ***GENERATOR*** menu.

FIGURE 5.13 Exponential Function Generator



Sampling Rate	8000
Number of Samples	1024
Gain Factor	1
Exponent (negative)	0.01
Output File Format	Binary
Output Number Type	32 bit floating point
Random Noise Type	no noise
Frequency Unit	[✓ Hertz] Radians/second
Output File Name	output.tim

The exponential generator outputs a waveform using the following formula:

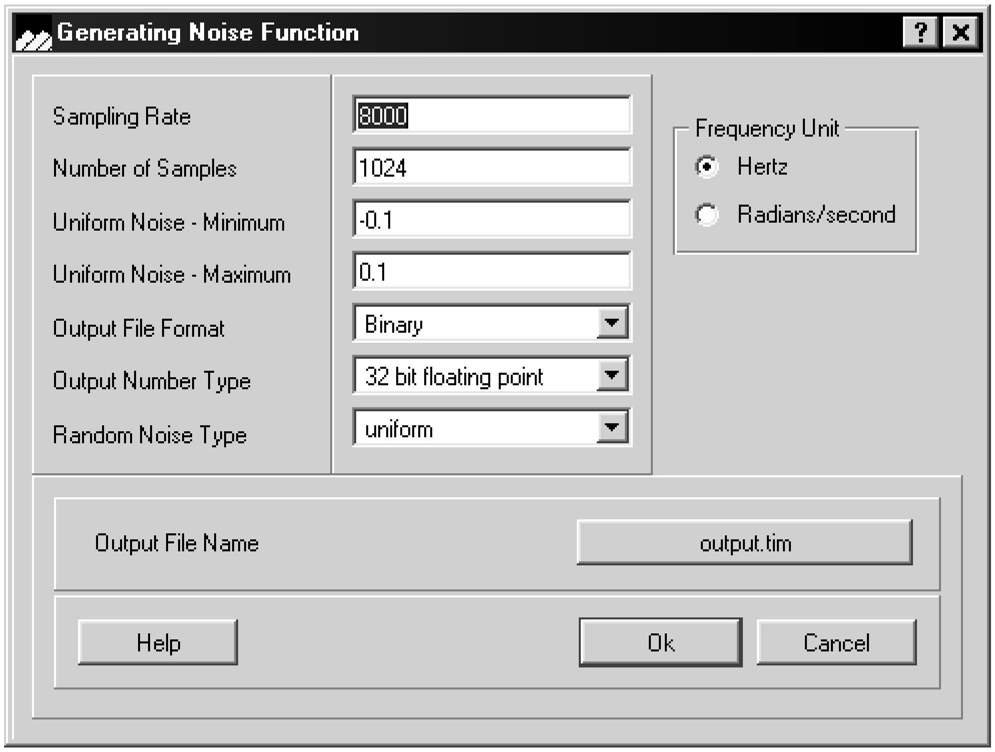
$$y(n) = Ae^{-an} \qquad \text{(EQ 5.6)}$$

where A is the gain factor and a is the exponent.

5.11 Noise Function

The dialog box as shown in Figure 5.14 appears if *NOISE FUNCTION* was selected on the ***GENERATOR*** menu.

FIGURE 5.14 Noise Function Generator


Generating Noise Function

Sampling Rate: 8000
Number of Samples: 1024
Uniform Noise - Minimum: -0.1
Uniform Noise - Maximum: 0.1
Output File Format: Binary
Output Number Type: 32 bit floating point
Random Noise Type: uniform

Frequency Unit: [✓ Hertz] Radians/second

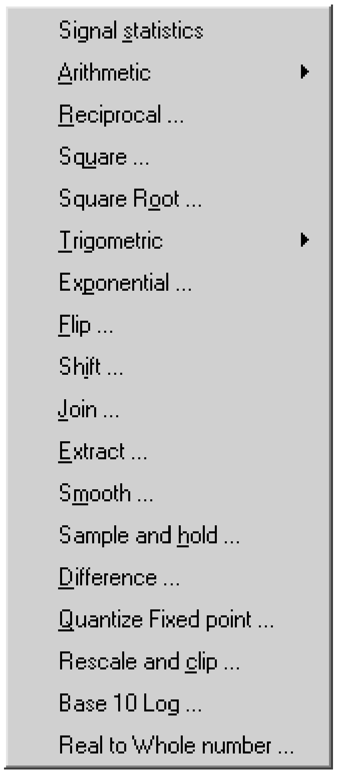
Output File Name: output.tim

Two noise functions are available. If Uniform noise is selected, then the minimum and maximum noise values are required to define the function. If Normal (or Gaussian) Noise is selected, then the mean and variance must be specified.

CHAPTER 6 *Operation Menu*

The ***OPERATION*** menu allows the selection of operations on time domain sequence files, either those generated by using the ***GENERATOR*** menu or data acquired by reading external data to a disk file. The ***OPERATION*** menu is shown in Figure 6.1:

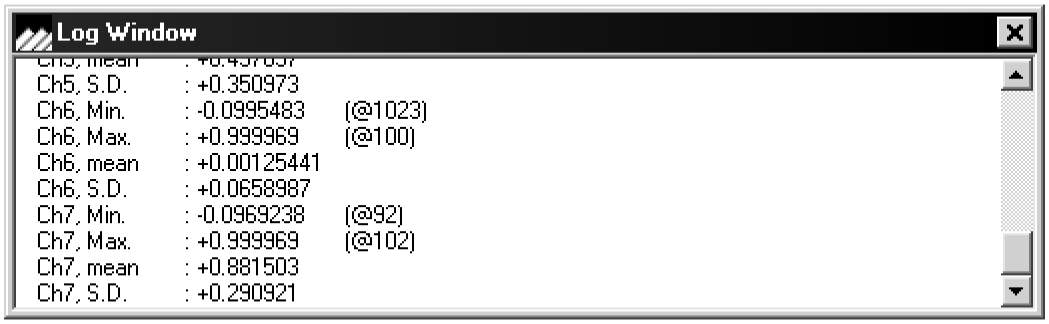
FIGURE 6.1 **Operation Menu**



6.1 Signal Statistics

Signal statistics are displayed in the Log Window. Selecting the signal statistics operation will extract the statistical detail pertaining to the signal currently displayed in the active graph window. An example of these statistics is shown in Figure 6.2:

FIGURE 6.2 Log Window

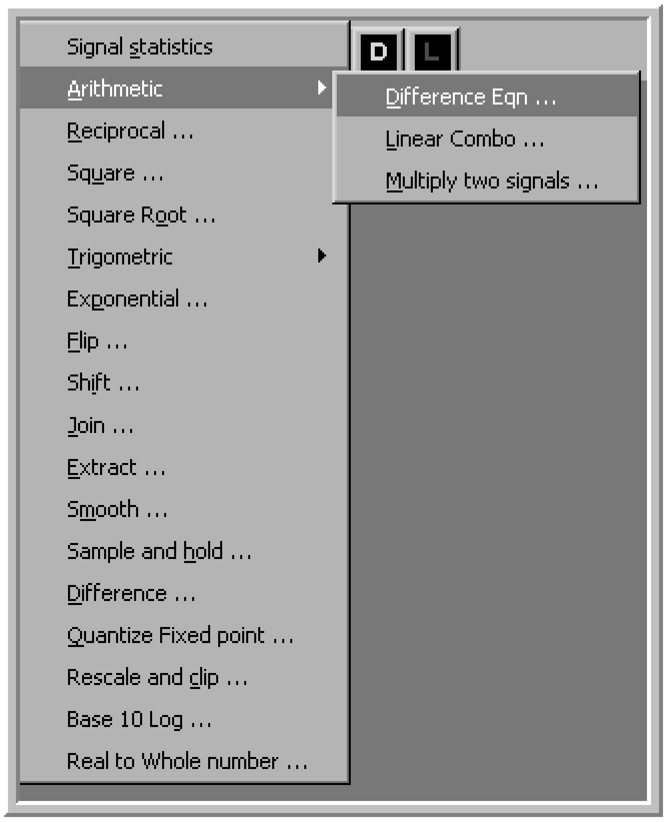


```
Ch5, S.D.      : +0.350973
Ch6, Min.      : -0.0995483    (@1023)
Ch6, Max.      : +0.999969     (@100)
Ch6, mean      : +0.00125441
Ch6, S.D.      : +0.0658987
Ch7, Min.      : -0.0969238    (@92)
Ch7, Max.      : +0.999969     (@102)
Ch7, mean      : +0.881503
Ch7, S.D.      : +0.290921
```

6.2 Arithmetic

This function supports the following arithmetic operations - Difference Equations; Linear Combination of Two signals with Offset and Multiplication of Two Signals, all of which are discussed in further detail below. The available options are shown in Figure 6.3.

FIGURE 6.3 Arithmetic Functions



6.2.1 Difference Equation

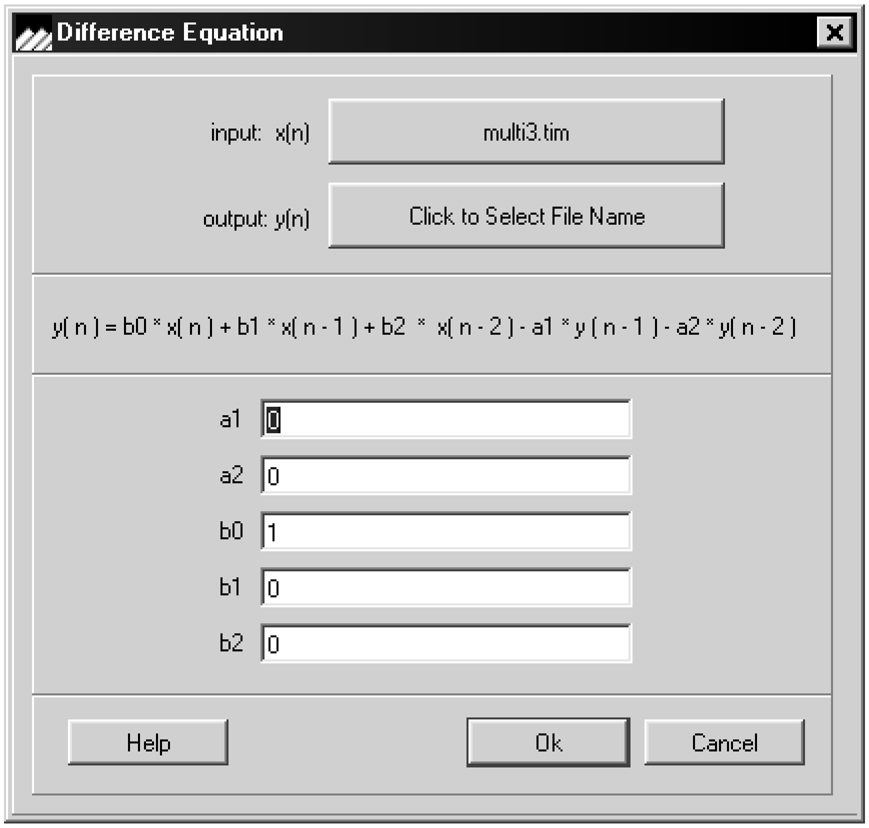
This operation is used to implement a second order difference equation.

$$y(n) = b_0x(n) + b_1x(n-1) + b_2x(n-2) - a_1y(n-1) - a_2y(n-2) \quad . \qquad \textbf{(EQ 6.1)}$$

The output file sampling rate is set to the input file sampling rate. The number of output data values is equal to the number of data values in the input file. The difference equation will process the input file until and EOF (End of File) is reached. If the User wants the difference equations to process a number of zero samples at the end of the signal, these zeros must be included in the file.

The dialog box shown in Figure 6.4 is used to obtain the input filename, output filename and the coefficient values for the second order difference equation.

FIGURE 6.4 **Difference Equation Input**



Difference Equation

input: x(n) multi3.tim

output: y(n) Click to Select File Name

y(n) = b0 * x(n) + b1 * x(n - 1) + b2 * x(n - 2) - a1 * y (n - 1) - a2 * y(n - 2)

a1	0
a2	0
b0	1
b1	0
b2	0

An input filenames (x(n)) and an output filename y(n)) must be entered. Click on the x(n) field or y(n) field to pop up a list of available files. Click on the desired file to select it, the file name will appear on the button. For a new output file, type in the filename in the Select field. If the filename is not suffixed by TIM, 'TIM' will be concatenated with the entered filename. Click on Save and the filename will appear on the output filename button. Selecting 'OK' will start the operation.

The Difference Equation operation can be used to estimate IIR or FIR filter responses. In such a case, the a_i and b_i values act as filter coefficients, or poles and zeros of the digital filter described by the difference equation.

If the poles which are defined by the a_1 and a_2 values are outside the unit circle, the system will be unstable. dsPICworks software will warn of an unstable difference equation, but will allow the execution of an unstable system. For a floating point data file, this will eventually cause the output to be set to a maximum value. For fixed point files an unstable system will cause wrap around to opposite sign values or saturation depending on the wrap/saturate setting.

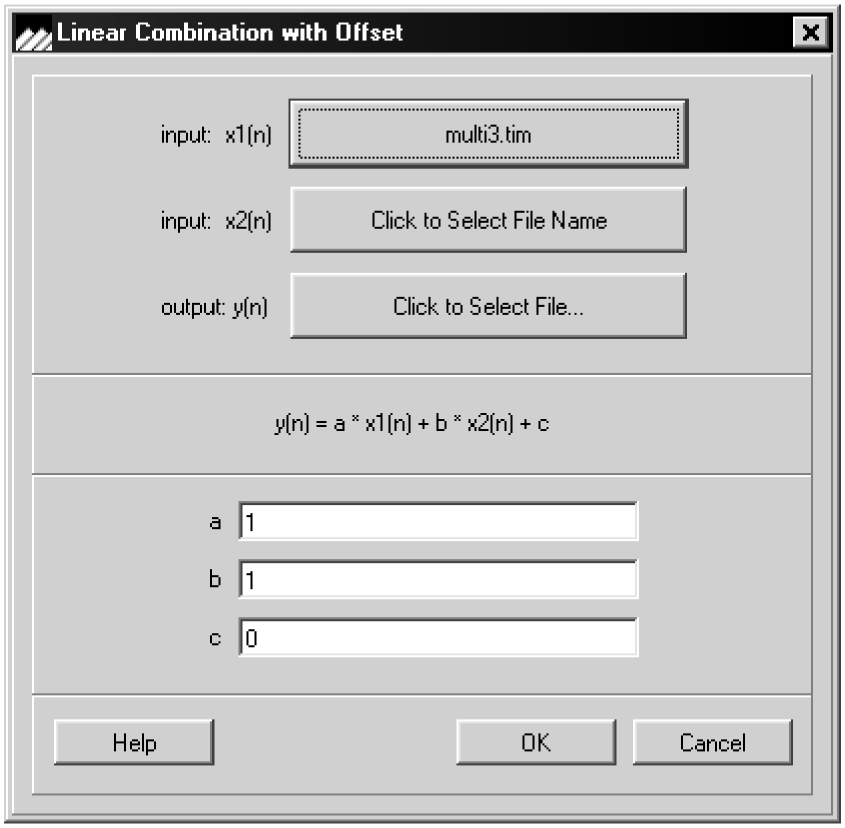
6.2.2 Linear Combination

This operation is used to add two signals on a sample by sample basis as follows:

$$y(n) = ax_1(n) + bx_2(n) + c \qquad \textbf{(EQ 6.2)}$$

For this operation to be valid, the time domain files (File 1 and File 2) representing the signals $x_1(n)$ and $x_2(n)$ respectively, should have the same sampling frequency, however, unequal sampling frequencies are allowed. The number of output data values is equal to the maximum of the number of data values in File 1 or File 2. If the number of data values in File 1 and File 2 are not equal, the shorter file is logically extended with zeros. If the number types of the input files are different, the output number type will be floating point. The dialog box shown in Figure 6.5 appears if *LINEAR COMBINATION* was selected on the ***OPERATION*** menu. Two input filenames (File 1 and File 2) and an output filename must be entered.

FIGURE 6.5 **Linear Combination of Two Signals with an Offset**


Linear Combination with Offset
input: x1(n)
multi3.tim
input: x2(n)
Click to Select File Name
output: y(n)
Click to Select File...
y(n) = a * x1(n) + b * x2(n) + c
a
1
b
1
c
0
Help
OK
Cancel

If the value of parameter a and/or b is zero, then the operation is defined as follows

$$y(n) = ax_1(n) + c \quad \text{if b=0} \qquad \textbf{(EQ 6.3)}$$

$$y(n) = bx_2(n) + c \quad \text{if a=0} \qquad \textbf{(EQ 6.4)}$$

$$y(n) = c \quad \text{if a=b=0} \qquad \textbf{(EQ 6.5)}$$

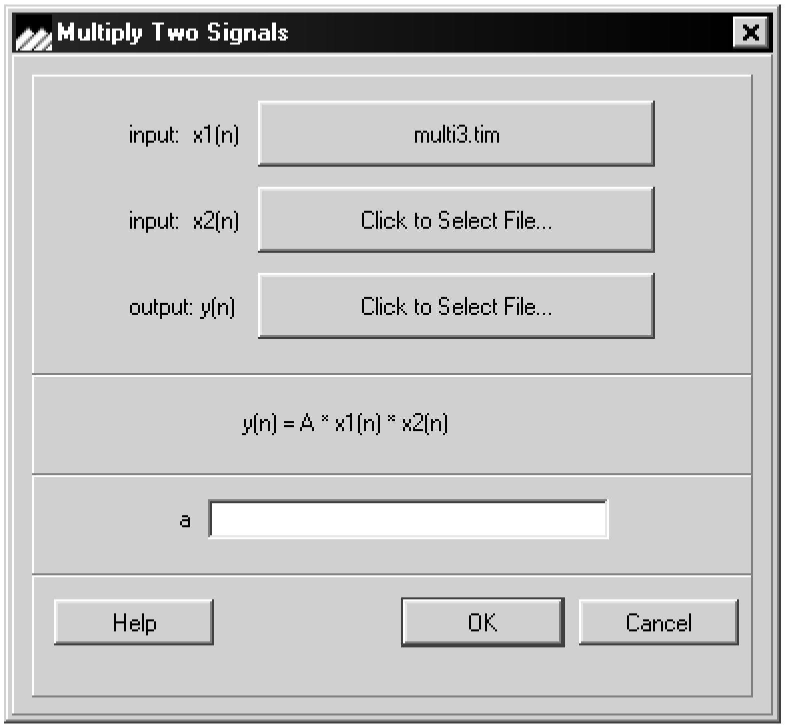
6.2.3 Multiplication

This operation is used to multiply two signals on a sample by sample basis, or to multiply a constant value ‘c’ by a time domain signal as follows:

$$y(n) = ax_1(n) \times x_2(n) \qquad \textbf{(EQ 6.6)}$$

For this operation to be valid, the time domain files, File 1 and File 2 representing the signals x1(n) and x2(n) respectively, should have the same sampling frequency. The output file sampling rate is set to the input file sampling rate. If the two files have a different number of sample values, the shorter is logically extended with 1.0 for floating point data format and 7FFF (hexadecimal) with fixed point fractional format. The number types for File 1 and File 2 must be the same. If the number types of the input files are different the output number type will be floating point. The dialog box shown in Figure 6.6 is used to obtain the input filenames and the output filename for multiplying two signals sample by sample.

FIGURE 6.6 **Multiply Two Signals**



6.3 Reciprocal

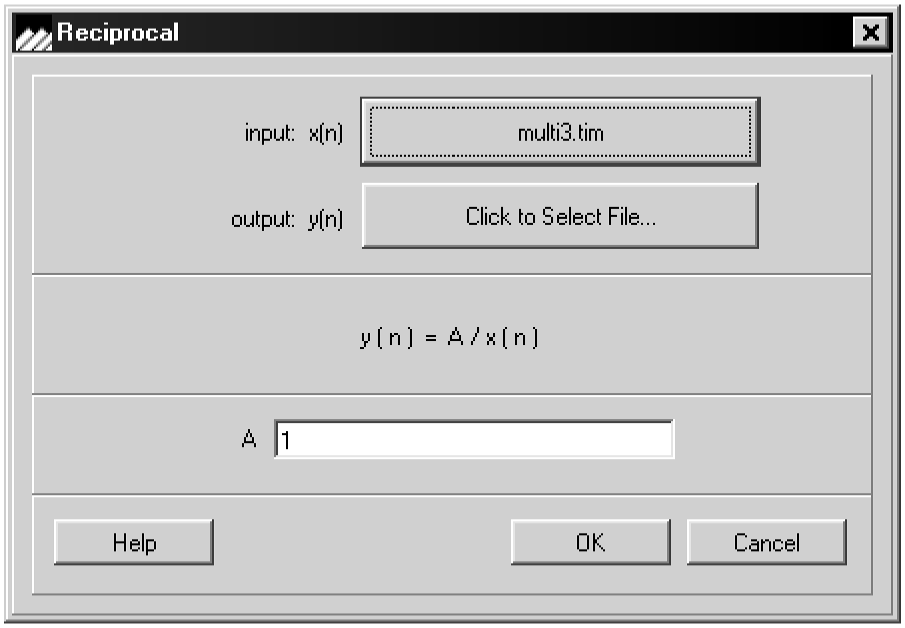
This operation is used to determine the multiplicative inverse of a signal on a sample by sample basis:

$$y(n) = \frac{a}{x(n)} \qquad \text{(EQ 6.7)}$$

The reciprocal operation is valid only for floating point data. The reciprocal values of fractional fixed point are greater than 1 and cannot be represented in fractional fixed point. The reciprocal of zero in the IEEE standard is 'not a number' (NAN).

The dialog box shown in Figure 6.7 is used to obtain the input filename for x(n) and the output filename for the reciprocal of x(n).

FIGURE 6.7 Form Reciprocal of a Signal



input: x(n) multi3.tim

output: y(n)

y(n) = A / x(n)

A 1

6.4 Square

This operation is used to perform the squaring of a signal on a sample by sample basis:

$$y(n) = A[x(n)]^2 \qquad \textbf{(EQ 6.8)}$$

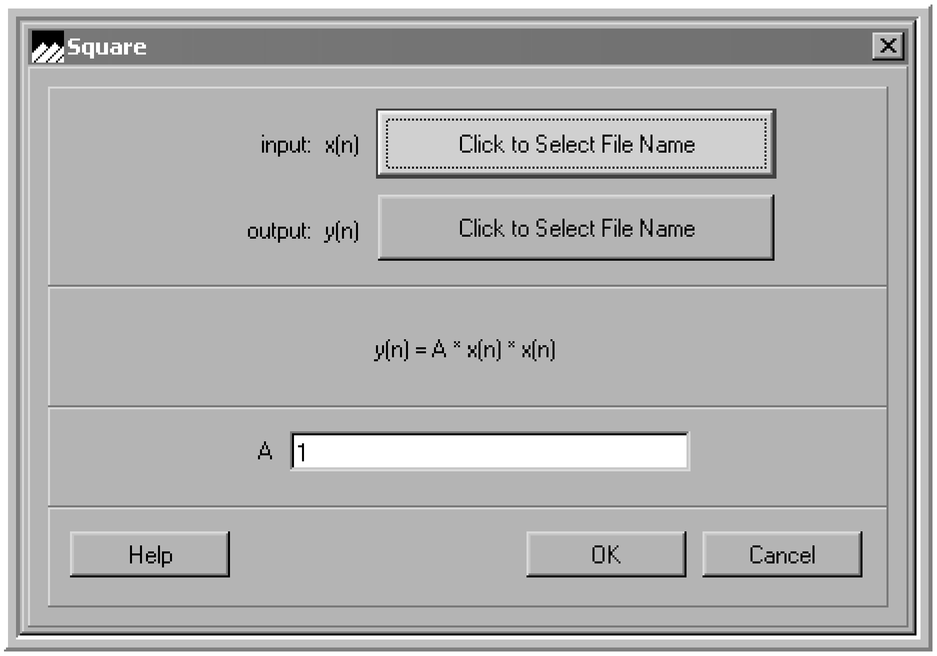
A special case occurs for the fractional fixed-point square defined by:

$$-1 \times -1 \qquad \textbf{(EQ 6.9)}$$

Ordinarily, this would result in +1.0 but +1.0 cannot be represented by a fractional fixed point data type. Hence the product results in the largest positive fractional fixed point number (1- 2^{15}) or 7FFF in hexadecimal format.

The dialog box displayed in Figure 6.8 is used to obtain the input filename for x(n) and the output filename for the square of x(n).

FIGURE 6.8 **Square a Signal**



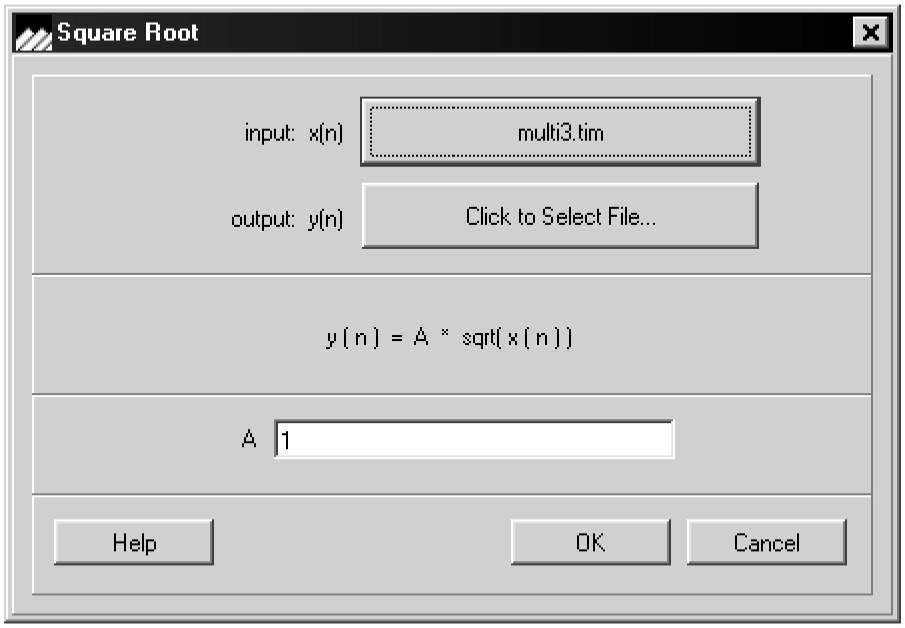
6.5 Square Root

This operation is used to take the square root of a signal on a sample by sample basis:

$$y(n) = A\,\mathrm{sgn}\,x\sqrt{|x|} \qquad \textbf{(EQ 6.10)}$$

The dialog box as displayed in Figure 6.9 is used to obtain the input filename for x(n) and the output filename for the square root of absolute x(n).

FIGURE 6.9 Square Root of a Signal



input: x(n) multi3.tim

output: y(n) Click to Select File...

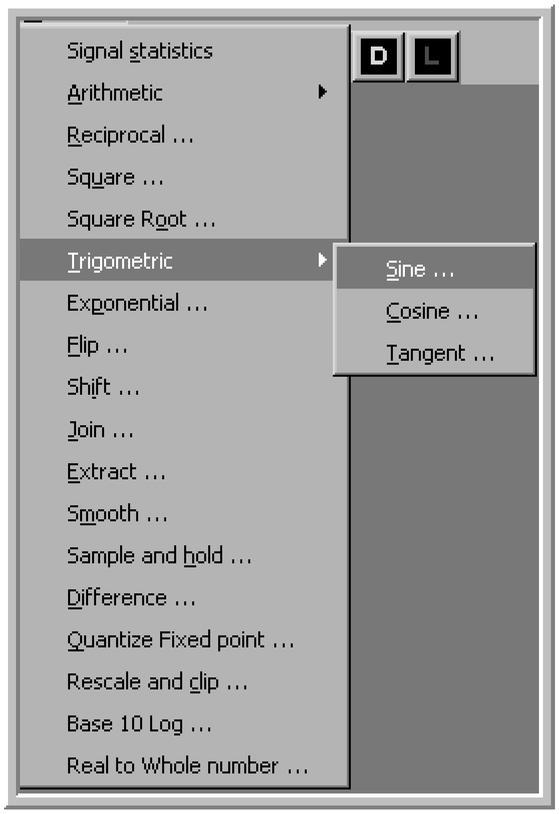
y (n) = A * sqrt(x (n))

A 1

6.6 Trigonometric Functions

Select either the *SINE, COSINE* or *TANGENT* trigonometric functions from the menu as displayed in Figure 6.10.

FIGURE 6.10 Trigonometric functions



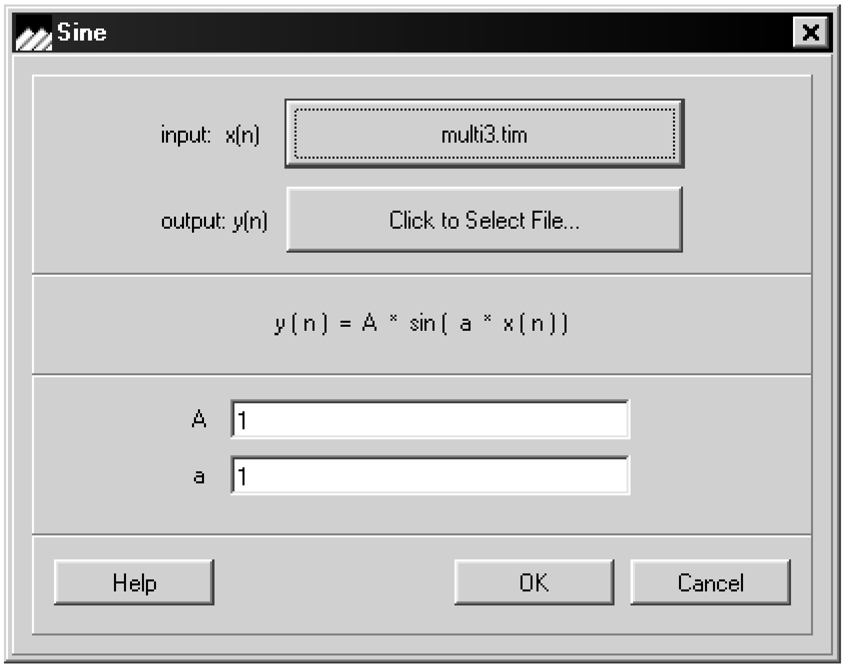
6.6.1 Sine

This operation is used to compute the sine of a signal on a sample by sample basis:

$$y(n) = A\sin[ax(n)] \qquad \textbf{(EQ 6.11)}$$

The dialog box as displayed in Figure 6.11 is used to obtain the input filename for x(n) and the output filename for y(n) where A is the amplitude of the signal and ax(n) is the argument of the sine function in radians, with a default of 1 for the amplitude and 1 for parameter a.

FIGURE 6.11 **Sine of a Signal**



Sine

input: x[n] multi3.tim

output: y[n] Click to Select File...

y[n] = A * sin[a * x[n]]

A 1

a 1

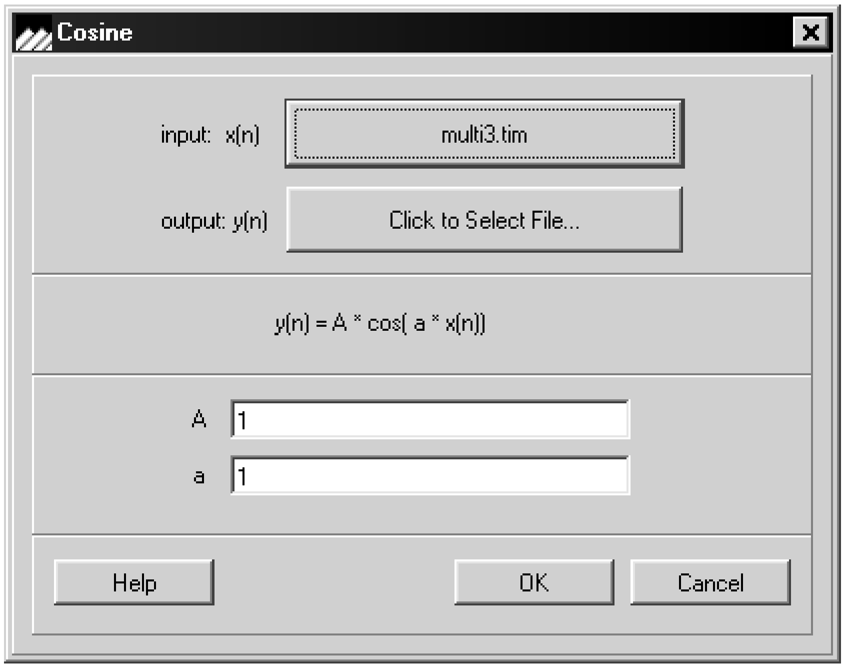
6.6.2 Cosine of a Signal

This operation is used to compute the cosine of a signal on a sample by sample basis:

$$y(n) = A\cos[ax(n)] \tag{EQ 6.12}$$

The dialog box as displayed in Figure 6.12 is used to obtain the input filename for x(n) and the output filename for y(n) where A is the amplitude of the signal and ax(n) is the argument of the cosine function in radians, with a default of 1 for the amplitude and 1 for parameter a.

FIGURE 6.12 Cosine of a Signal


Cosine

input: x(n) multi3.tim

output: y(n) Click to Select File...

y(n) = A * cos(a * x(n))

A 1

a 1

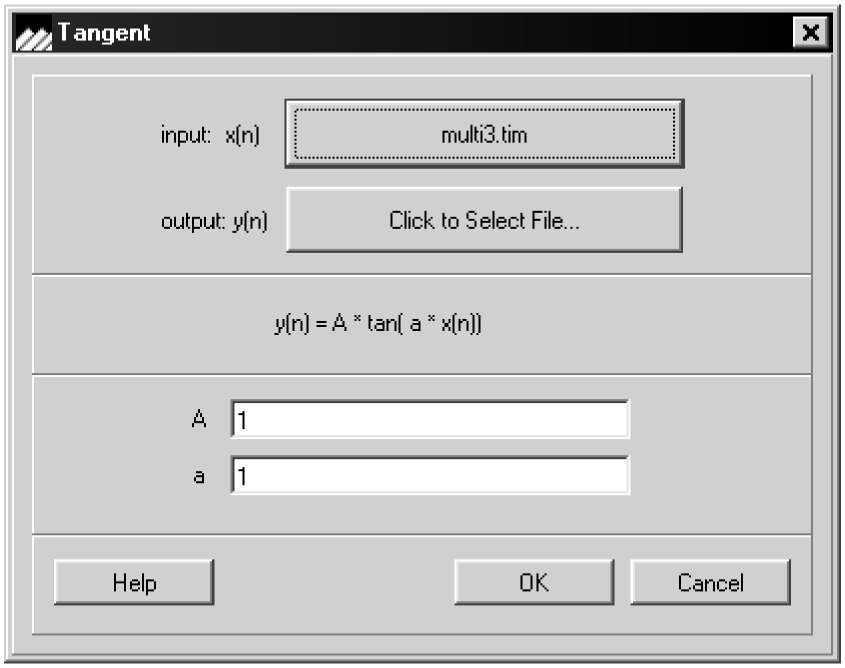
6.6.3 Tangent of a Signal

This operation is used to compute the tangent of a signal on a sample by sample basis:

$$y(n) = A\tan[ax(n)] \quad \textbf{(EQ 6.13)}$$

The dialog box as displayed in Figure 6.13 is used to obtain the input filename for x(n) and the output filename for (n) where A is the amplitude of the signal and ax(n) is the argument of the tangent function in radians, with a default of 1 for the amplitude and 1 for the parameter a.

FIGURE 6.13 Tangent of a Signal



input: x(n) multi3.tim

output: y(n) Click to Select File...

y(n) = A * tan(a * x(n))

A 1

a 1

When the tangent operator is applied to fixed-point fractional data type, results larger than +1.0 are either saturated or wrapped around as dictated by the settings in ***UTILITIES/*** *CONTROL CENTER* (Chapter 9.1, Control Center).

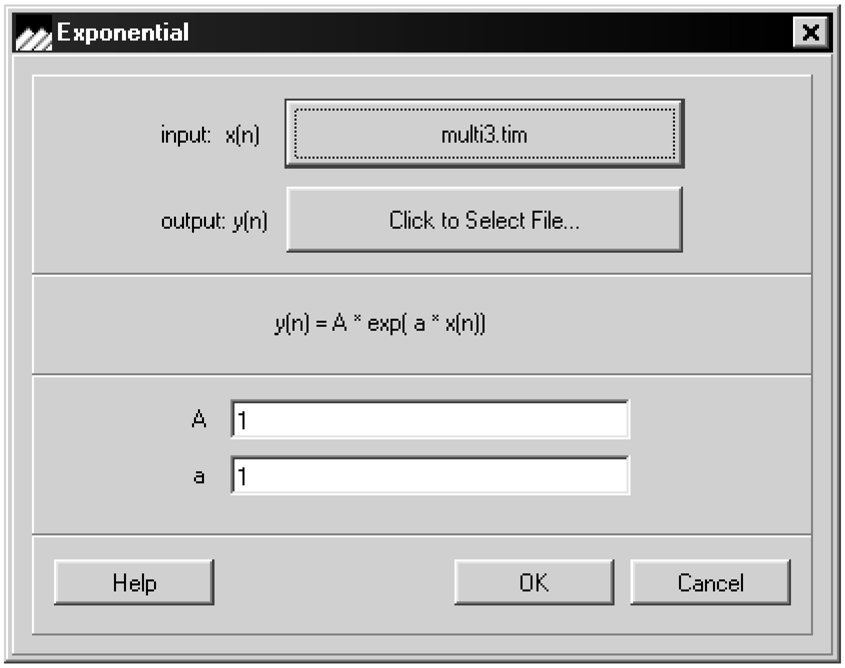
6.7 Exponential

This operation is used to compute the exponential of a signal on a sample by sample basis:

$$y(n) = Ae^{ax(n)} \qquad \text{(EQ 6.14)}$$

The dialog box as displayed in Figure 6.14 appears if Exponential was selected on the Operation menu. Enter the input filename for x(t) and an output filename for y(n) where A is a multiplicative constant and Bx(n) is the argument of the exponential function. To create $Ab^{x(n)}$ set a=ln b.

FIGURE 6.14 Exponential of a Signal



input: x(n) multi3.tim

output: y(n) Click to Select File...

y(n) = A * exp(a * x(n))

A 1

a 1

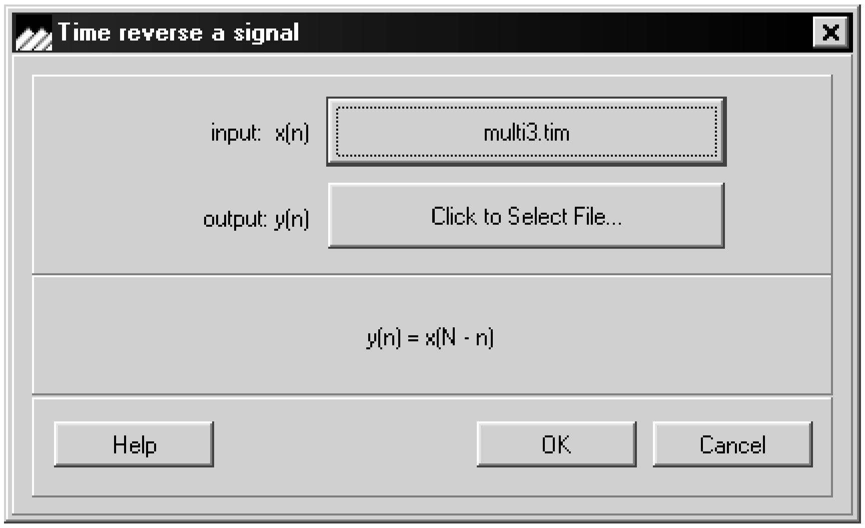
6.8 Flip

This operation reverses the order of samples in a given sequence as follows:

$$y(n) = x(N - n) \text{ for } n=0, 1,..., N \qquad \textbf{(EQ 6.15)}$$

The following dialog box as shown in Figure 6.15 appears if *FLIP* was selected on the ***OPERATION*** menu. Enter the input filename for x(n) and an output filename for the reverse image of x(n).

FIGURE 6.15 **Flip a Signal**



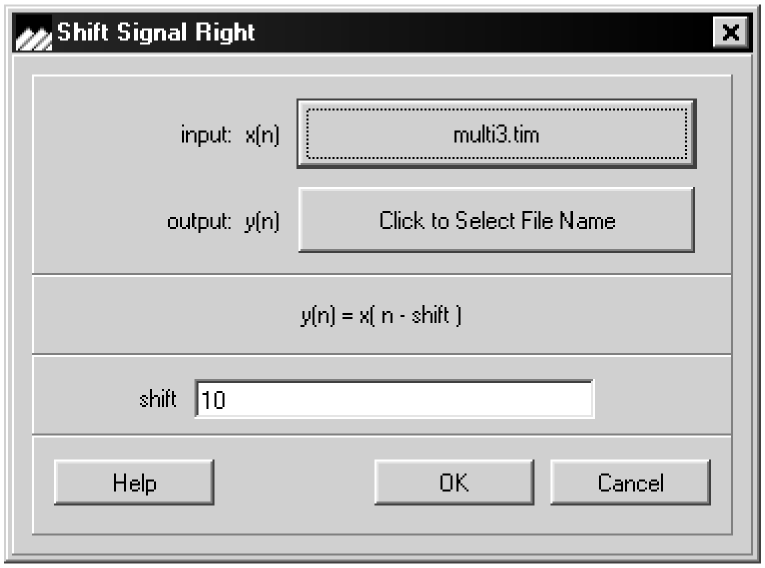
6.9 Shift

This operation is used to shift a signal a number of time steps either to the right or left.

$$y(n) = x(n-d) \quad \text{where } d \text{ is the shift count} \tag{EQ 6.16}$$

Note a positive shift moves the function to the right. The dialog box as displayed in Figure 6.16 appears if *SHIFT* was selected on the ***OPERATION*** menu. Enter the input filename for x(n) and an output filename for shifted output y(n).

FIGURE 6.16 **Shift a Signal**


Shift Signal Right

input: x[n] multi3.tim

output: y[n] Click to Select File Name

y[n] = x[n - shift]

shift 10

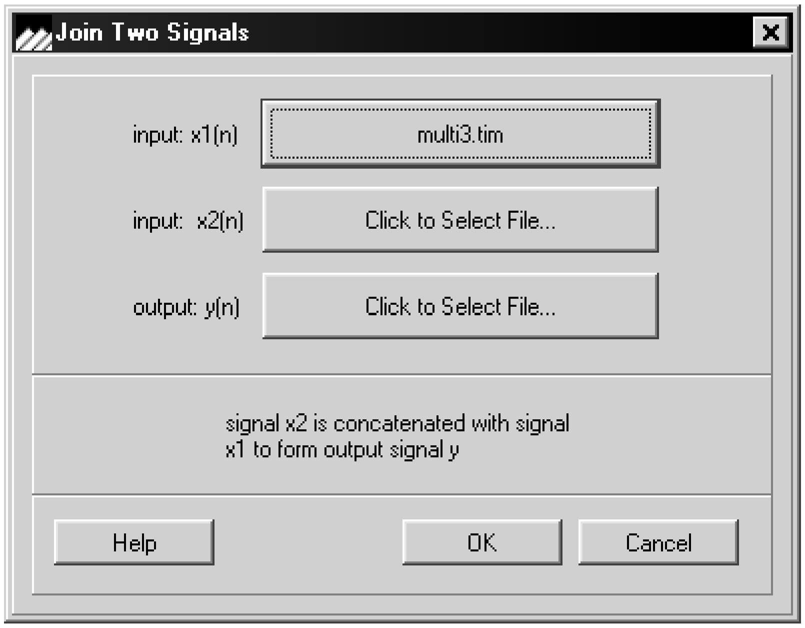
6.10 Join

This operation concatenates or joins two signals. Let x1(n) and x2(n) be sequences of length N and M respectively, then the concatenated sequence of x1(n) and x2(n) denoted y(n) would be:

$$y(n) = \begin{cases} x_1(n), & n = 0, \ldots, N-1 \\ x_2(n-N), & n = N, \ldots, N+M-1 \end{cases} \qquad \textbf{(EQ 6.17)}$$

For this operation to be valid, the time domain files representing signals x1(n) and x2(n) respectively should have the same frequency, however, unequal sampling frequencies are allowed. The sampling frequency of the output file {y(n)} is set to the input file sampling frequency. The dialog box as displayed in Figure 6.17 is used to obtain the input filenames for x1(n) and x2(n) and the output filename for y(n).

FIGURE 6.17 **Join (Concatenate) two Signals**


Join Two Signals

input: x1(n) multi3.tim

input: x2(n) Click to Select File...

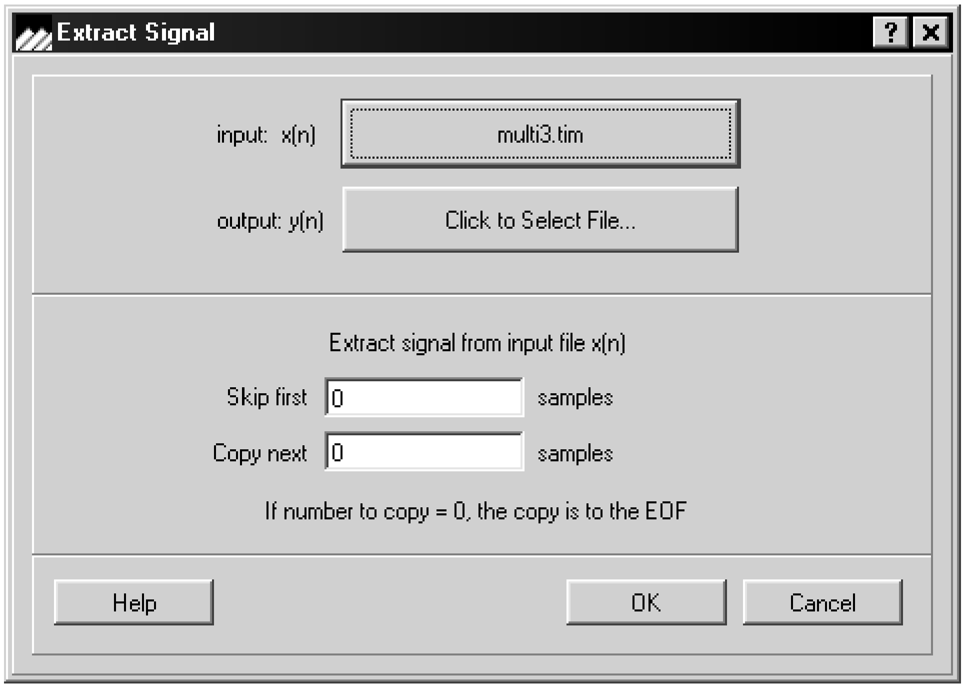
output: y(n) Click to Select File...

signal x2 is concatenated with signal
x1 to form output signal y

6.11 Extract Operation

This feature allows a segment of data to be extracted from one file to another file as shown in Figure 6.18.

FIGURE 6.18 Extract a Segment of a Signal



input: x(n) multi3.tim

output: y(n) Click to Select File...

Extract signal from input file x(n)

Skip first 0 samples

Copy next 0 samples

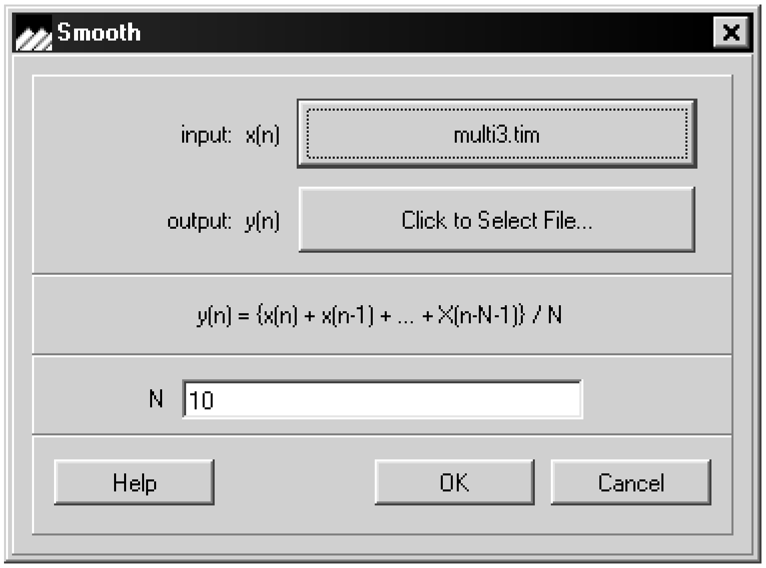
If number to copy = 0, the copy is to the EOF

6.12 Smooth

This function averages the current x(n) and the previous x(n-i) values for i=1,...,N-1. as shown in Figure 6.19.

$$y(n) = \frac{1}{d}\sum_{N}^{N-1} x(n-i) \qquad \text{(EQ 6.18)}$$

FIGURE 6.19 Smooth a Signal


Smooth

input: x(n) multi3.tim

output: y(n) Click to Select File...

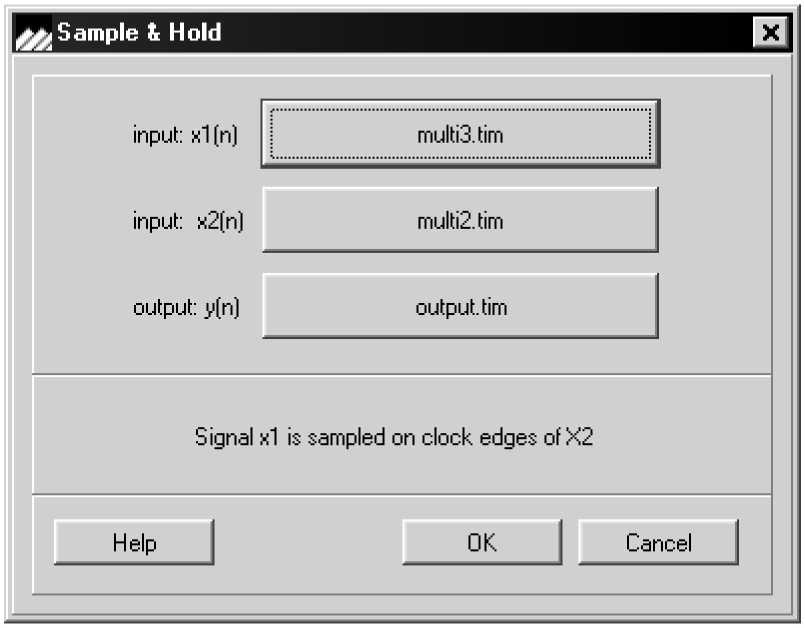
y(n) = {x(n) + x(n-1) + ... + X(n-N-1)} / N

N 10

6.13 Sample and Hold

This operation is used to digitally sample and hold a signal. The output signal is the holding values. The initial holding value is 0. The output signal value is set equal to the input signal value at time nT whenever the clock signal makes a positive transition. If the clock signal does not make a positive transition, the output signal value is set to the output signal value at time (n-1)T. A positive transition is said to occur at time nT if the clock at nT has a negative value and the clock at (n+1)T has a positive value.The clock signal can be any time domain waveform file. The sampling rate of the input file and the clock file should be the same although the system does not enforce this. The dialog box as displayed in Figure 6.20 to select the filenames for the signal, the clock and the output signal.

FIGURE 6.20 **Sample and Hold**



input: x1(n) multi3.tim

input: x2(n) multi2.tim

output: y(n) output.tim

Signal x1 is sampled on clock edges of X2

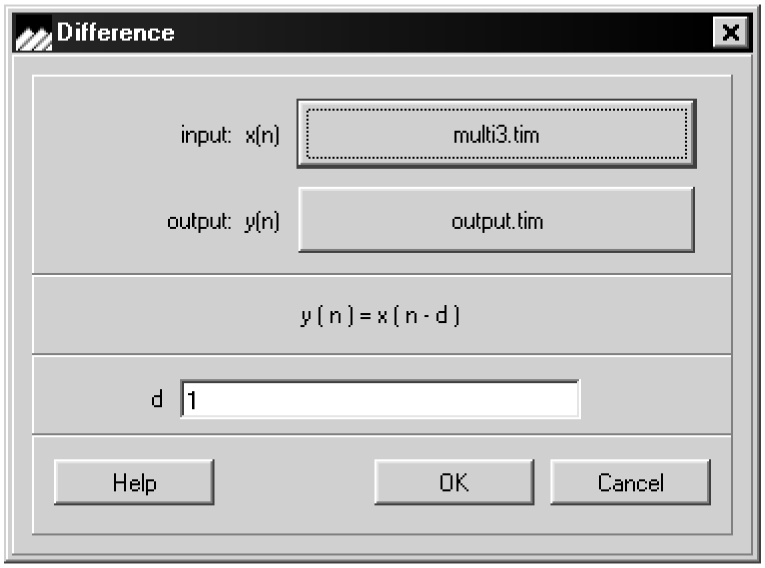
6.14 Difference

This operation is used to compute the difference of a signal on a sample by sample basis and is the equivalent to continuous time differentiation for the case of d=1.

$$y(n) = x(n) - x(n-d) \qquad \text{(EQ 6.19)}$$

The dialog box as displayed in Figure 6.21 appears if *DIFFERENCE* was selected on the ***OPERATION*** menu. Enter the input filename for x(t) and an output filename for x(n) - x(n-d)

FIGURE 6.21 **Difference**



Difference

input: x(n) multi3.tim

output: y(n) output.tim

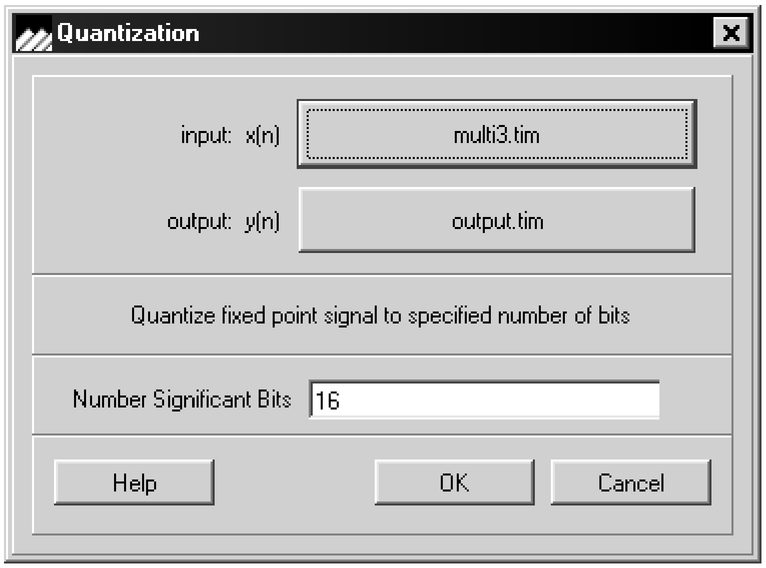
y(n)=x(n-d)

d 1

6.15 Quantize Fixed Point

This operation is as shown in Figure 6.22 used to simulate the impact of fixed point quantization on a sample by sample basis where x(t) is the signal to be quantized and y(t) represents the output with the number of significant digits selected by the user. Note that quantization may vary from 0-16. The output y(t) is truncated to the selected number of bits.

FIGURE 6.22 Quantize a Signal


Quantization

input: x(n) multi3.tim

output: y(n) output.tim

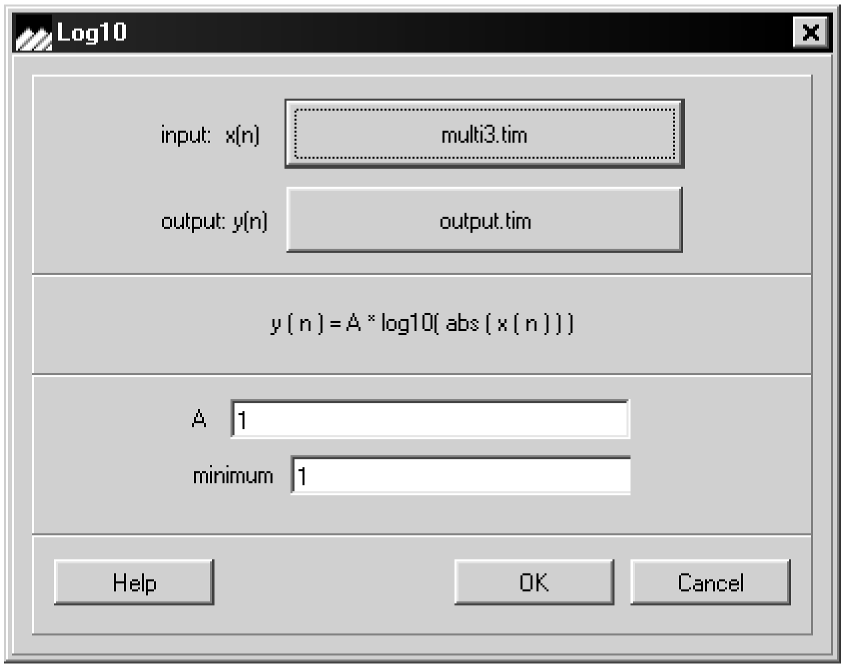
Quantize fixed point signal to specified number of bits

Number Significant Bits 16

6.16 Base10 Log

This operation as shown in Figure 6.23 takes the Log10 of the input signal on a sample by sample basis and sets the output file samples to that value

FIGURE 6.23 Base Log 10



input: x(n) multi3.tim

output: y(n) output.tim

y (n) = A * log10(abs (x (n)))

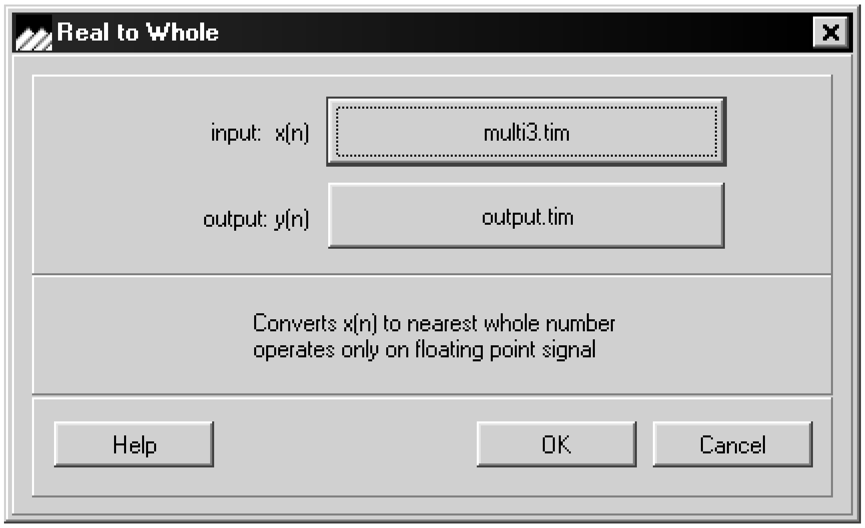
A 1

minimum 1

6.17 Real to Whole Number

This dialog box as shown in Figure 6.24 converts real numbers to whole numbers on a sample by sample basis and sets the output file samples accordingly.

FIGURE 6.24 Real to Whole Number Conversion



Converts x(n) to nearest whole number
operates only on floating point signal

6.18 Rescale and Clip

This operation as shown in Figure 6.25 allows rescaling a signal and saturating or clipping the signal as follows.

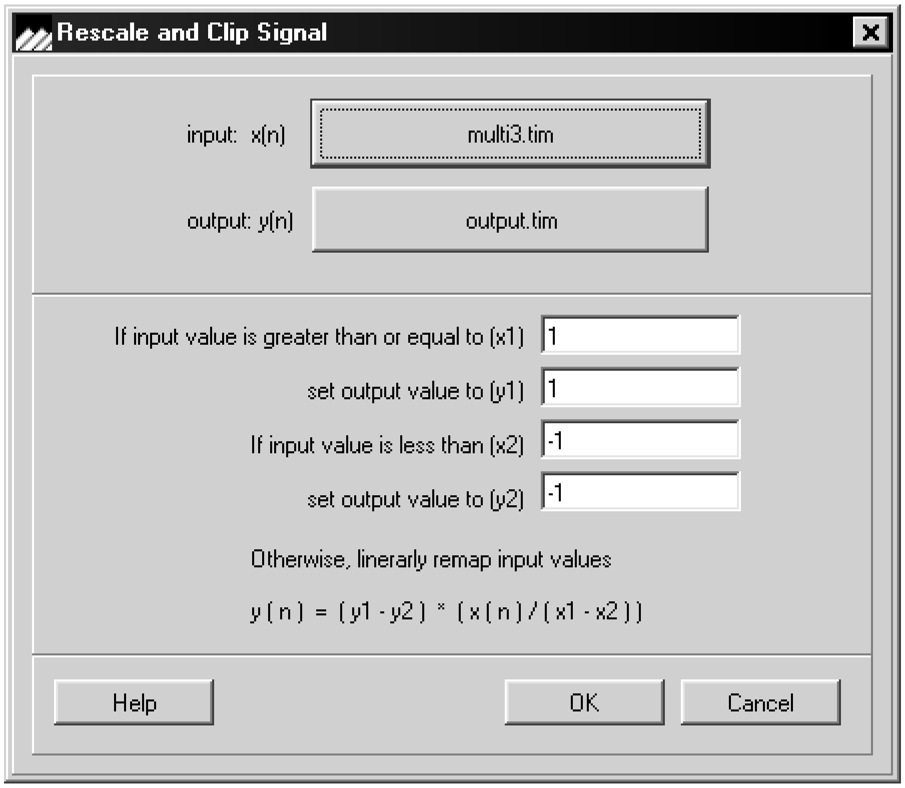
If x(n) < X2, set y(n) = Y2

If x(n) > or = X1, set y(n) = Y1

If x(n) is strictly between X2 and X1, linearly map y(n) between Y2 and Y1

Note: These rules imply that X2 is < or = X1 and all three conditions are mutually exclusive.

FIGURE 6.25 Rescale and Clip


Rescale and Clip Signal

input: x(n) multi3.tim

output: y(n) output.tim

If input value is greater than or equal to (x1) 1

set output value to (y1) 1

If input value is less than (x2) -1

set output value to (y2) -1

Otherwise, linerarly remap input values

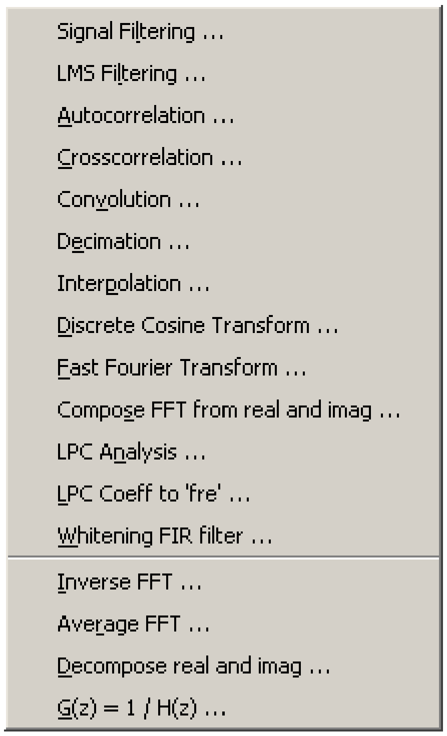
y (n) = (y1 - y2) * (x (n) / (x1 - x2))

CHAPTER 7 *DSP Menu*

This section describes basic DSP functions for manipulating discrete time data sequences and performing most common digital signal processing operations such as Signal Filtering, Autocorrelation, Fast Fourier Transform and LPC analysis.

To perform each of these functions time file inputs are needed. As before, brief descriptions of each function and explanations of dialog boxes are presented. The ***DSP*** menu when selected is shown in Figure 7.1:

FIGURE 7.1 **DSP Menu**



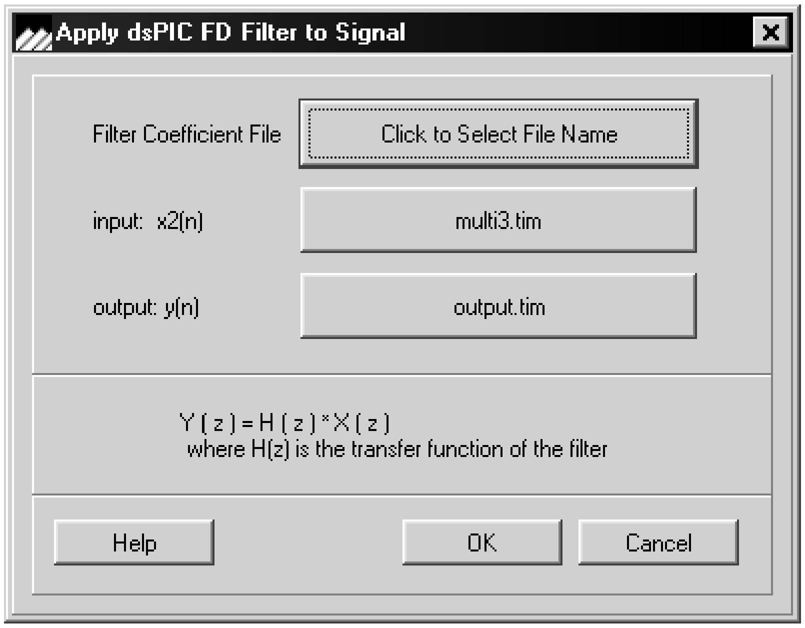
7.1 Signal Filtering

Signal filtering allows the filtering of a time domain signal using the filter coefficients generated by dsPIC Filter Design Tool. This allows IIR filters to be implemented as difference equations using the filter coefficients rather than convolving the impulse response of the filter with the signal.

The dsPIC Filter Design Tool will create a file with the quantized filter coefficients. These files are all suffixed by '.FLT'. Thus to apply a filter to a time domain signal, the filter must be designed and an (.FLT) file with the coefficients must be created. There are two types of '.FLT' filter coefficient files - one for floating point coefficients and one for fixed point fractional. The output file data type is set according to the input file type. Computations are performed in either fixed point fractional or floating point depending on the filter coefficient type. If the sampling rate in an FLT file does not match the sampling rate of a time domain signal, a frequency shift will occur.

The dialog box as shown in Figure 7.2 is used to obtain the filenames for the filter coefficient file, the input signal and the output signal.

FIGURE 7.2 Apply a Filter to a Signal



Filter Coefficient File: Click to Select File Name
input: x2(n): multi3.tim
output: y(n): output.tim

Y(z)=H(z)*X(z)
where H(z) is the transfer function of the filter

7.2 LMS Adaptive Filter

The LMS Adaptive filter creates an FIR filter where the coefficients are altered on each input sample up to the specified number of iterations.

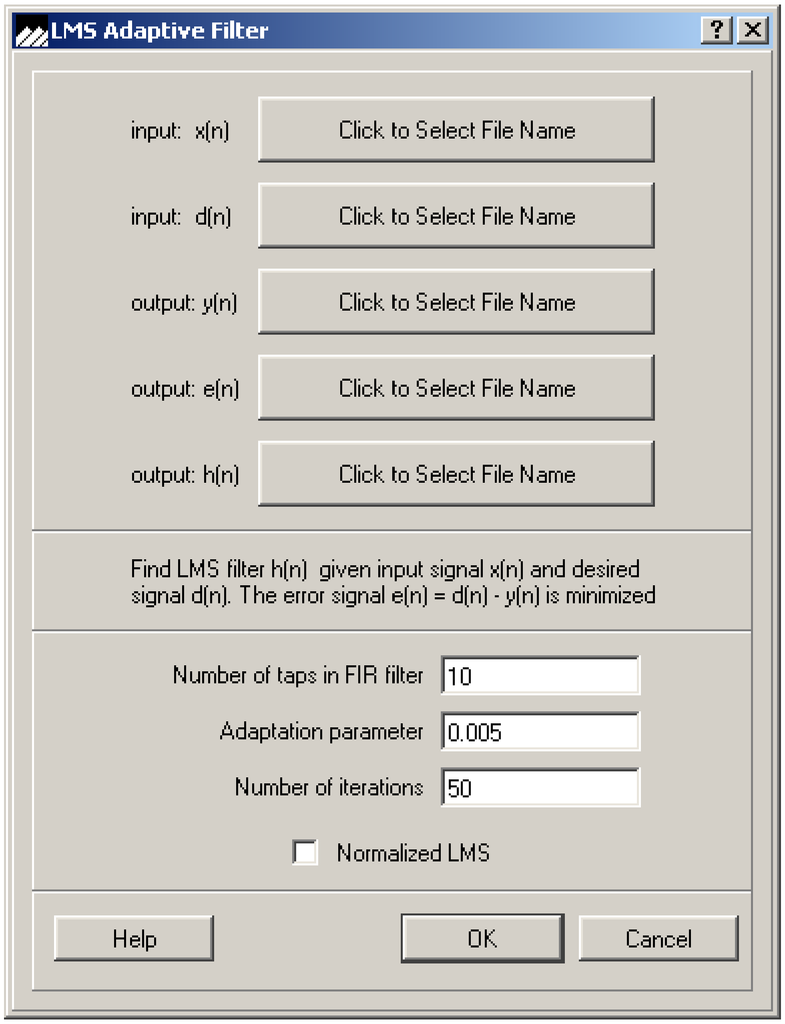
The input signal and the desired signal are specified as x(n) and d(n) respectively.

The output of the filter is y(n) and the difference between y(n) and d(n) is the output e(n). When the last sample in the input file is processed, the filter h(n) is written as a file.

The dialog box as shown in Figure 7.3 on page 88 is used to obtain the input filename x(n), the desired output filename y(n), the error filename e(n) and the filter filename h(n).

See "Lattice Filters" on page 135 for implementation recommendations on Lattice Filters.

FIGURE 7.3 **LMS Adaptive Filter Parameters**



LMS Adaptive Filter

input: x(n) Click to Select File Name

input: d(n) Click to Select File Name

output: y(n) Click to Select File Name

output: e(n) Click to Select File Name

output: h(n) Click to Select File Name

Find LMS filter h(n) given input signal x(n) and desired signal d(n). The error signal e(n) = d(n) - y(n) is minimized

Number of taps in FIR filter: 10

Adaptation parameter: 0.005

Number of iterations: 50

☐ Normalized LMS

There are three parameters that control the operation of the LMS filter: ***Number of Taps*** in the filter, ***Adaptation parameter*** and ***Number of Iterations***.

The filter length is specified by the number of taps. The filter is initialized to zero for each filter coefficient.

For each iteration up to the specified number of iterations, the FIR filter coefficients are modified as follows:

$$\omega_{\boldsymbol{i}}(\boldsymbol{n}+1) = \omega_{\boldsymbol{i}}(\boldsymbol{n}) + \upsilon \boldsymbol{en})\boldsymbol{x}(\boldsymbol{n}) \qquad \textbf{(EQ 7.1)}$$

where u is the adaptation parameter and i=0,...,N and N= number of taps-1. If the normalized LMS box is checked, the adaptation parameter is also modified on each iteration.

7.3 Autocorrelation

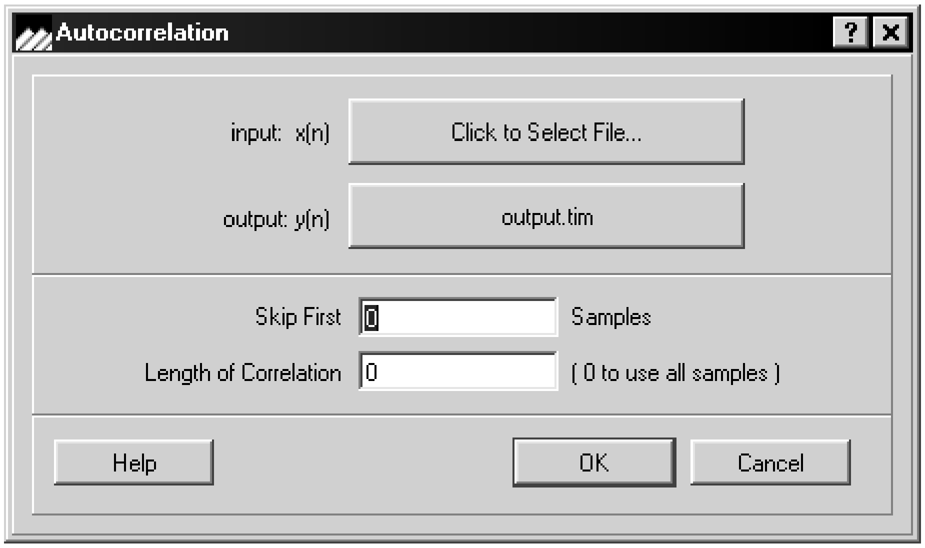
Autocorrelation is a special case of cross correlation to be discussed in the next section. The autocorrelation sequence indicates which samples of a sequence are related. This function computes the autocorrelation estimate for a given sequence using the following estimation approach:

$$r_{xx}(k) = \sum_{n=k}^{N-1} x(n)x(n-k)(\text{where K} \geq 0) \qquad \textbf{(EQ 7.2)}$$

The autocorrelation function is symmetric about zero. The system shifts the autocorrelation function so that the function is defined for positive time only.

The dialog box as displayed in Figure 7.4 is used to obtain the parameters for the Autocorrelation option. The input is a time domain waveform either generated by using the Generator menu or acquired by reading data from an external source. Usually N is the length of the sequence, however. dsPICworks software provides the flexibility of performing autocorrelation on a subset of a given sequence by specifying the first sample and length of correlation. Note that the length of the output sequence is 2N-1.

FIGURE 7.4 Autocorrelation of a Signal



input: x(n) Click to Select File...

output: y(n) output.tim

Skip First 0 Samples

Length of Correlation 0 (0 to use all samples)

7.4 Crosscorrelation

In contrast to Autocorrelation, two signal sequences are involved in Crosscorrelation. The objective in computing the Crosscorrelation between two signals is to measure the degree to which the two signals are similar. The following formula is used to estimate the cross-correlation of two signals x1(n) and x2(n)

$$r_{x1x2}(l) = \sum_{n=l}^{N-|k|-1} x1(n)x2(n-l) \qquad \text{for } l \geq 0 \qquad \textbf{(EQ 7.3)}$$

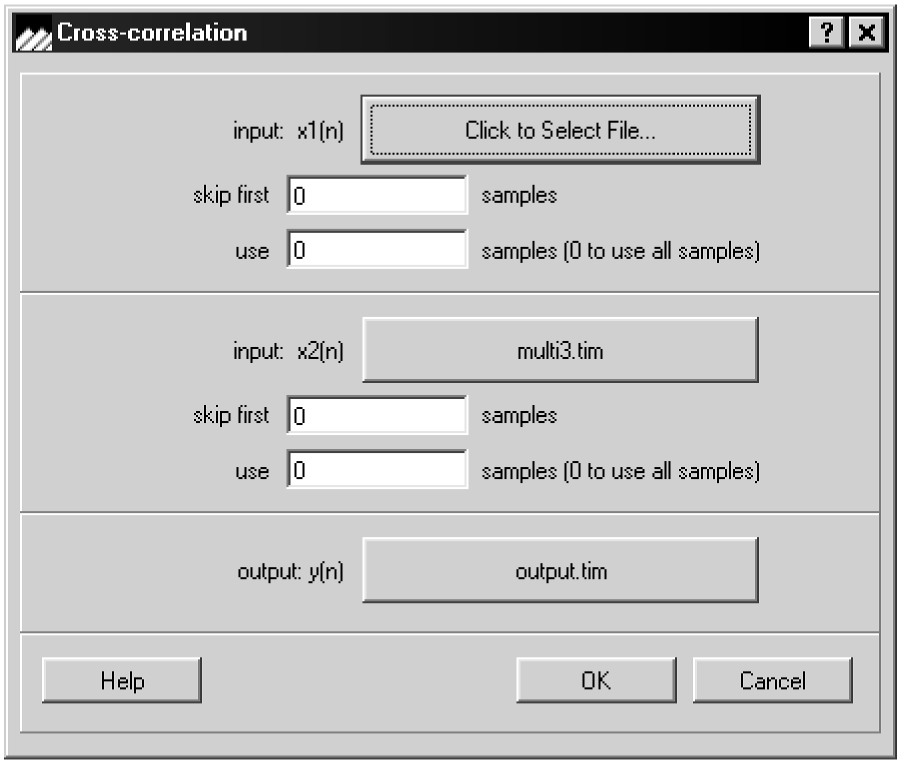
$$r_{x1x2}(l) = \sum_{n=l}^{N-1} x1(n)x2(n-l) \qquad \text{for } l \geq 0, \ r=l, \ k=0 \qquad \textbf{(EQ 7.4)}$$

$$r_{x1x2}(1) = \sum_{n=0}^{N-|l|-1} x1(n)x2(n-l) \qquad \text{for } l < 0, \ r=0, \ k=l \qquad \textbf{(EQ 7.5)}$$

The result is shifted to the right so that r_{x1x2} is defined for $n \geq 0$.

Assuming x1(n) and x2(n) are causal sequences of length N; if the length of one of the sequences is shorter than N then it is zero padded. The dialog box as displayed in Figure 7.5 is used to obtain the parameters for the Crosscorrelation option.

FIGURE 7.5 **Crosscorrelation between two Signals**



Cross-correlation

input: x1(n) [Click to Select File...]

skip first [0] samples

use [0] samples (0 to use all samples)

input: x2(n) [multi3.tim]

skip first [0] samples

use [0] samples (0 to use all samples)

output: y(n) [output.tim]

On both x1(n) and x2(n), it is possible to skip a number of points before starting the cross-correlation calculation. Also, the number of samples used in the calculation can be controlled for each file by setting a non-zero value in the number of samples to use field. If there are N samples from the first sequence and M samples from the second sequence, then the number of entries in the crosscorrelation sequence is N+M-1.

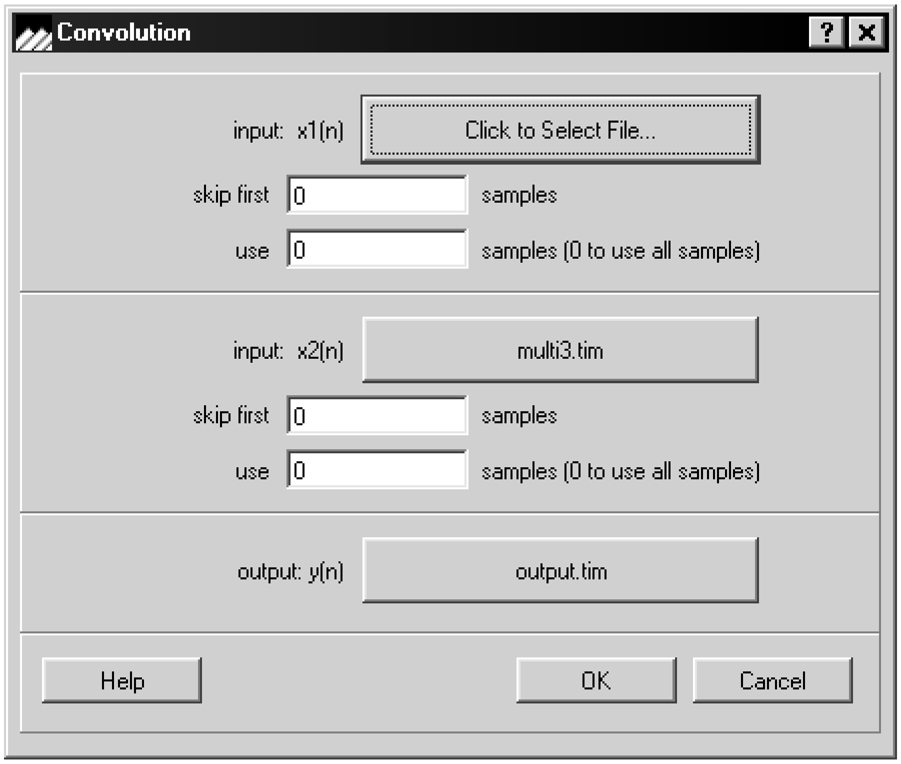
7.4.1 Convolution

This operation computes the linear convolution of two sequences. Consider a sequence x1(n) whose length is L points and a sequence x2(n) whose length is P points. The linear convolution of these two sequences is computed as follows:

$$y(n) = \sum_{k=0}^{n} x1(k)x2(n-k) = \sum_{k=0}^{n} x1(n-k)x2(k) \qquad \text{(EQ 7.6)}$$

The output sequence y(n) contains a most (L+P-1) samples. The primary application of convolution is to compute the response of a relaxed linear time invariant system. The dialog box as displayed in Figure 7.6 is used to obtain the input filenames and the output filename for the convolution operation.

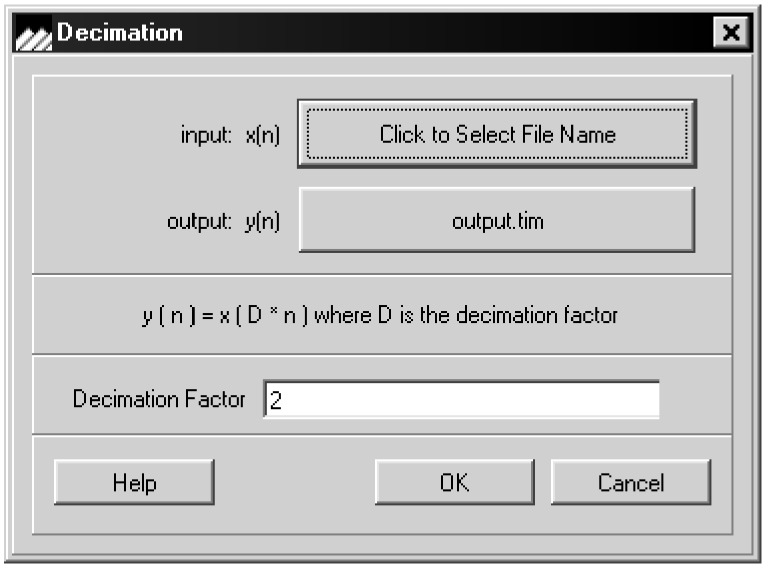
FIGURE 7.6 **Convolution Operation**



7.5 Decimation

This operation as displayed in Figure 7.7 forms a new time domain sequence y(n) by selecting every Nth term of the input sequence x(n), where N is input as the factor. The sampling rate of the output file is the input sampling rate divided by the decimation factor. To avoid aliasing the output file should be pre-processed with a lowpass filter where the cutoff frequency is approximately the new sampling rate.

FIGURE 7.7 **Decimation Operation**



input: x(n) Click to Select File Name

output: y(n) output.tim

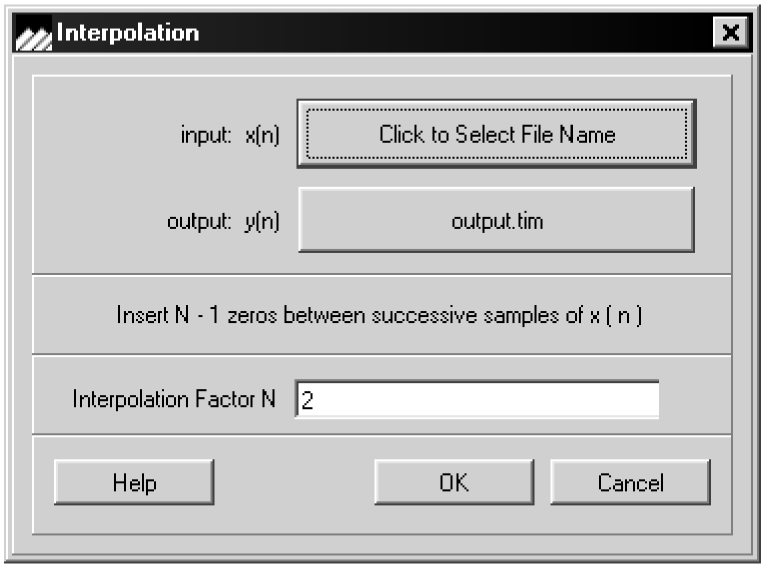
y (n) = x (D * n) where D is the decimation factor

Decimation Factor 2

7.6 Interpolation

This operation as shown in Figure 7.8 forms a new time domain sequence y(n) by inserting zeros between successive samples of the input sequence x(n). The number of zeros is specified on the input screen by specifying the interpolation factor. Note that inserting one zero value between successive input samples is equivalent to interpolating by a factor of 2. The sampling rate of the output file is the input sampling rate multiplied by the interpolation factor. To prevent imaging in the output file, this file should be post-processed with a lowpass filter where the cutoff frequency is approximately the original sampling rate.

FIGURE 7.8 **Interpolation Operation**


Interpolation

input: x(n) Click to Select File Name

output: y(n) output.tim

Insert N - 1 zeros between successive samples of x (n)

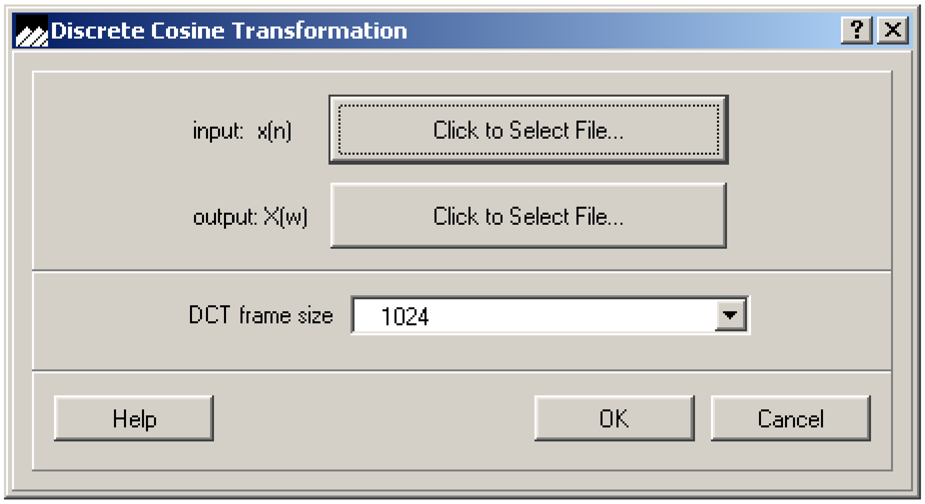
Interpolation Factor N 2

The gain of the output file should be multiplied by the interpolation factor L if the gain is to remain at 0 dB compared to the input file.

7.7 Discrete Cosine Transform

The Discrete Cosine Transform implements a fast Discrete Cosine Transform type II transform. This Discrete Cosine Transform is typically used for video and audio compression. A time signal is the input and a frequency file is the output of this operation. There is only one parameter - the Discrete Cosine Transform frame size. The file names and Discrete Cosine Transform frame size are shown in Figure on page 96.

FIGURE 7.9 **Discrete Cosine Transform Operation**



input: x(n)
output: X(w)
DCT frame size 1024

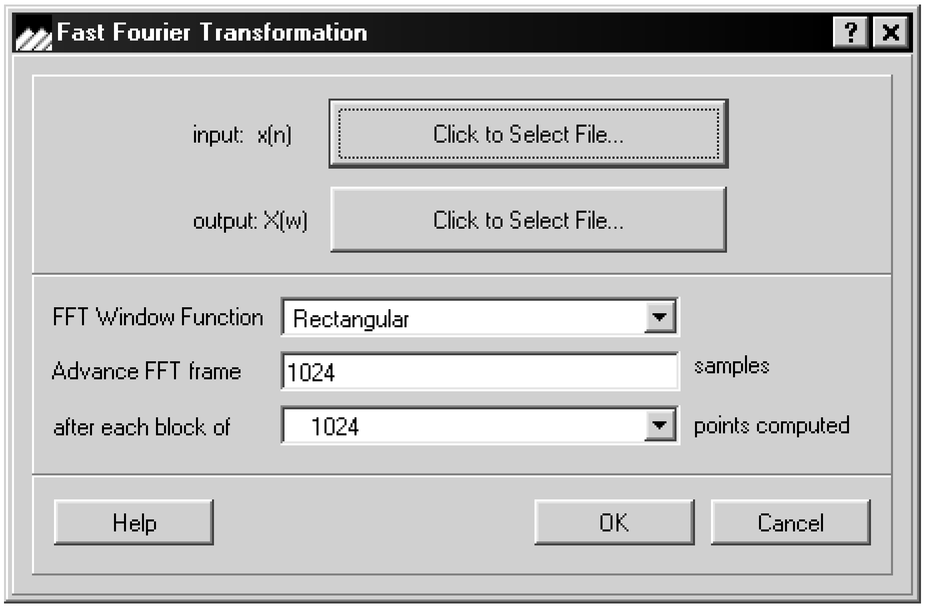
7.8 Fast Fourier Transform

Fast Fourier Transform (FFT) is a collection of efficient algorithms used to compute Discrete Fourier Transform (DFT) which plays an important role in the analysis, design and implementation of discrete time signal processing algorithms and systems. FFT algorithms are based on the fundamental principle of decomposing the computation of the DFT of a sequence of length N into successively smaller DFTs. dsPICworks software utilizes radix-2 decimations in time to implement FFTs. the number of time domain samples used in the calculation of the FFT is always the same as the number of points in the FFT calculation. For example, if the FFT length (number of computation points) is set at 1024, then 1024 points from the time domain file will be used in the computation independent of the value in the 'Advance FFT' field. This field can be used to create frequency domain blocks from overlapped time domain data or to skip time domain samples between successive FFTs. Only the last frame of data will be zero padded to fill the frame.

To perform FFTs on overlapped time domain blocks of data, set the 'Advance FFT' frame to a value less than the FFT length (number of computation points). This will cause more blocks of frequency domain data in the output file than the number of non-overlapped blocks of data in the time domain file. To skip time domain data values in the FFT calculation, set 'Advance FFT' frame to a value greater than the FFT length (number of computation points).

The dialog box shown in Figure 7.10 appears if *FAST FOURIER TRANSFORM* was selected on the ***DSP*** menu.

FIGURE 7.10 **Fast Fourier Transform**


Fast Fourier Transformation

input: x(n) Click to Select File...

output: X(w) Click to Select File...

FFT Window Function: Rectangular

Advance FFT frame: 1024 samples

after each block of: 1024 points computed

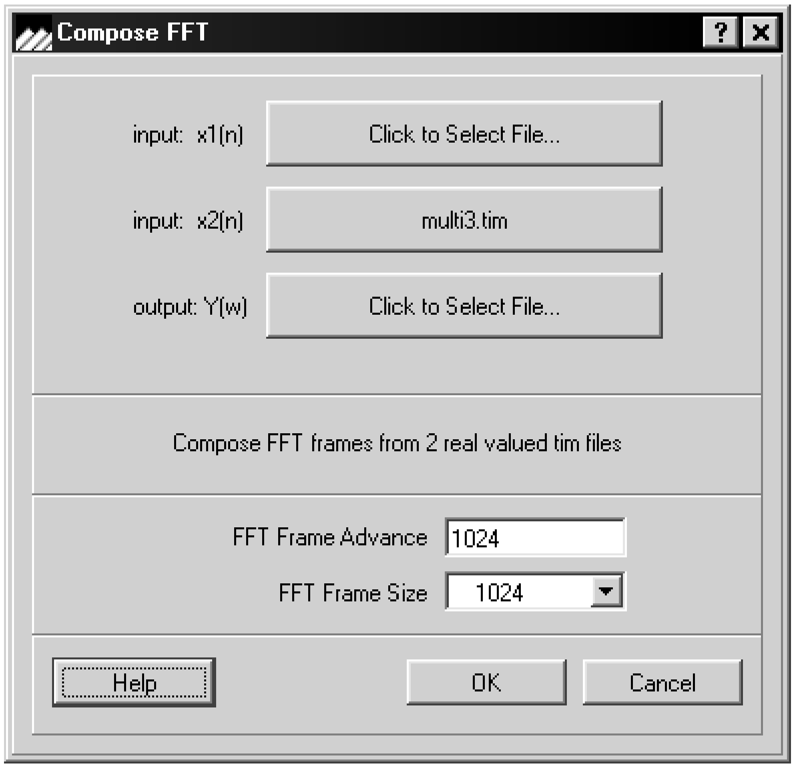
A time domain file must be specified for input and the output filename must be entered for the output of the FFT calculations.

The output file will be created with an extension of '.FRE'. The frequency domain file consists of a series of blocks of data, each block corresponding to one FFT calculation. The parameters of the calculation are stored in the header record so that the *Frequency/ Inverse FFT* can reverse the process and essentially create the original waveform. Note that the frequency domain file data is stored in polar format representing magnitude and phase. If the time domain file data values are to be windowed, click on the FFT window function field to select the desired window function as per the following dialog box. For further information on the Window functions, please refer to Section 5.7 on page 52 ***GENERATOR/****WINDOW* functions.

7.9 Compose FFT from Real and Imaginary Files

This function as shown in Figure 7.11 allows the creation of a frequency file from two timefiles. This function in conjunction with the decompose operation allows arithmetic operations to be active on the real and imaginary parts separately and then composed back together again. The frame size and advance size are required for creating frequency files, but these parameters are not actually used in this operation.

FIGURE 7.11 Compose FFT from Real and Imaginary Files Operation



Compose FFT frames from 2 real valued tim files

FFT Frame Advance 1024

FFT Frame Size 1024

7.10 LPC Analysis

This menu selection performs Linear Predictive Analysis (LPC.)

In the discrete-time model of speech production speech is synthesized by sending a sequence of impulses for voiced speech or a sequence of white noise for unvoiced speech to a time-varying digital all-pole filter having transfer function H(z):

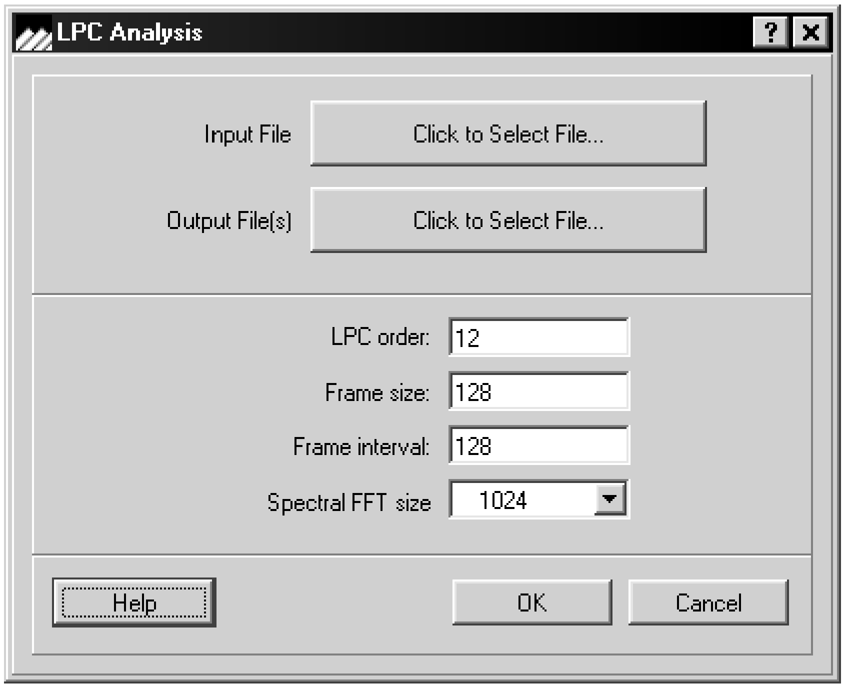
$$H(z) = \frac{G}{1 - \sum_{k=1}^{p} a_k z^{-k}} \qquad \textbf{(EQ 7.7)}$$

In LPC Analysis the coefficients a_k are assumed to be constant for a short time segment of speech and a variety of techniques can be used to estimate the coefficients a_k from this short time segment of sampled speech.

In dsPICworks software the Harmonic-Mean Lattice Solution is implemented. The coefficients a_k and the reflection coefficients k_m are the output of this operation. Spectral responses are also computed from a_k as part of the LPC Analysis. Spectral responses can also be computed as a separate operation without performing *LPC ANALYSIS* via the menu selection '*LPC COEFF. TO '.FRE*'...'. All computations are carried out using double precision floating point numbers.

When the '*LPC ANALYSIS*...' menu selection is selected, the dialog box as displayed in Figure 7.12 appears:

FIGURE 7.12 **LPC Analysis screen**


LPC Analysis

Input File: Click to Select File...

Output File(s): Click to Select File...

LPC order: 12

Frame size: 128

Frame interval: 128

Spectral FFT size: 1024

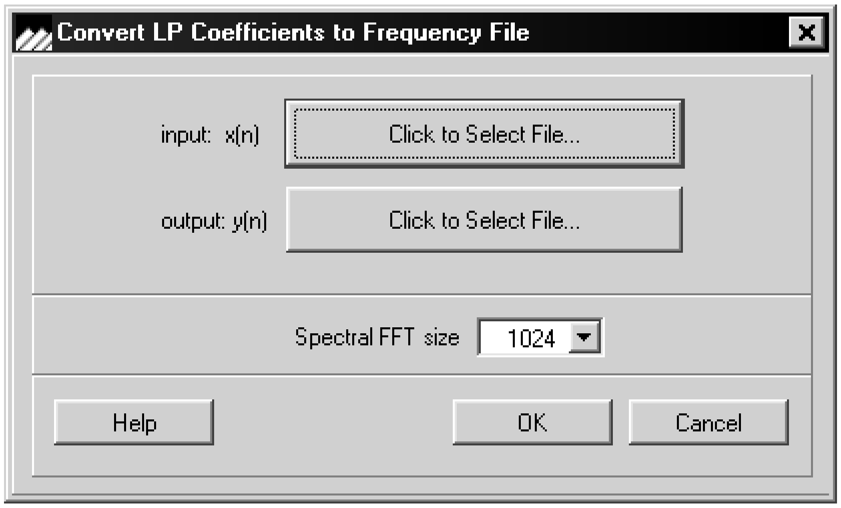
TABLE 7-1 LPC Functions

Input	Function
Input file	Select a time file for analysis
LPC order	The order of the denominator polynomial; also equals the number of coefficients.
Frame size	The length of the short time segment in number of samples. The rule of thumb is using segment containing approximately 10 mSec to 50 mSec of speech
Frame interval	The interval between the left edge of each successive frame:
Spectral FFT size	Upon completion of the LPC Analysis, the spectrum response is computed by evaluating H(z) using the a_n's on the unit circle (where $Z = e^{-j\omega}$).) The resulting file format is identical to the Fast Fourier Transformation. This parameter represents the FFT size
Output file	Enter the file name for the output file. There are two output files: <filename>.lpc file contains the all-pole filter coefficients and the reflection coefficients, <filename>.fre file contains the spectral response

7.11 LPC Coeff. to '.fre'...'

This operation computes the spectral responses from previously computed LPC Analysis. The dialog box as displayed in Figure 7.13 is used to obtain the required information.

FIGURE 7.13 Compute Spectral Response



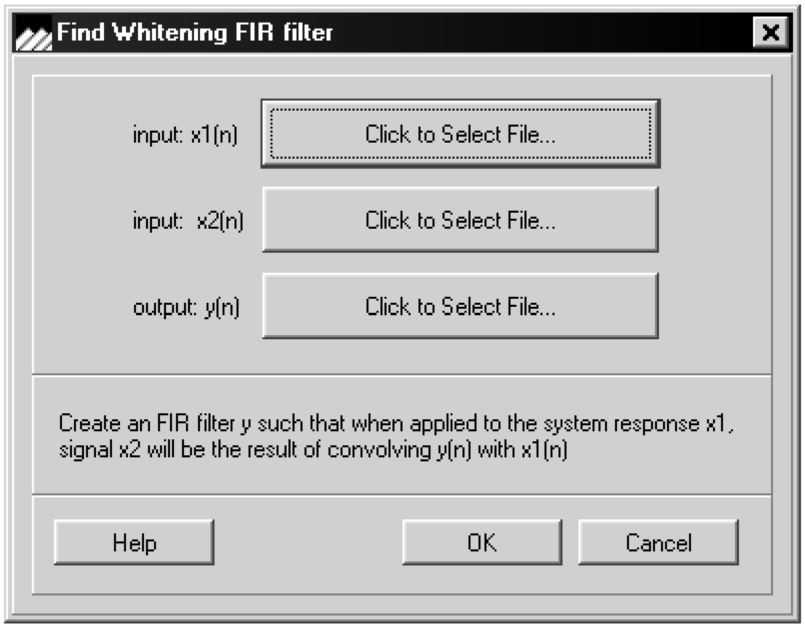
Spectral FFT size 1024

An input file of LPC coefficients must be selected. The size of the FFT must be entered. The spectrum response is computed by evaluating H(z) using the LPC coefficients on the unit circle. The output file name must be selected. The resulting file format is identical to the FFT. The output file will automatically have the.FRE extension.

7.12 Whitening FIR Filter

This operation will compute a Whitening FIR (or inverse Wiener filter). This filter when convolved with an impulse response of a system will give a delayed impulse response. Since the spectral content of an impulse response is a constant this operation is usually referred to as a "Whitening" filter. The method used to calculate the inverse filter is the Levinson-Durbin Recursion Equations for solving the Toeplitz matrix system. The dialog box as displayed in Figure 7.14 is used to input the required parameters.

FIGURE 7.14 Whitening FIR Filter Input



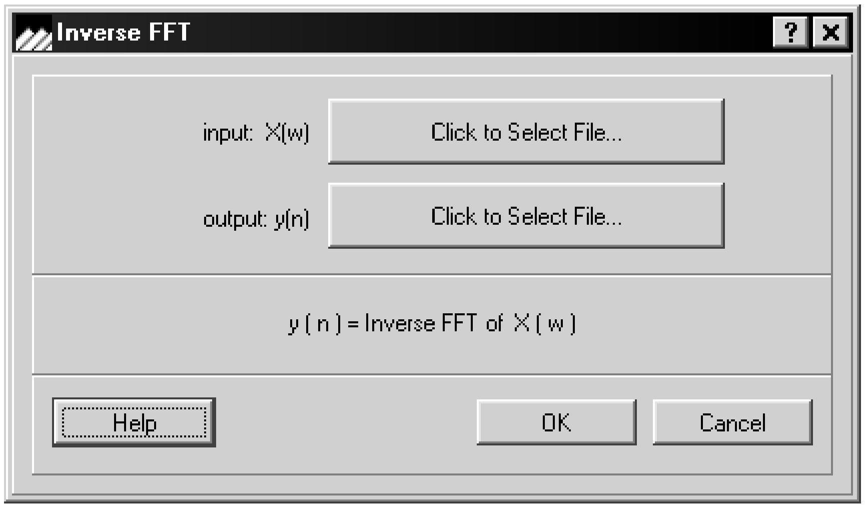
Create an FIR filter y such that when applied to the system response x1, signal x2 will be the result of convolving y(n) with x1(n)

Three filenames are required. The first filename is for the impulse response of a system, the second filename is the desired response - usually a delayed impulse, and the third file-name is for the resulting inverse Wiener filter.

7.13 Inverse FFT

The Inverse Discrete Fast Fourier Transform is closely related to the Discrete Fourier Transform in that a given signal x(n) can be reconstructed from its discrete Fourier Transform X(ω) by using the Inverse Discrete Fourier Transform. dsPICworks software implements the Inverse Discrete Fourier Transform by using an Inverse Fast Fourier Transform. The dialog box shown in Figure 7.15 is used to obtain the required parameters.

FIGURE 7.15 **Inverse FFT Input**

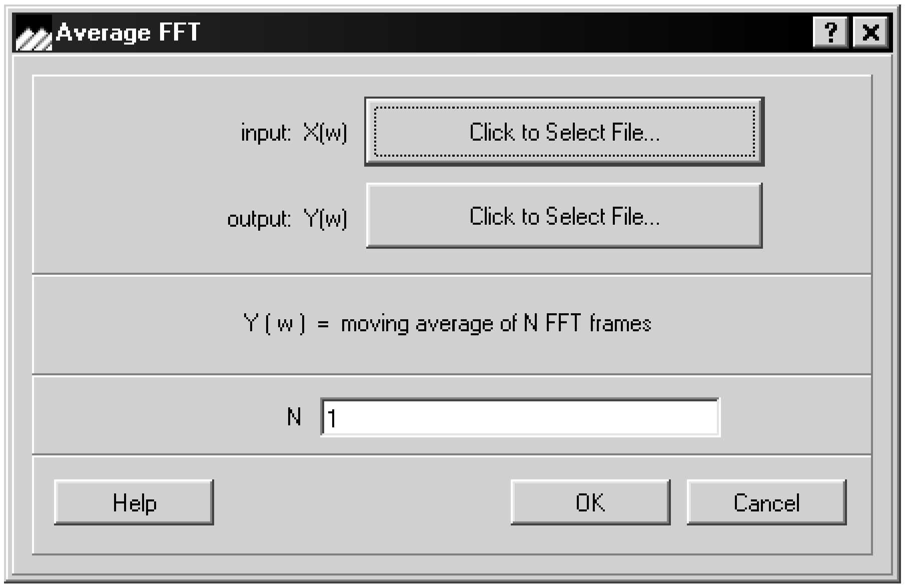


The parameters for the Inverse FFT (filter length, window type, overlap) are extracted from the header of the input frequency domain file. At least one overlapping point in the original FFT is required for this function. The result of the inverse FFT is divided by the original windowing function. This allows exact recreation of the original time domain The parameters for the Inverse FFT (filter length, window type, overlap) are extracted from the header of the input frequency domain file. At least one overlapping point in the original FFT is required for this function. The result of the inverse FFT is divided by the original windowing function. This allows exact recreation of the original time domain signal.

7.14 Average FFT

This function averages the magnitude of a specified number of FFT frames.The input frequency file and the output frequency file and the number of FFT frames are input in the dialog box as shown in Figure 7.16.

FIGURE 7.16 **Average FFT function**



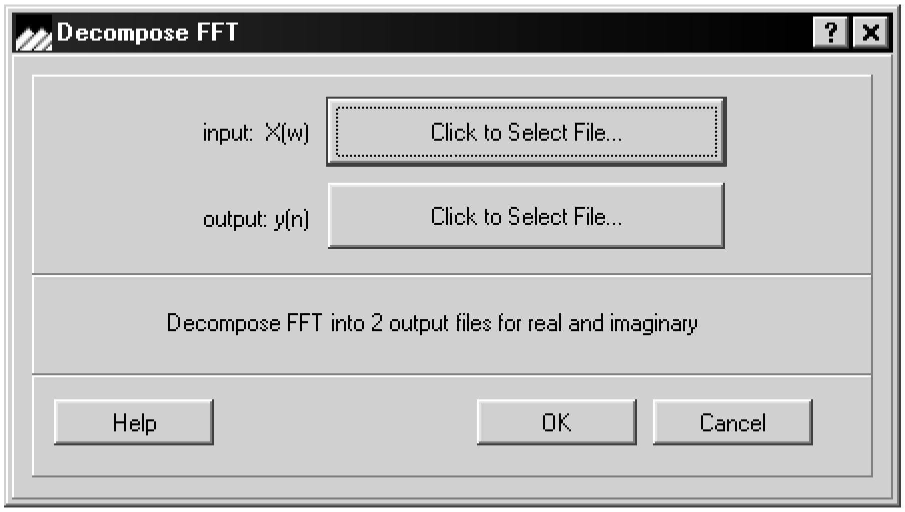
Y (w) = moving average of N FFT frames

N 1

7.15 Decompose Real & Imaginary

This function as shown in Figure 7.17 decomposes the complex valued frequency function into two files - one is for the real and the other for the imaginary part. The resulting real part signal file will be suffixed with an ‘r’, and the imaginary part file will be suffixed with an ‘i’.

FIGURE 7.17 **Decomposition**



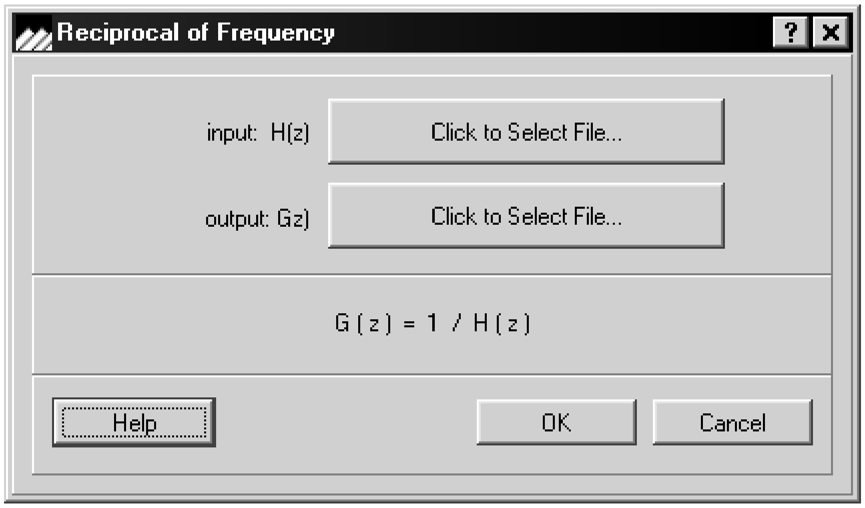
Decompose FFT into 2 output files for real and imaginary

7.16 Reciprocal of a frequency file

This operation computes G(z) = 1/H(z) on an element by element basis.

Use dialog box as displayed in Figure 7.18 appears to obtain input and output frequency filenames:

FIGURE 7.18 Reciprocal Function



CHAPTER 8 *Display Menu*

This section describes functions featured in dsPICworks software that display stored time-domain waveforms and frequency spectra.

dsPICworks software stores time-domain waveform data natively in *.TIM files and complex spectral data in *.FRE files. These files may be generated within dsPICworks software or imported from other data files. The functions in the ***DISPLAY*** menu are used to plot the *.TIM and *.FRE files on graphs that are displayed on screen.

An explanation detailing each menu item of the ***DISPLAY*** menu and associated dialog boxes will be presented next. However, we first define some terms below to clarify the display functions.

The frequency response of a signal h(t) is as follows:

$$F\{h(t)\} = |H(j\omega)|e^{-j\Theta(\omega)} \text{ where F\{ \} is the Fourier Transform} \qquad \textbf{(EQ 8.1)}$$

where

the magnitude display is represented by:

$$|H(j\omega)| \qquad \textbf{(EQ 8.2)}$$

the power display is represented by:

$$|H(j\omega)|^2 \qquad \textbf{(EQ 8.3)}$$

and the phase display is represented by:

$$\Theta(\omega) \tag{EQ 8.4}$$

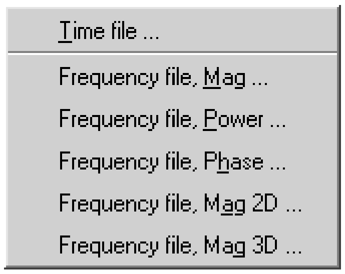
The waveform response of a signal is:

$$y(n) = f(x(n)) \tag{EQ 8.5}$$

where x(n) is a sample at time nT.

Note: Shortcuts exist for both time files and one-dimensional frequency files from the dsPICworks toolbar. The ***DISPLAY*** menu when selected is shown in Figure 8.1.

FIGURE 8.1 **Display Menu**


Time file ...
Frequency file, Mag ...
Frequency file, Power ...
Frequency file, Phase ...
Frequency file, Mag 2D ...
Frequency file, Mag 3D ...

8.1 Displaying Time and Frequency Files

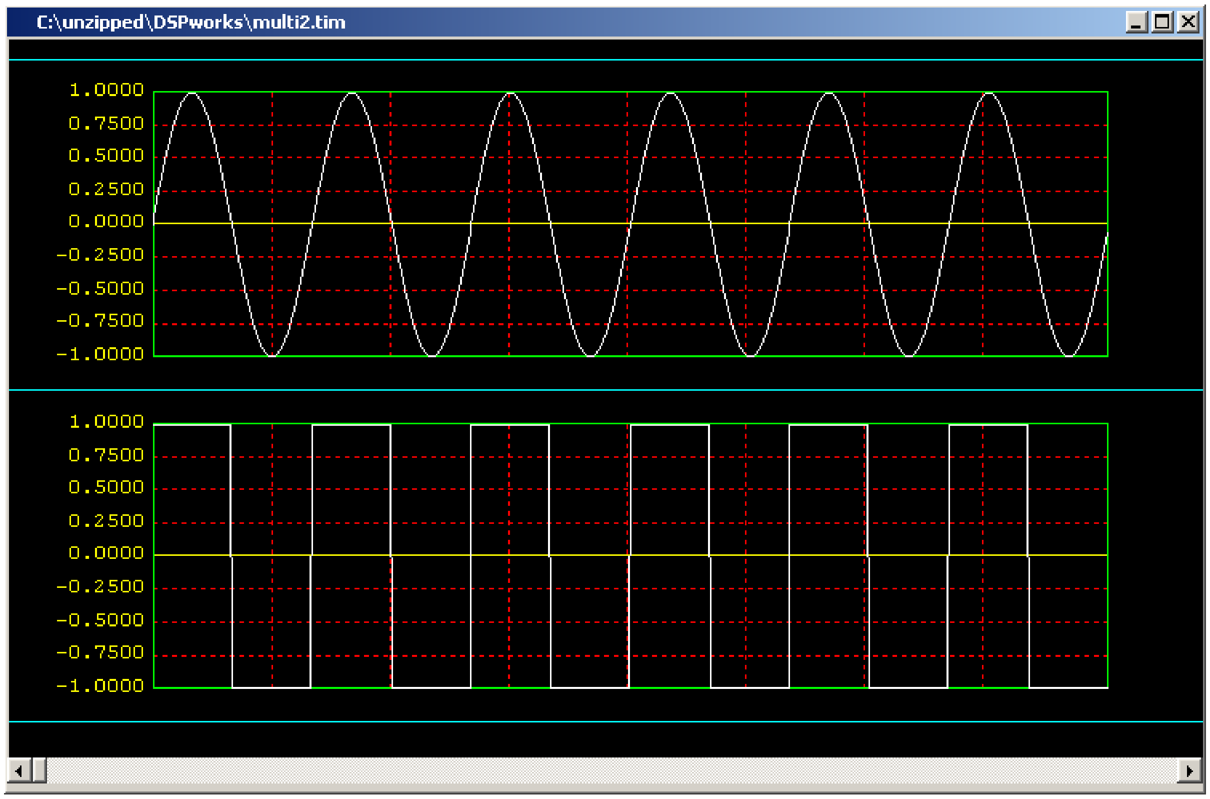
After selecting ***TIME FILE***, a standard file open dialog box will appear with all files in the files list having Time extensions (.TIM). Selecting a file will cause that file to be displayed. Waveform editing and cursor tracking with automatic readout of amplitude and time values or amplitude and sample numbers is available.

Cursor tracking is also available on the frequency displays (Magnitude (1D), Phase, Power). The function value is displayed in the upper left corner of the graph. The tracking cursor is enabled by holding down the left mouse button and moving the cursor with the graph window. Double clicking the left mouse button toggles between the frequency value readout and the FFT bin count within the current frequency frame.

On frequency, magnitude and power displays the right mouse button can be used to enclose the frequency band for power calculations. These calculations will automatically appear in the Log window.

A sample time-domain file is shown below in Figure 8.2. In this example a multi-channel time-domain signal is displayed.

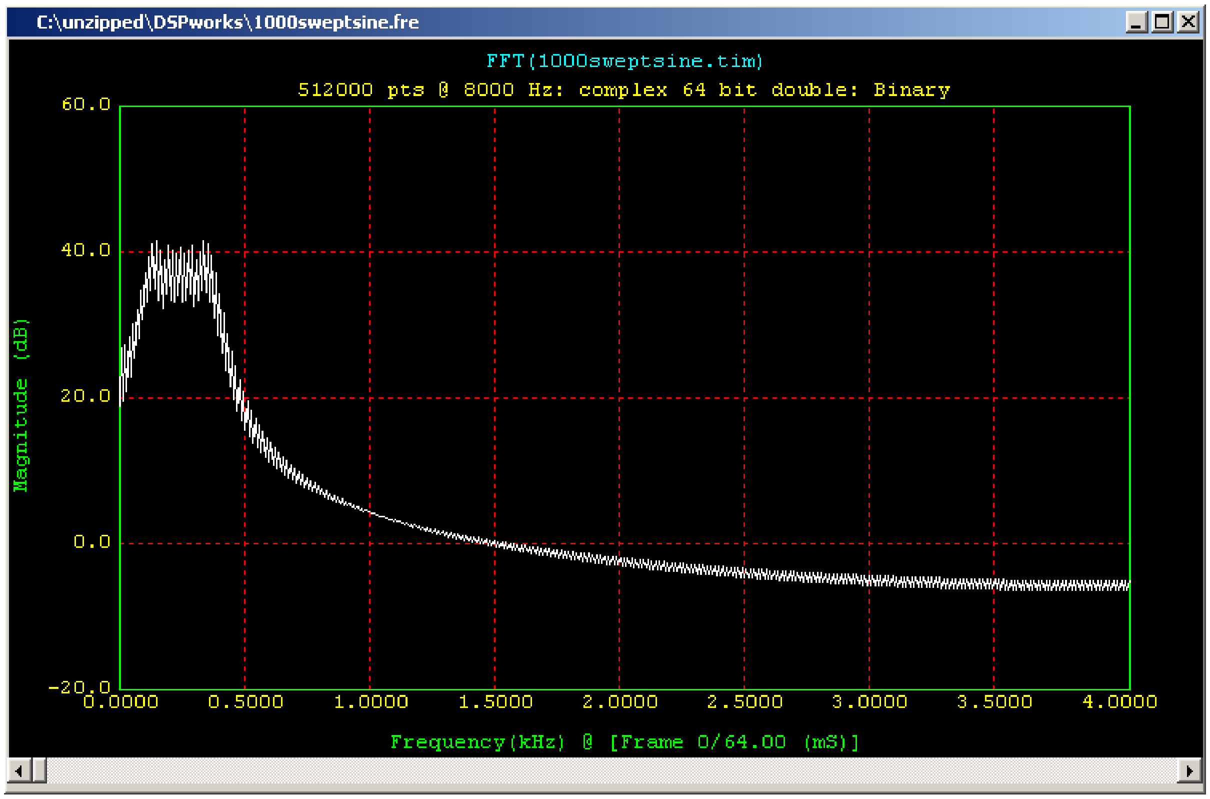
FIGURE 8.2 Multichannel Waveform Display



8.2 Magnitude Displays

These following display selections are available: 1, 2 or 3-dimensional. The user may select any of these options from the pull-down menu bar. A one-dimensional magnitude display is shown in Figure 8.3.

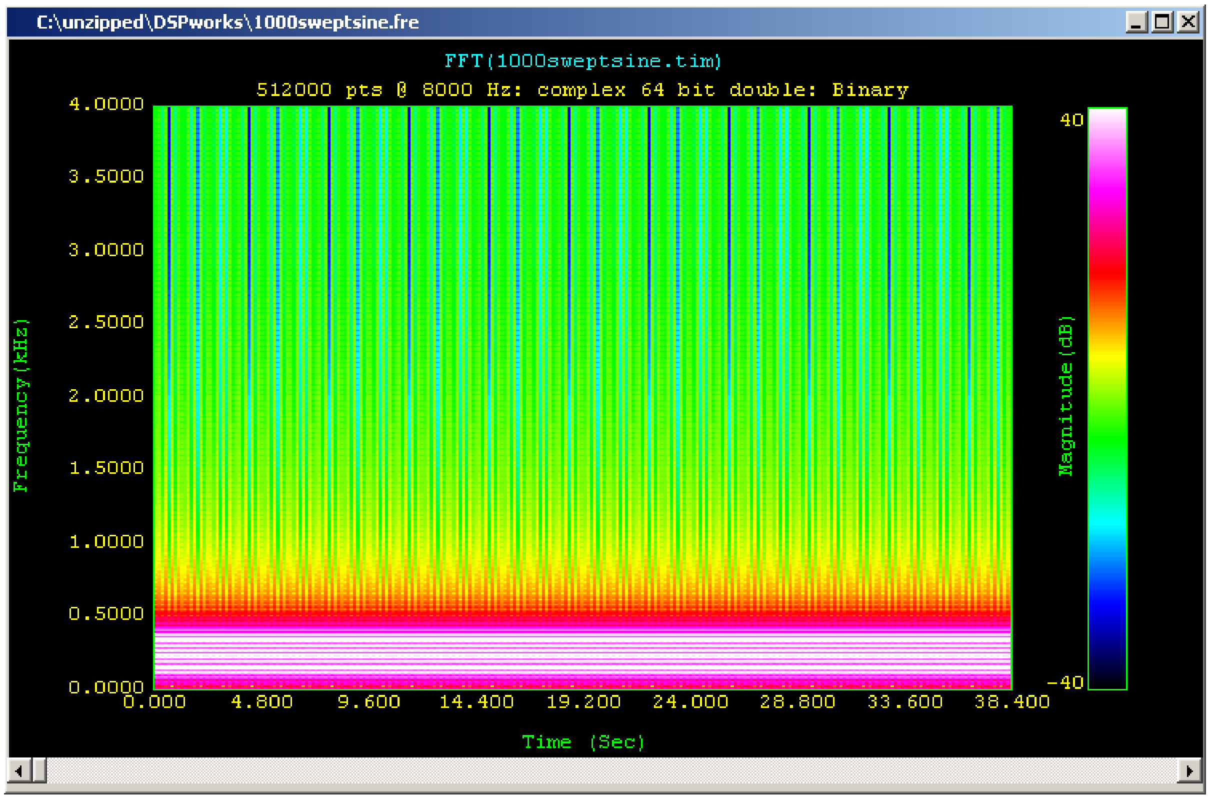
FIGURE 8.3 1D Magnitude Display


C:\unzipped\DSPworks\1000sweptsine.fre
FFT(1000sweptsine.tim)
512000 pts @ 8000 Hz: complex 64 bit double: Binary
60.0
40.0
20.0
0.0
-20.0
Magnitude (dB)
0.0000
0.5000
1.0000
1.5000
2.0000
2.5000
3.0000
3.5000
4.0000
Frequency(kHz) @ [Frame 0/64.00 (mS)]

8.2.1 2D display

The *2D DISPLAY* shows time along the x-axis and frequency along the y-axis. Thus the 2D display shows how frequency varies as a function of time. The magnitude value in dB is depicted by the color scale.

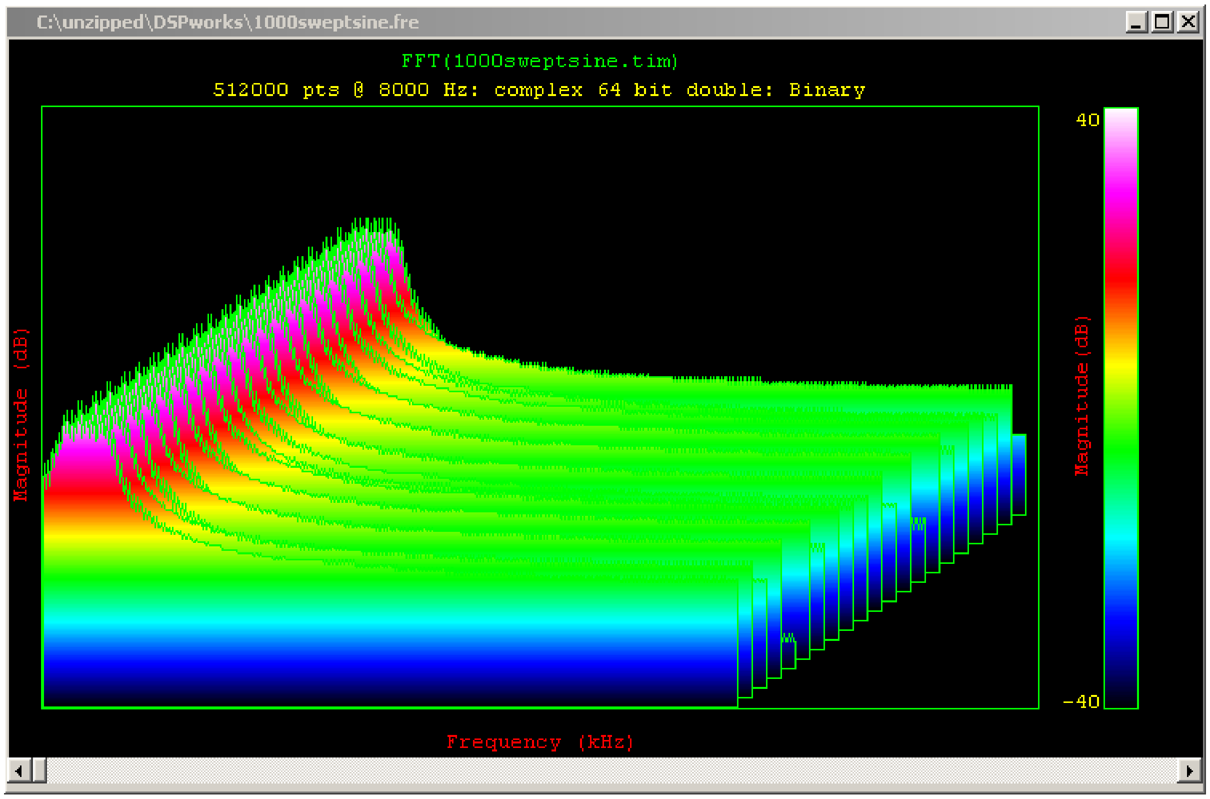
FIGURE 8.4 2D Display



8.2.2 3D display

In the 3D display frequency is shown along the x-axis, magnitude along the y-axis and time in the z-direction. The magnitude value in dB is depicted by the color scale

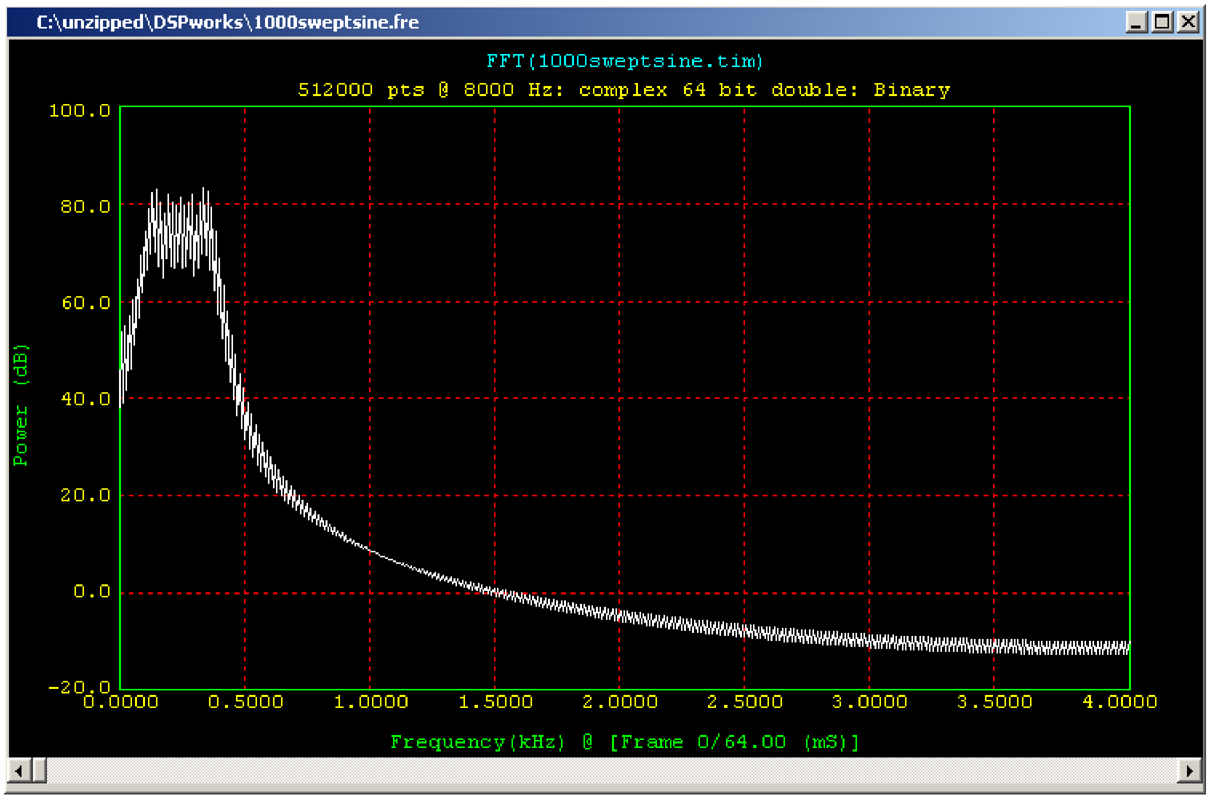
FIGURE 8.5 **3D Magnitude Display**


C:\unzipped\DSPworks\1000sweptsine.fre
FFT(1000sweptsine.tim)
512000 pts @ 8000 Hz: complex 64 bit double: Binary
Magnitude (dB)
Frequency (kHz)
Magnitude(dB)
40
-40

8.3 Power Display

Power displays may be selected by loading the desired frequency file. Again the cursor may be placed on the function to display its exact location on the upper left hand corner of the graph.

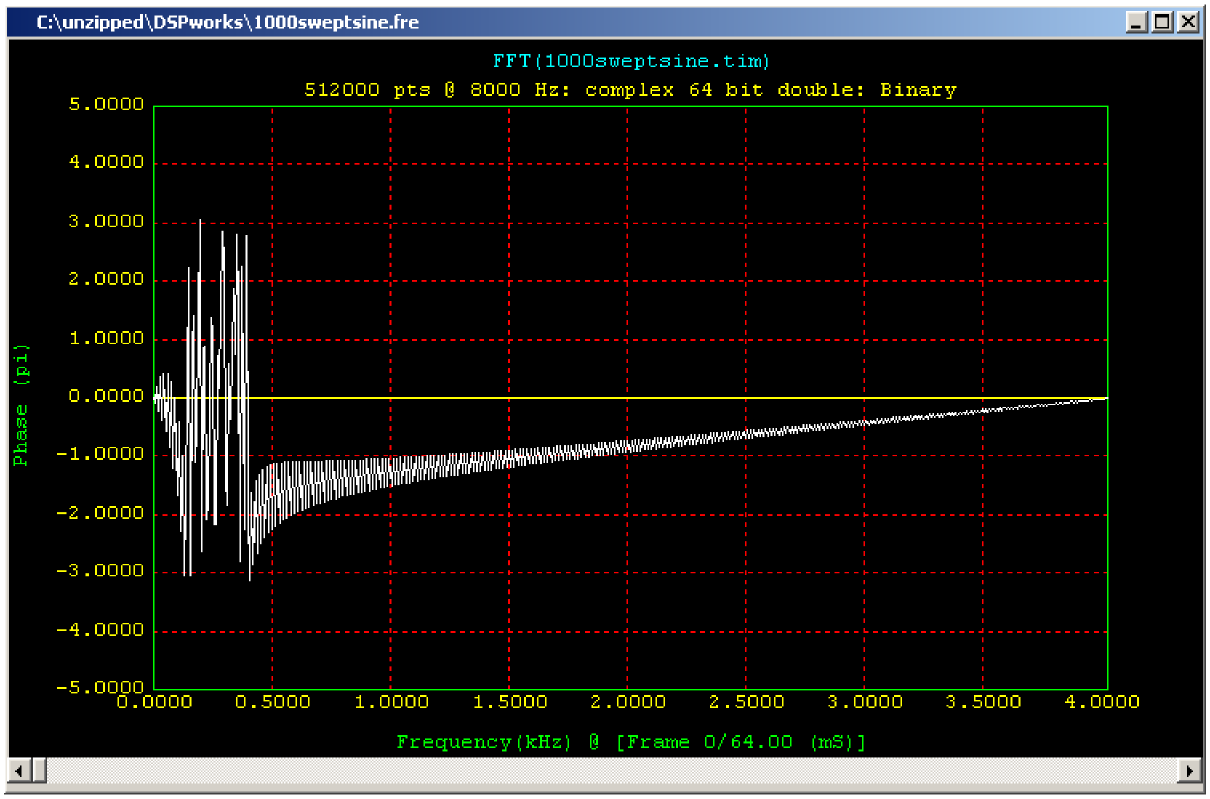
FIGURE 8.6 Stored waveform power display


C:\unzipped\DSPworks\1000sweptsine.fre
FFT(1000sweptsine.tim)
512000 pts @ 8000 Hz: complex 64 bit double: Binary
Power (dB)
100.0
80.0
60.0
40.0
20.0
0.0
-20.0
0.0000
0.5000
1.0000
1.5000
2.0000
2.5000
3.0000
3.5000
4.0000
Frequency(kHz) @ [Frame 0/64.00 (mS)]

8.4 Phase Display

Phase Displays may be selected by loading the desired frequency file. Again the cursor may be placed on the function to display its exact location on the upper left hand corner of the graph.

FIGURE 8.7 Phase Display

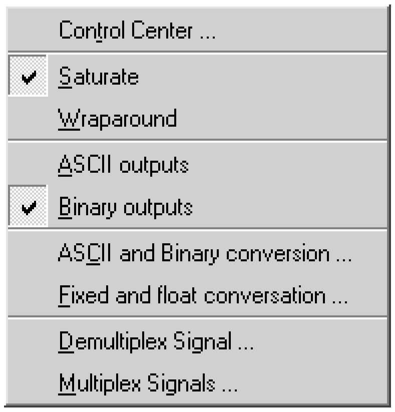


CHAPTER 9 *Utilities Menu*

This section describes various functions which are used to facilitate processing such as conversion between file types, and number types.

An explanation follows for each menu item and the associated dialog boxes. The ***UTILITIES*** menu when selected is shown in Figure 9.1.

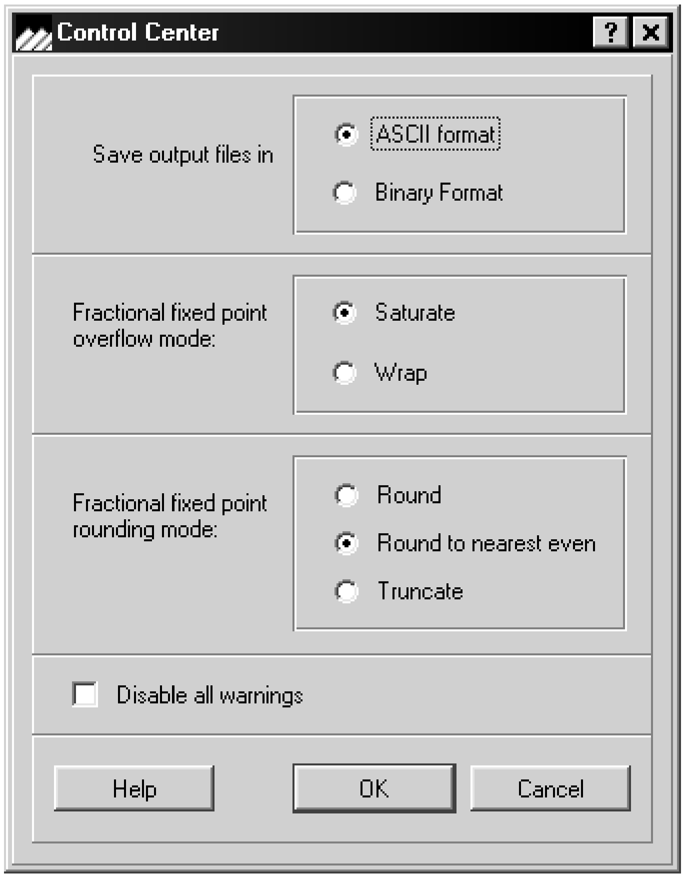
FIGURE 9.1 **Utilities Menu**



9.1 Control Center

This dialog box as displayed in Figure 9.2 allows the user to control the output file format, fractional fixed point arithmetic overflow, rounding and real-time display graphics.

FIGURE 9.2 **Control Center Panel**



Control Center

Save output files in: [✓ ASCII format] Binary Format

Fractional fixed point overflow mode: [✓ Saturate] Wrap

Fractional fixed point rounding mode: Round [✓ Round to nearest even] Truncate

☐ Disable all warnings

9.1.1 Save Output Files

dsPICworks software processes data stored in TIM or FRE files, in the process creating new TIM or FRE files where required or specified. These TIM or FRE files can be created as ASCII text or binary files by selecting the appropriate choice in this dialog box. The newly selected file format is used for all subsequent file generation activity.

The "Output File Format" selection in the ***GENERATOR*** dialog box, overrides this Control Center option for the specific instance when the Generator menu function is used. The "Save Output File" option in the Control Center is unrelated to the Import/Export dialogs in the ***FILE*** menu and should not be confused with them.

Typically ASCII file format provides readability while the Binary format enables dsPICworks software to process the file faster.

- **ASCII Output**

  If ASCII is selected, then all output files created from that time on will be in ASCII format. This means that all data values are written in ASCII and the data values can be displayed in any editor window. ASCII output requires conversion to binary for arithmetic operations and so it is slower than using binary for output files.

- **Binary Output**

  Binary files have all data written in binary format. Thus, no data conversion is required for arithmetic operations. However, the data values cannot be displayed. If it is necessary to display the data values, convert the file using ***UTILITIES/****BINARY TO ASCII*. Note that the setting of the ASCII or binary output does not affect the file headers which are always in ASCII format.

9.1.2 Fractional Fixed Point Overflow Mode

This selection only affects fractional fixed point operations. Possible selections are 'Saturate' or 'Wrap'. In saturate mode under the overflow condition the results are set to the largest possible values of the same sign. For example the sum of 0x8010 and 0xff20 is 0x8000.

In wraparound mode, the extra bits produced as a result of overflow are discarded and the lower 16-bits are saved as the results. For example, the sum of 0x8010 and 0xff20 is 0x7f30. Thus the sum of two negative numbers is a positive and the result has been wrapped to the opposite sign and hence the name.

- **Saturation**

  If the 'Saturate' option is selected, then any arithmetic operation for fixed point fractional arithmetic that results in a value equal to or greater than 1.0 is set to the highest positive value (7fff in hexadecimal). If an arithmetic operation results in a value that is less than -1.0, that value is set to -1.0 or 8000 in hexadecimal.

- **Wraparound**

  If wraparound is selected, then the arithmetic values are allowed to wraparound. Thus, it is possible for the addition of two positive numbers to result in a negative number. The default is set to 'Saturate'. Refer to Section 1.4.2 on page 6 for more detailed information on fixed point fractional numbers.

9.1.3 Fractional Fixed Point Rounding Mode

This selection only affects the fractional fixed point operations.

Possible selections are:

- **Round**
- **Round to the nearest even**
- **Truncate**

Fractional fixed point multiplications produce 31-bit products, i.e. it has 15 extra least significant bits than 16-bit fractional fixed point numbers. These extra bits must be disposed of before they can be stored as 16-bit results. This selection affects how the results are computed.

In order to facilitate explanation, the dimension is restricted to positive numbers. Let Δ be defined as the smallest difference between two 16-bit fractional fixed point numbers.

i.e. $\Delta = \frac{1}{2^{15}} \cong 0.000305175$ also let δ be defined the difference between the 31-bit product and the 16-bit fractional fixed point part of the product.

- **Round**

  If δ is less than $\Delta/2$, the extra least significant bits are discarded and the16 most significant bits are stored. If δ is equal to or greater than $\Delta/2$, the extra bits are discarded but Δ is added to the remaining 16-bits and stored.

- **Round to the Nearest Even**

  This is similar to simple rounding except the special handling of the cases where $\delta = \Delta/2$. In these cases the extra bits are discarded and Δ is added to or subtracted from the remaining 16-bits such that the results are a multiple of 2Δ.

- **Truncate**

  The extra bits are simply discarded without paying attention to the sign of the numbers.

9.2 Saturate /Wraparound

This sets the option for handling fixed point overflow for fixed point fractional numbers. The setting of this option has no effect of floating point data operations.

Please refer to more detailed write-ups on these sections on “Saturation” on page 119 and “Wraparound” on page 119.

This option allows the user to quickly modify the Control Center settings for Saturate and Overflow modes and serves as a status indicator of the current setting.

9.3 ASCII/Binary Output

These two settings are mutually exclusive.

This sets the option for generating either Binary or ASCII format files.

Please refer to more detailed write-ups on these sections on “Binary Output” on page 119 and “ASCII Output” on page 119.

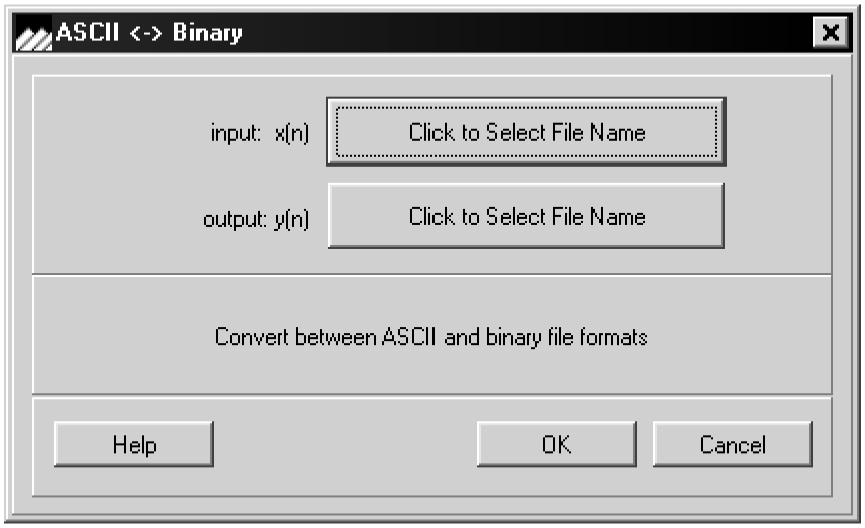
This option allows the user to quickly modify the *CONTROL CENTER* settings for Output file types serves as a status indicator of the current setting.

This option can also be set in the Control Center panel.

9.4 ASCII & Binary Conversion

This utility allows the conversion of a file with data values written in ASCII or binary to be converted the selected file format. The following dialog box as shown in Figure 9.3 obtains the input and output filenames.

FIGURE 9.3 **ASCII/Binary Conversion**


ASCII <-> Binary
input: x(n)
Click to Select File Name
output: y(n)
Click to Select File Name
Convert between ASCII and binary file formats
Help
OK
Cancel

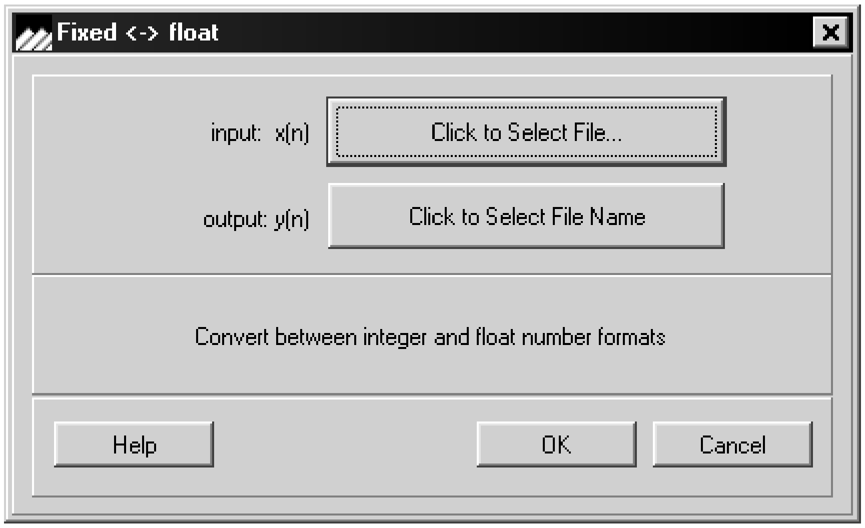
9.5 Fixed and Float Conversion

This utility converts numeric data types. If the output file is a fixed point fractional numeric data type, the output will be a floating point numeric data type. If the input numeric data is floating point, the output numeric data type will be fixed point fractional. If the value of a floating point number exceeds the range [-1, 1] of fixed point fractional, the output values will be saturated to $+(1\text{-}2^{15})$ or -1 as dictated by the Control Center setting.

The file format (ASCII or binary) of the output file will be set to the file format of the input file.

The input and output filenames are specified in the dialog box as shown in Figure 9.4.

FIGURE 9.4 **Integer/Floating Point Conversion**

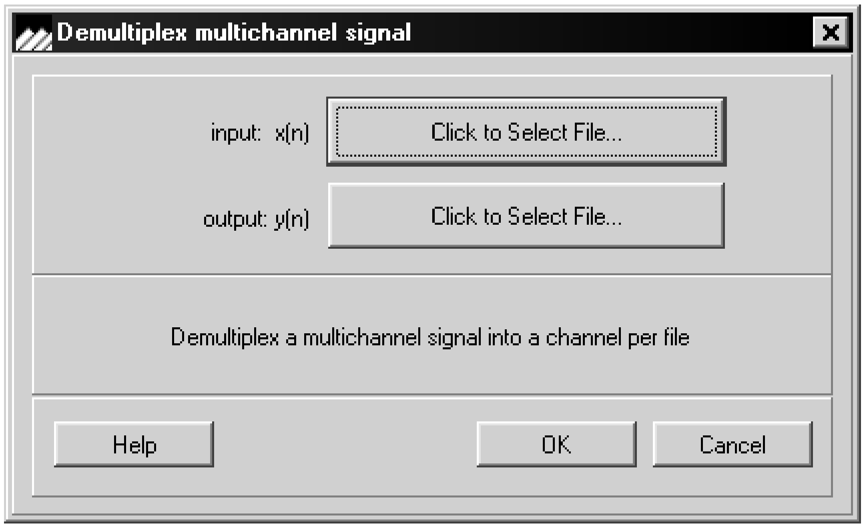


Convert between integer and float number formats

9.6 Demultiplex Signal

This utility separates a multi-channel recorded signal file into separate signal files, each containing signals for one channel. The output prefix should have six or less characters '-0' is appended to the prefix to make output file name for channel 0 (or left channel) signal and '-1' for channel 1 (or right channel) signal, etc. up to the 32-channel limit of the system. The dialog box used to obtain the filenames is displayed in Figure 9.5.

FIGURE 9.5 Demultiplex Multichannel File

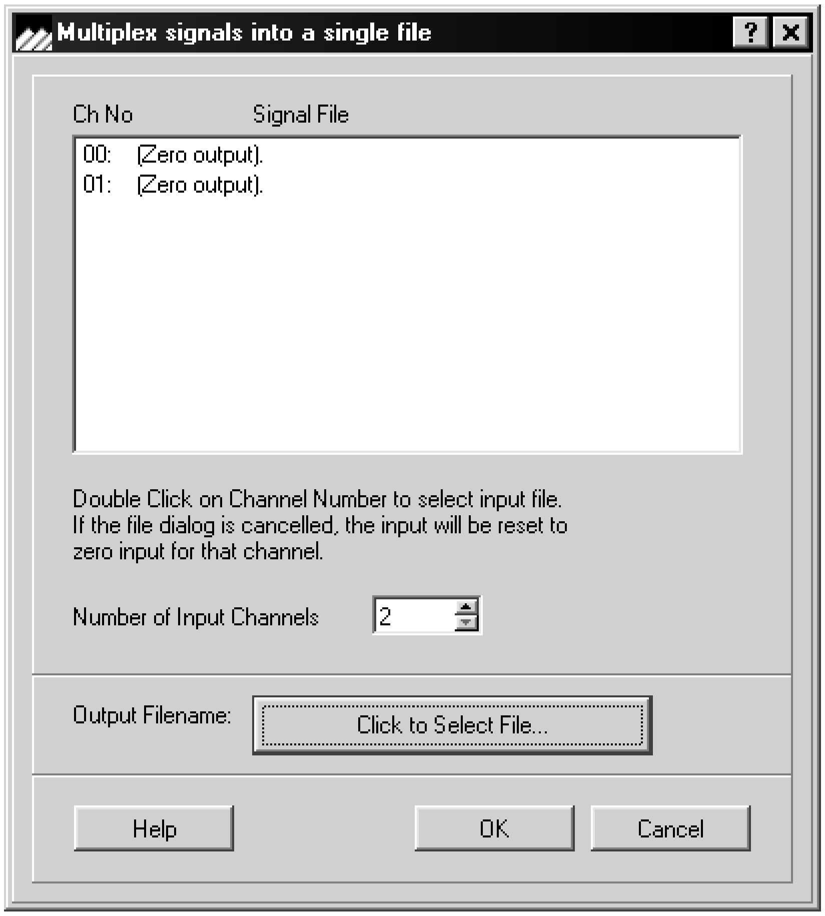


Demultiplex a multichannel signal into a channel per file

9.7 Multiplex Signals

This utility combines two or more signal files into a single file suitable for playback on a multi-channel board (from 2 - 32 channels). The input signal files may be in floating point or fractional fixed point format. Floating point signal files are automatically converted to fractional fixed point. If the files are of unequal length, the shorter files will be padded with zeros to the length of the longest file. If the sampling rates are different, the sampling rate of channel 0 signal file is used. An example of this dialog box is displayed in Figure 9.6.

FIGURE 9.6 **Create Multichannel File**



00: (Zero output).
01: (Zero output).

Double Click on Channel Number to select input file.
If the file dialog is cancelled, the input will be reset to zero input for that channel.

Number of Input Channels: 2

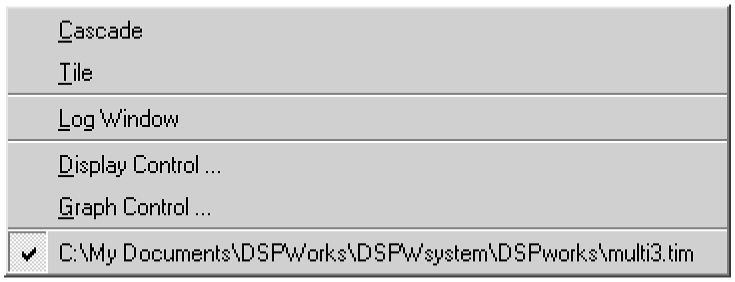
Output Filename: Click to Select File...

CHAPTER 10 *Window Menu*

10.1 Window Options

The ***WINDOW*** menu allows selection of the display windows and setting certain options for graph windows as shown in Figure 10.1.

FIGURE 10.1 Window Menu Display



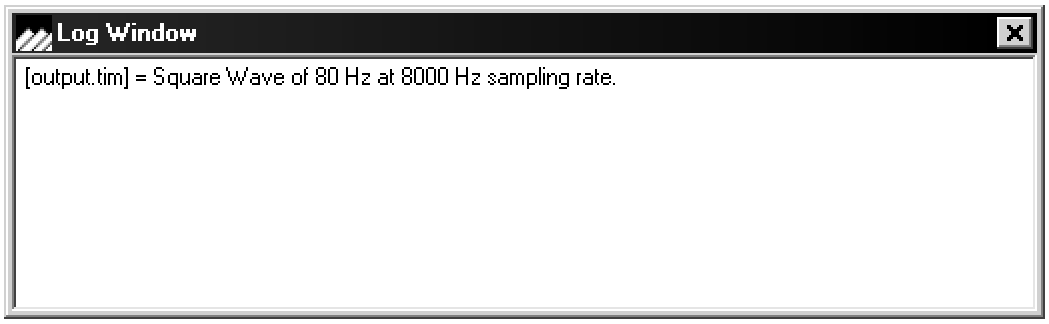
10.1.1 Tile/Cascade

Allows window displays to be tiled or cascaded.

10.1.2 Log Window

Various statistical and status messages are displayed in the log window as shown in Figure 10.2 if this option is selected.

FIGURE 10.2 Example of Log Window

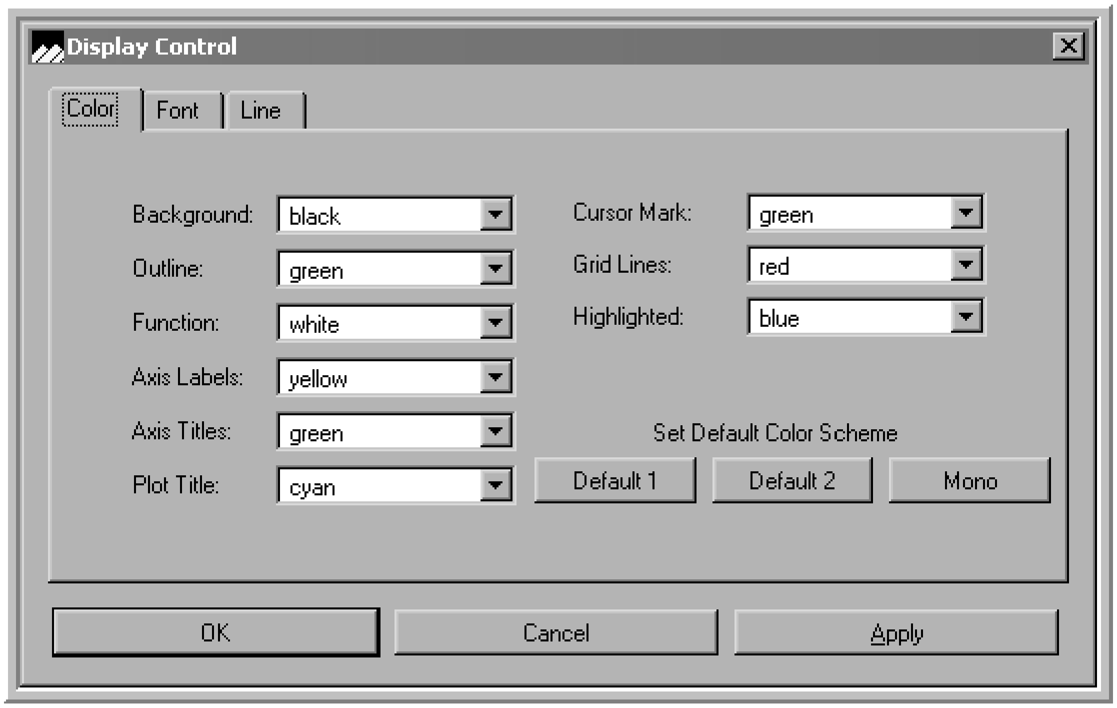


```
[output.tim] = Square Wave of 80 Hz at 8000 Hz sampling rate.
```

10.1.3 Display Control

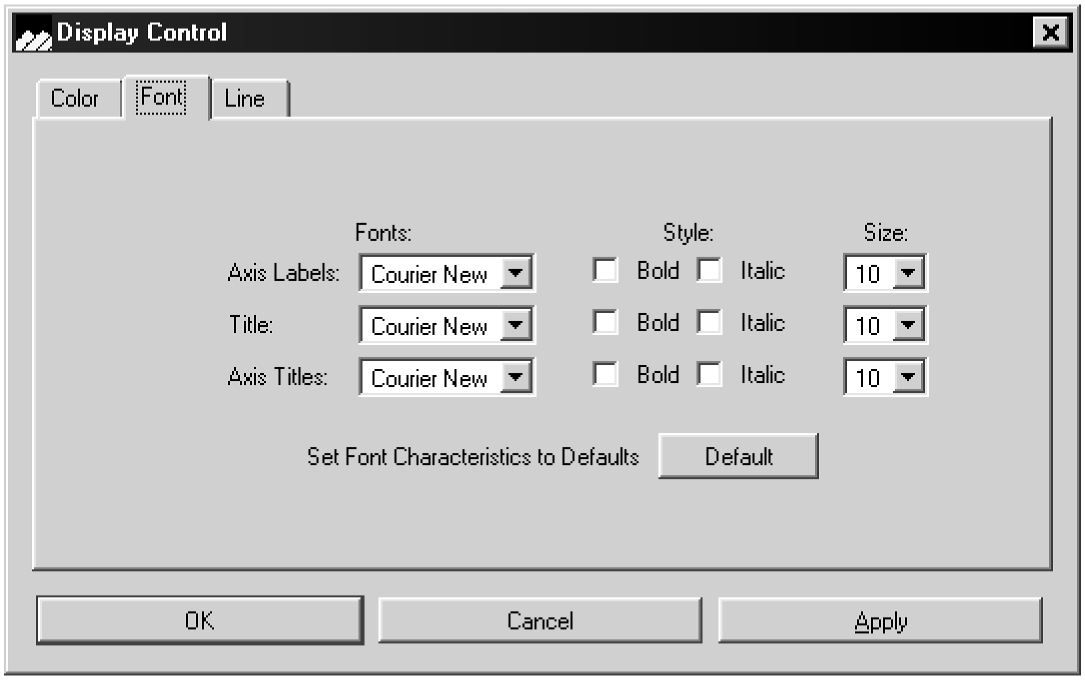
Selecting *DISPLAY CONTROL* displays dialog boxes as shown in Figure 10.3, Figure 10.4 and Figure 10.5 which allow the user to select the color, font and line options.

FIGURE 10.3 **Display Control -Color**



Background: black
Outline: green
Function: white
Axis Labels: yellow
Axis Titles: green
Plot Title: cyan
Cursor Mark: green
Grid Lines: red
Highlighted: blue

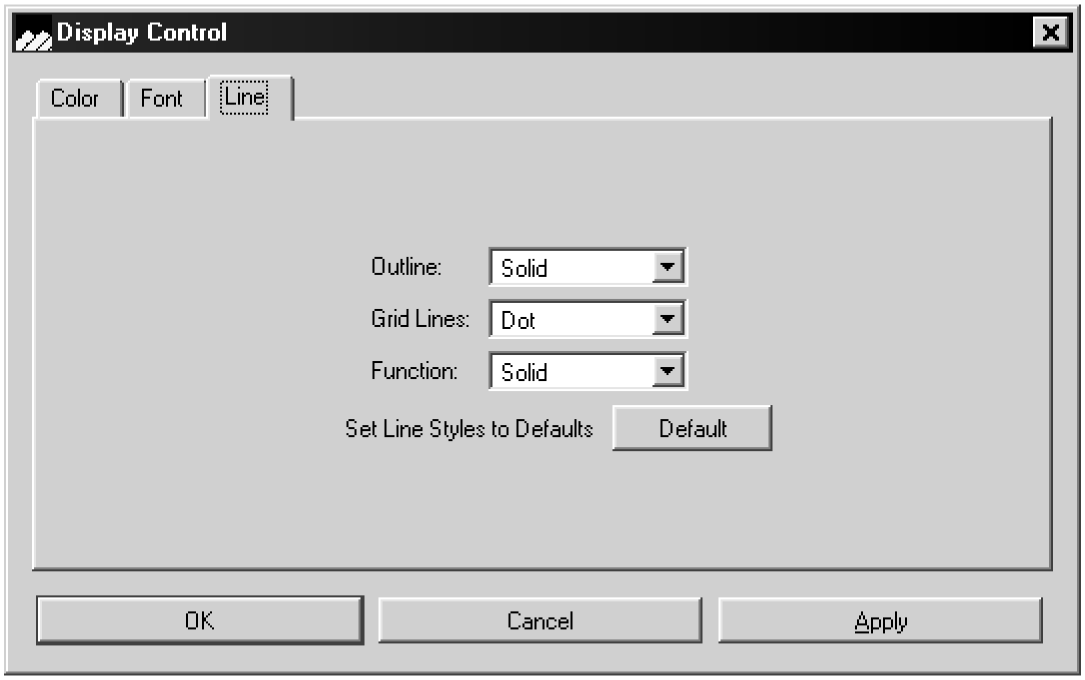
FIGURE 10.4 Display Control - Font



	Fonts:	Style:		Size:
Axis Labels:	Courier New	☐ Bold	☐ Italic	10
Title:	Courier New	☐ Bold	☐ Italic	10
Axis Titles:	Courier New	☐ Bold	☐ Italic	10

Set Font Characteristics to Defaults Default

FIGURE 10.5 Display Control - Line Selection



Outline: Solid

Grid Lines: Dot

Function: Solid

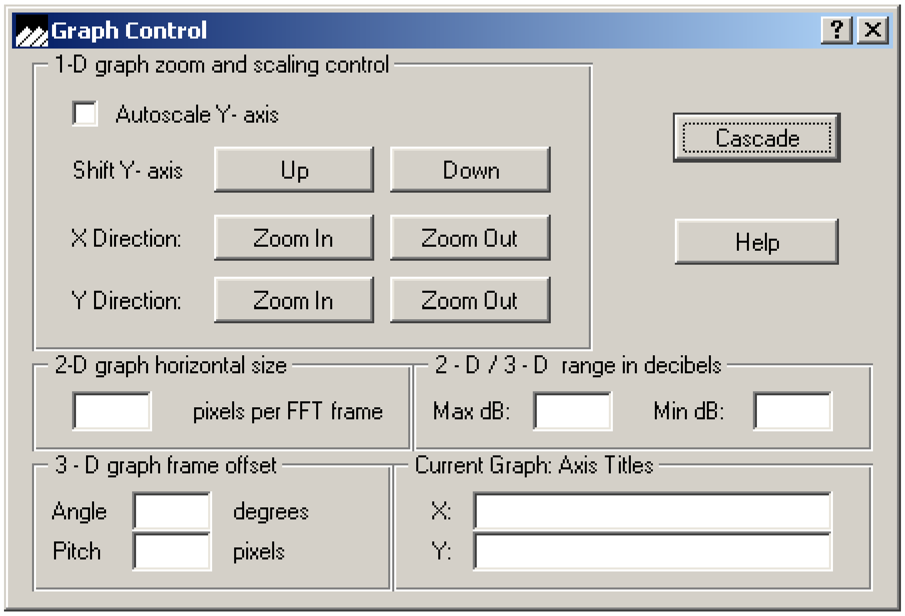
Set Line Styles to Defaults Default

This option allows the user to select various font, size, style and color options to customize graphical output to individual requirements.

10.1.4 Graph Control

The graph control dialog as shown in Figure 10.6 allows scaling of the x- and y-axes and/or y-axes titles as well as other characteristics of 2-D or 3-D frequency domain graph displays. Only the top half of the dialog box showing the more frequently used items will be visible at the bottom of the screen when the Graph Control feature is invoked. To see the other control items, drag the dialog box by the title bar away from the bottom of the screen. The Graph Control feature can also be activated by typing Control-C or using a toolbar shortcut. Note that when a selection is made by right-clicking on a frequency graph, the log window will pop up and display some signal power characteristics.

FIGURE 10.6 **Graph Control**



The various features are explained individually below.

TABLE 10-1 **Graph Control Functions**

Feature	Function
Next	Changes the current active window.
Tile	Same as “Window/Tile”
OK	Accept changes
X-in, X-out	Zoom in/Zoom out on the X-Axis. The maximum number of points displayed is 600 whereas the minimum number of points is limited to 10. While zooming in/out, the center of the graph remains fixed.
Y-in, Y-out	Zoom in/out on the Y-axis. This will also turn autoscale Y-axis feature off.
Shift Y Axis	Up/Down graph curve is out of Y-axis range - rescaling to make visible. When the scale has been adjusted such that the function is off the visible plotting area, the scale will automatically be adjusted so that the function is again visible and a message to this effect is displayed.

Feature	Function
Autoscale Y-axis	The autoscale option is a system default. When this feature is on, the Y-axis will automatically be adjusted so that the function will occupy approximately the entire graph. Y-in, Y-out will turn this feature off.
X:title Y:title	Enter these new graph titles for the current active window. The Y-title will display only if the current Y-title font is scalable
2-D Graph Horizontal Size	In the 2-D display the pixels are the width of the vertical stripe that represent a frame.
3-D graph pitch and angle	In the 3-D display, specify the relative position of the adjacent frames. 0^o angle causes the older frames to be shifted right horizontally and 90^o vertically up. The pitch is the approximate distance in pixels between adjacent frames.
2-D/3-D dB Range	The 2-D/3-D displays specify the maximum and minimum display range in dB.

CHAPTER 11 *Addendum - Lattice Filters*

11.1 Creating Lattice IIR Filters

The following equations are used to implement the lattice IIR filters:

$$\begin{aligned} f_N &= x(n) \\ f_{m-1}(n) &= f_m(n) - K_m g_{m-1}(n-1) \\ g_m(n) &= K_m f_{m-1}(n) + g_{m-1}(n-1) \end{aligned} \qquad \textbf{(EQ) A}$$

for m = N,...,1 where N is the filter order and K_m are the “Kappa” coefficients

$$y(n) = \sum_{m=0}^{N} v_m g_n(n) \qquad \textbf{(EQ) B}$$

where v_m are the “gamma” coefficients.

Note that N is the order of the filter and there are N+1 gamma coefficients and N kappa coefficients.

To implement a Lattice IIR Filter, it is recommended that the Lattice file IIR.FLT as shown in Example 12 on page 135 be used as a template. Simply change the filter order and number of sections as needed, and change the gamma and kappa coefficients.

Note that this FLT file can be automatically created by QEDesign2000™.

EXAMPLE 12 **Lattice IIR Filter Template**

```
FILTER COEFFICIENT FILE
IIR DESIGN
FILTER TYPE                 LOW PASS
ANALOG FILTER TYPE          ELLIPTIC
PASSBAND RIPPLE IN -dB        -.1000E+01
STOPBAND RIPPLE IN -dB        -.7000E+02
PASSBAND CUTOFF FREQUENCY   0.200000E+04 HERTZ
STOPBAND CUTOFF FREQUENCY   0.210000E+04 HERTZ
SAMPLING FREQUENCY          0.800000E+04 HERTZ
FILTER DESIGN METHOD: BILINEAR TRANSFORMATION
FILTER ORDER           10    Ah
NUMBER OF SECTIONS      5    5h
NO. OF QUANTIZED BITS  24   18h
QUANTIZATION TYPE - FRACTIONAL FIXED POINT
COEFFICIENTS SCALED FOR LATTICE STRUCTURE
 0.10000000E+01        /* overall gain                      */
 0.18687015E-03        /* Gamma Coeff G0 */
 0.66846691E-03        /* Gamma Coeff G1 */
 0.86480659E-02        /* Gamma Coeff G2 */
 -.12298214E-01        /* Gamma Coeff G3 */
 -.69204085E-01        /* Gamma Coeff G4 */
 0.19095197E-01        /* Gamma Coeff G5 */
 0.19068243E+00        /* Gamma Coeff G6 */
 0.25954828E+00        /* Gamma Coeff G7 */
 0.17273812E+00        /* Gamma Coeff G8 */
 0.62015012E-01        /* Gamma Coeff G9 */
 0.10546458E-01        /* Gamma Coeff G10 */
 -.17744888E-01        /* Kappa Coeff K1 */
 0.99816036E+00        /* Kappa Coeff K2 */
 -.11106321E+00        /* Kappa Coeff K3 */
 0.97254151E+00        /* Kappa Coeff K4 */
 -.37554330E+00        /* Kappa Coeff K5 */
 0.80778378E+00        /* Kappa Coeff K6 */
 -.61036742E+00        /* Kappa Coeff K7 */
 0.60916859E+00        /* Kappa Coeff K8 */
 -.45976728E+00        /* Kappa Coeff K9 */
 0.23930417E+00        /* Kappa Coeff K10 */
```

12.1 Creating FIR Lattice FIR Filters

The following equations are used to implement the lattice FIR filters:

$$f_0(n) = g_0 = x(n)$$
$$f_m(n) = f_{m-1}(n) + K_m g_{n-1}(n-1) \qquad \textbf{(EQ) C}$$
$$g_m(n) = K_m f_{n-1}(n) + g_{m-1}(n-1)$$

fm=1,...,N where N is the filter order of the FIR filter

$$y(n) = f_N(n) \qquad \textbf{(EQ) D}$$

To implement a Lattice FIR Filter, it is recommended that the Lattice file FIR.FLT as shown in Example 2 on page 136 be used as a template Then simply change the filter order and number of sections as needed, and change the gamma and kappa coefficients.

EXAMPLE 2 **Lattice FIR Filter Template**

```
FILTER COEFFICIENT FILE
FIR DESIGN          FLOATING POINT LATTICE
SAMPLING FREQUENCY          0.800000E+04 HERTZ
   3                  /* filter order in decimal     */
   3                  /* filter order in hexadecimal */
24                    /* number of bits in quantized coefficients (dec) */
  18                  /* number of bits in quantized coefficients (hex) */
 0.25 /* Kappa Coeff K1  */
 0.50 /* Kappa Coeff K2  */
 0.333333333333333333 /* Kappa Coeff K3  */
```

CHAPTER 12 *References*

1. Antoniou, Andreas. *Digital Filter Analysis and Design*: McGraw Hill, 1979
2. Bernhardt, Paul A. *Simplified Design of High Order Recursive Group Delay Filters*, IEEE Transactions on Acoustics, Speech, and Signal Processing. Vol. ASSP-29, No. 5, October 1980
3. Crochiere, R. E. & Rabiner L. R., *Multirate Digital Signal Processing*: Prentice Hall, 1983
4. Elliott Douglas F. *Handbook of Digital Signal Processing Engineering Applications*: Academic Press Inc, 1987
5. Harris F.J., *On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform*. Proc. IEEE, Vol 66, No. 1, pp. 51-83, Jan 1978
6. Hogenauer, E. B. *An Economical Class of Digital Filters for Decimation and Interpolation*; IEEE Transactions on Acoustics, Speech, and Signal Processing. Vol. ASSP-29, No. 2, April 1981
7. Jackson, L. B. *Digital Filters and Signal Processing*. Kluwer, 1986
8. Jayant, N. S. & Noll P. *Digital Coding of Waveforms*. Prentice Hall, 1984
9. Kaiser, J. F. *Nonrecursive Digital Filter Design using the IO - SinH Window Function*: Proceedings of IEEE International Symposium on Circuits and Systems, 1984
10. Nuttal A. H. *Some Windows with Very Good Sidelobe Behavio*r, IEEE Trans. Acoust., Speech and Signal Processing, vol ASSP-29, No. 1, pp 84-91, Feb. 1981
11. Oppenheim, Alan V. & Shafer, Ronald W. *Digital Signal Processing*: Prentice Hall, 1975
12. Parks, T.W. & Burrus C.S. *Digital Filter Design*. John Wiley & Sons, Inc., 1987
13. Proakis John G & Manolakis Dimitris G. *Digital Signal Processing Principles, Algorithms, and Applications*. 2nd Edition. Macmillan Publishing, New York, 1992
14. Rabiner, Lawrence R & Gold, Bernard. *Theory & Application of Digital Signal Processing*: Prentice Hall, 1975
15. Roberts, Richard A & Mullis, Clifford T. *Digital Signal Processing*: Addison-Wesley, 1987
16. Schuler, Charles and Chugani, Mahesh. *Digital Signal Processing: A Hands-On Approach:* McGraw-Hill, 2005
17. Webster R.J. *On Qualifying Windows for FIR Filter Design*, IEEE Trans. Acoust., Speech & Signal Processing, vol. ASSP-31, no. 1, pp 237-240, Feb. 1983