SpectraPLUS

FFT Spectral Analysis System

Version 5.0

Users Guide

Table of Contents

Chapter 1 - Installation

Installing SpectraPLUS

To install *SpectraPLUS,* insert the CD-ROM in the drive on your system
and the installation procedure will automatically start. If not then navigate
to the CD drive and run the "specplus_50_setup.exe" file.

The installation program will take over from there, and will ask you which
directory you wish to install the program to. In addition, the installation
program will create items on the Start Menu and desktop for you.

Starting SpectraPLUS

Once the installation is complete, you can start by double clicking on the
SpectraPLUS icon on the desketop or in the SpectraPLUS menu on the
start menu.

Chapter 2 - Introduction

What is a spectrum analyzer?

A spectrum analyzer is an instrument used to convert a signal from the time domain (amplitude-vs-time) to the frequency domain (amplitude-vs-frequency). If you are familiar with an ordinary oscilloscope you know what a time domain display looks like. A frequency domain display is known as a spectrum. Unless you are measuring a single tone, an oscilloscope provides little in the way of frequency information; however, a spectrum analyzer clearly reveals this information. An Audio Spectrum Analyzer, by definition, is limited to processing signals in the audio band. The specific frequency limit is determine by the capabilities of your sound card (see sampling rate for more details).

What Can it Do?

An Audio Spectrum Analyzer is very useful for measuring the fundamental frequency components which are contained in an audio signal. It can accurately measure the frequency of single or multiple tones and the frequency difference between them.

How does it work?

The program works in conjunction with the sound card on your computer. Plug the audio signal to be measured into the Line-In or Microphone jack on the back of the sound card. SpectraPLUS then uses the card to perform an "Analog-to-Digital" conversion on the audio signal. This digitized audio is then passed through a math algorithm known as a Fast Fourier Transform (FFT) which converts the signal from the time domain to the frequency domain. The CPU on your computer is used to perform this transformation.

System Requirements

Minimum system requirements:

- 150 Mhz Pentium CPU recommended minimum
- 128 MB RAM minimum
- 256 Color or greater VGA monitor (1024 x 768 or greater resolution)
- Hard Disk with 20MB free space. Additional space is required to save audio files.
- Windows compatible sound card
- Mouse or other pointing device

Chapter 3 - Making a Measurement

Basic Operation

The program has three distinct operating Modes and five separate Views.

Modes

- **Real-Time** mode takes the audio directly off of the sound card, processes it and displays the results. The raw digitized audio is not stored in memory and cannot be saved to disk. You can however, run for an indefinite period of time.
- **Recorder** mode allows you to save the raw digitized audio to your hard disk in the form of a .WAV file. You can also playback the audio through the speaker connected to your sound card.
- **Post-Processing** mode allows you to process audio data which was previously recorded and stored on disk in a .WAV file. This mode allows greater flexibility than the Recorder or Real-Time modes in the analysis process. In particular, this mode allows the use of overlap processing to effectively stretch the time resolution in both the Spectrogram and 3-D Surface plots.

Views

- The Time Series view shows the digitized audio waveform. It is similar to an oscilloscope display (amplitude versus time).
- The Spectrum view displays the amplitude of the signal versus frequency.
- The Phase view displays the phase of the signal versus frequency.
- The Spectrogram view displays the spectra over time. The amplitude is shown in color or grayscale.
- The 3-D Surface view displays a perspective of the spectra over time.

Setting up the analyzer

Setup the hardware:

Install the sound card and driver (refer to instructions included with your hardware) and connect the audio signal you wish to measure to the Line-In or Microphone jack on the rear edge of the sound card.

Frequency Range:

Prior to making a measurement, you need to determine what frequency range you are interested in, and what frequency resolution is required. Go to the Settings dialog box by selecting the <Settings> command under the

<Options> menu item. Set the Sampling Rate to at least twice the highest rate that you wish to measure. Now adjust the FFT size so that the spectral line resolution meets your objectives.   Click OK to close the dialog box.

Axis Scaling

Go to the Scaling dialog box by selecting the <Scaling> command under the <Options> menu item. Select the appropriate scaling options. The program defaults to Logarithmic amplitude scaling and Linear frequency scaling.

Open any of the views you wish to display

Each view can be opened and closed by toggling the corresponding menu item. A check mark will appear next to any open view. Views can be opened and closed even when the processor is running but in the case of the 3-D Surface and Spectrogram views, the contents are erased once closed.

Select the operating mode

Select the Recorder mode if you wish to save the audio data for later processing. Select Real-Time mode if you wish to run in a monitor mode without saving the data. The Real-Time mode will allow you to run continuously whereas the Recorder mode is limited by available hard disk space or memory (depending on the selected File Options) on the computer. Post Processing mode is used for processing from a previously recorded WAV file.

Start the analyzer

Click <Run> or <Rec> on the toolbar or use the associated accelerator key <alt> <R> to start the processor. The displays will then begin updating. Optionally, you can press the enter key when in the real-time mode and the analyzer will make a single measurement and then stop.

Stop the analyzer and make a measurement

Click <Stop> on the toolbar or use the associated accelerator keys <alt> <S> to stop the processor. Use the mouse to point to a feature of interest on one of the views and press the left mouse button. Measurement boxes will appear which contain the corresponding plot values. (See Cursor Measurements for more information).

Print the results

Selecting the <File> <Print> menu item will print the active view on the selected printer. You will also have an opportunity to enter custom annotation which is printed along the top center of the page.

Save the configuration

You may save the current analyzer configuration at any time by using the <Config> <Load Configuration> menu item. All current settings are saved as well as the size and position of each view and any open WAV file.

Cursor Measurements

Left Mouse Button:

Pressing the left mouse button anywhere within the plot region will display horizontal and vertical measurement boxes containing the values of the X and Y positions pointed to.

Control Key + Left Mouse button:

Holding down the Control Key and clicking the left mouse button anywhere within the plot region and dragging will display horizontal and vertical measurement boxes containing the differential values of the X and Y positions relative to the starting point. This is very useful for measuring the difference between two points on the plot.

Shift Key + Left Mouse button:

Holding down the Shift Key and clicking the left mouse button anywhere within the plot region will display a set of harmonic cursors. These cursors show the first 10 harmonic frequencies. This is very useful for identifying harmonically related components within a complex spectrum.

Control + Shift Key + Left Mouse button:

Holding down both the Control and the Shift Key and clicking the left mouse button anywhere within the plot region will display a set of "sideband cursors". These cursors show additional measurement lines on both the upper and lower sides of the start point. This is very useful for identifying modulation related components within a complex spectrum.

Right Mouse Button:

Pressing and holding the right mouse button within any view will activate a popup menu containing additional editing and processing options. The contents of this popup menu depends upon the specific view. See the Right Click Action Menu topic for more information.

Notes:

- It is recommended that you "Stop" the Analyzer while using these mouse actions. There are no corresponding keyboard commands.

- The Spectrum and Phase plots will "pull" the cursor to the trace along the amplitude axis. This can be disabled from the spectrum/phase display options dialog box.
- The 3-D Surface plot can be deceptive; you must measure along the base (X-axis) of the plot to determine the correct frequency. Aligning with the top of a peak will skew the measurement to the left because of the perspective orientation of the display.

Right Click Action Menus

A single click with the right mouse button within the plot area of any view will activate a menu containing a variety of additional actions. Although some of these actions are familiar clipboard operations, many provide powerful "cross display" functions for post processing and analysis.

Time Series View
- **Cut** - cuts selected time segment and places it on the clipboard in .WAV format.
- **Copy As Wave** - copies the selected time segment to the clipboard in .WAV format.
- **Copy As Bitmap** - copies the plot image to the clipboard in a standard bitmap format.
- **Copy As Text** - copies the selected time segment to the clipboard in a tab delimited text format.
- **Mute** - mutes the selected time segment.
- **Filter** - filters the selected time segment.
- **Play** - plays the selected time segment through current output device.
- **Autoscale Display** - Evaluates the data and sets the Plot Top/Range parameters to optimum values.
- **Compute and Display Average Spectrum** - computes and displays the average spectrum for the selected time segment using infinite averaging. The spectrum view is opened if necessary.
- **Compute and Display Spectrogram** - computes and displays the spectrogram for the selected time segment using the current FFT size and averaging settings. An overlap percentage is used so that the entire time segment will fill the spectrogram plot. The spectrogram view is opened if necessary.
- **Compute and Display 3-D Surface** - computes and displays the 3-D Surface view for the selected time segment using the current FFT size and averaging settings. An overlap percentage is used so

that the entire time segment will fill the 3-D Surface plot. The 3-D Surface view is opened if necessary.

- **Compute and Display RT-60** – Opens and Calculates the Reverberation Time for the selectet time segment. See the topic on RT-60 for details.
- **Perform DTMF Analysis** – Decodes DTMF key presses (touch tones) for the selected time segment.
- **Compute RMS Level** - Computes the RMS level for the selected time segment.
- **Compute Crest Factor** - Computes and displays the crest factor for data set.
- **Compute Envelope Using Hilbert Transform** - Computes the Envelope (magnitude) of the selected time segment using the Hilbert Transform. The results are displayed in the time series view. Click on the "zoom out full" button to return to the normal display mode.
- **Compute Schroeder Integration** - Computes a reverse integration (right to left) based on M. R. Schroeder's method. The results are displayed in the time series view. Click on the "zoom out full" button to return to the normal display mode.
- **View Data Values** - displays a table containing the current time series values. The contents of this table will update whenever the time series display changes.
- **Properties** – activates the display options dialog box.
- **Help** - activates the online help.

Spectrum View

- **Copy as Text**- copies the displayed spectrum values to the clipboard in tab delimited text format.
- **Write to Text File** – write the displayed spectrum values to an ASCII text file.
- **Copy As Bitmap** - copies the plot image to the clipboard in a standard bitmap format.
- **Clear Spectrum** - clears the current spectrum. This is useful if you wish to view/print overlays with just the main spectrum trace or overlays.
- **Clear Peak Hold** - clears the peak hold data. This is useful if you wish to view/print overlays without the peak hold trace.
- **Autoscale Display** - Evaluates the data and sets the Plot Top/Range parameters to optimum values.
- **Inverse FFT** - performs an Inverse FFT and displays the results in the time series view.
- **Compute Cepstrum** - performs an FFT on the current spectral data.

- **Bandwidth and Q Factor** - computes and displays the half power bandwidth and Quality Factor for the maximum peak in the display.
- **Set Marker N** - activates marker N at the frequency of the mouse click point.
- **Clear Marker** - clears individual markers.
- **Marker Options** – activates the Marker Options Dialog box
- **Smooth Spectrum N** averages N adjacent bins to smooth the spectral data (requires narrowband freq scaling and logarithmic amplitude scaling).
- **View Data Values** - displays a table containing the current spectrum values. The contents of this table will update whenever the display changes.
- **Properties** – activates the display options dialog box.
- **Help** - activates the online help.

Phase View

- **Copy as Text** - copies the displayed phase values to the clipboard in tab delimited text format.
- **Copy As Bitmap** - copies the plot image to the clipboard in a standard bitmap format.
- **View Data Values** - displays a table containing the current phase values. The contents of this table will update whenever the display changes.
- **Properties** – activates the display options dialog box.
- **Help** - activates the online help.

3-D Surface View

- **Cut** - cuts selected time segment and places it on the clipboard in .WAV format.
- **Copy As Wave** - copies the selected time segment to the clipboard in .WAV format.
- **Copy As Bitmap** - copies the plot image to the clipboard in a standard bitmap format.
- **Mute** - mutes the selected time segment.
- **Filter** - filters the selected time segment.
- **Play** - plays the selected time segment through current output device.
- **Autoscale 3-D Surface** – Evaluates the data and sets the Plot Top/Range parameters to optimum values.
- **Display Time Series** - displays the time series corresponding to the selected segment. The time series view is opened if necessary.

- **Compute and Display Average Spectrum** - computes and displays the average spectrum for the selected time segment using infinite averaging. The spectrum view is opened if necessary.
- **Compute and Display Spectrogram** - computes and displays the spectrogram for the selected time segment using the current FFT size and averaging settings. An overlap percentage is used so that the entire time segment will fill the spectrogram plot. The spectrogram view is opened if necessary
- **Expand 3-D Surface** - expands the 3-D Surface view by recomputing the data for the selected time segment. An overlap percentage is used so that the entire time segment will fill the 3-D Surface plot..
- **Perform DTMF Analysis** – Decodes DTMF key presses (touch tones) for the selected time segment.
- **Properties** – activates the display options dialog box.
- **Help** - activates the online help.

Spectrogram View

- **Cut** - cuts selected time segment and places it on the clipboard in .WAV format.
- **Copy As Wave** - copies the selected time segment to the clipboard in .WAV format.
- **Copy As Bitmap** - copies the plot image to the clipboard in a standard bitmap format.
- **Mute** - mutes the selected time segment.
- **Filter** - filters the selected time segment.
- **Play** - plays the selected time segment through current output device.
- **Autoscale Spectrogram** – Evaluates the data and sets the Plot Top/Range parameters to optimum values.
- **Display Time Series** - displays the time series corresponding to the selected segment. The time series view is opened if necessary.
- **Compute and Display Average Spectrum** - computes and displays the average spectrum for the selected time segment using infinite averaging. The spectrum view is opened if necessary.
- **Expand Spectrogram** - expands the spectrogram by recomputing the data for the selected time segment. An overlap percentage is used so that the entire time segment will fill the spectrogram plot.
- **Compute and Display 3-D Surface** - computes and displays the 3-D Surface view for the selected time segment using the current FFT size and averaging settings. An overlap percentage is used so that the entire time segment will fill the 3-D Surface plot. The 3-D Surface view is opened if necessary.

- **Perform DTMF Analysis** – Decodes DTMF key presses (touch tones) for the selected time segment.
- **Properties** – activates the display options dialog box.
- **Help** - activates the online help.

Printing the results

The program allows you to print one View at a time.   Make the view you wish to print the active window by clicking anywhere in the plot. Size the view to display the information you wish to print; only the displayed data will be printed. Select the <Print> option from the <File> menu. Change any printer options and click OK.

The <File><Print Setup> menu command allows you to select various options specific to your printer.

Annotation: Two user defined lines of annotation are provided. These are centered along the top portion of the printout.

Margins: Set as desired.

Comments: You can include up to 10 comment lines which will be printed below the plot.

Notes:
- For best results when printing the color spectrogram on a monochrome printer, select the grayscale option on the spectrogram options dialog box.
- For printing the 3-D surface plot on a monochrome printer, you may wish to select a white background color and black line color from the 3-D Surface options dialog box.
- Color printers are supported.
- Pen Plotters are not supported for the spectrogram and 3-D surface plots

Chapter 4 - File Management

Working with Audio Files

Standard File Open/Save/SaveAs Dialog boxes are used to allow you to locate files on your system. Only the .WAV format is currently supported, however you may import raw data files from other acquisition systems using the File Import command.   The program also supports "Drag and Drop" from the Windows File Manager (refer to Winfile.hlp if you are unfamiliar with this feature).

Once a file is opened, or recording has started, the sampling rate and format cannot be changed. You must first close the file. This is because a .WAV file format requires a constant sampling rate and format throughout the entire file.

When the mode is changed to Real-Time, the current file will be closed and you will be prompted to save it if necessary.

The amount of memory required for each minute of audio varies widely depending upon the sampling rate and sampling format. Some typical values for mono recording are shown below (multiply the values by 2 for stereo recording):

Sampling Rate (Hz)	Sampling Precision (bits)	Bytes/Minute
11,025	8	661,500
11,025	16	1,323,000
22,050	8	1,323,000
22,050	16	2,646,000
44,100	8	2,646,000
44,100	16	5,292,000
44,100	24	7,938,000

Continuous Recording Capability

Wave files can store up to 2 GB of data in a single file (the corresponding time depends upon the sampling format). When this limit is reached, the program will momentarily stop recording, save the file and then start recording again in a new file. If the file is does not yet have a name, the program will give it a name using the computer date and time. The file will be stored in the \wave folder as specified in the <File><Set Paths> dialog. This feature applies to the hard disk recording mode only.

Example: c:\spectraplus\wave\recording_2002_11_26_102011.wav

(in this example the date and time were 10:20:11 Nov, 26 2002 when the file was saved)

Importing Files

Standard .WAV files are binary files containing a header followed by a block of data samples. The .WAV file header contains information about the sampling rate, number of channels, and number of bits per sample. Stereo files are interleaved (L,R,L,R...).
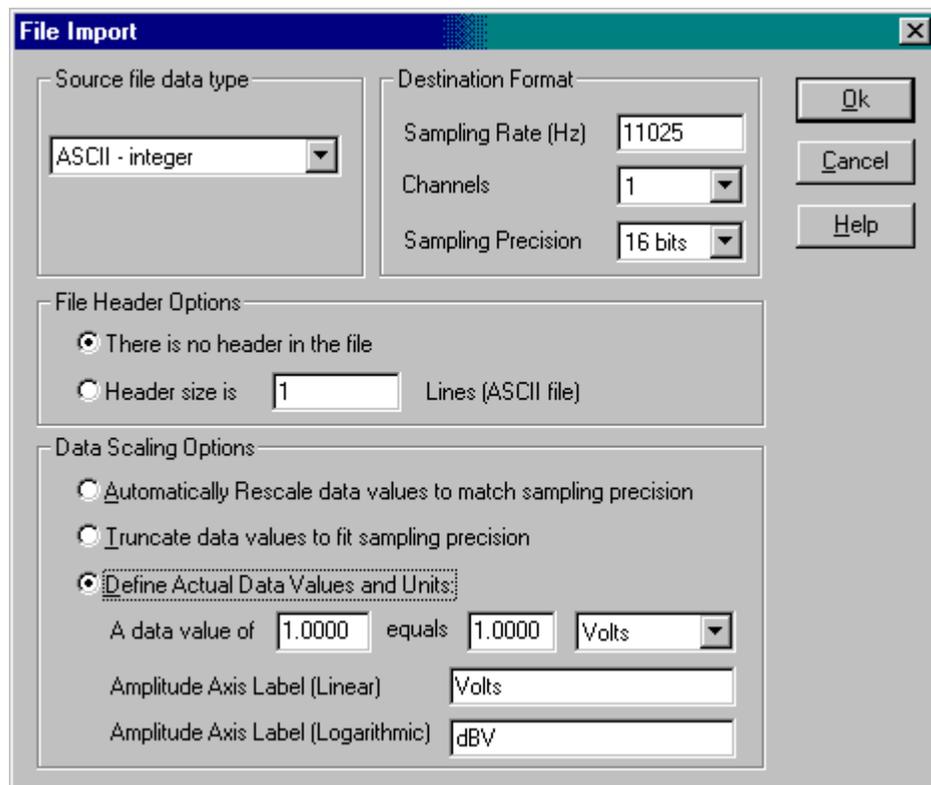
The import feature allows you to import raw digitized data from an external source and create a .WAV file from the contents. In order to properly build the WAV file header, you must specify the sampling rate represented by the data and the destination sampling format.

File Formats

- ASCII Integer - File is readable with any text editor. For mono files, each line contains a single sample value. For stereo files, each line contains both the left and right sample values separated by a comma.

- ASCII Float - File is readable with any text editor but the values are in floating point format (e.g. 21.034). For mono files, each line contains a single sample value. For stereo files, each line contains both the left and right sample values separated by a comma.
- Binary 8 bit - 8 bit integer values.
- Binary 16 bit - 16 bit integer values.
- Binary 32 bit - 32 bit integer values.
- Binary floating point - 32 bit floating point values.

Expected Sample Values
- 8 bit WAV data: 0...255
- 16 bit WAV data: -32767...+32767
- 24 bit WAV data: -8388607...+8388607

File Header Options
- If the data file contains a header, you can specify the size of this header in lines (ASCII file) or bytes (binary file). This header data will be skipped during the import process.

Data Scaling
- If "rescale" is checked, the program will determine the maximum and minimum values contained in the file and rescale all values to optimally fit within the allowable range of sample values.
- If "truncate" is checked, the program will accept all values as is and truncate those which exceed the allowable range of sample values.
- Define Actual Data Values and Units - use this option if you know the units contained in the data file. This will create the calibration parameters for properly scaling the data file.

Exporting Files

This feature allows you to export a .WAV file to a variety of file formats. These files can then be used with a spreadsheet, mathematical processing program, or your custom application.
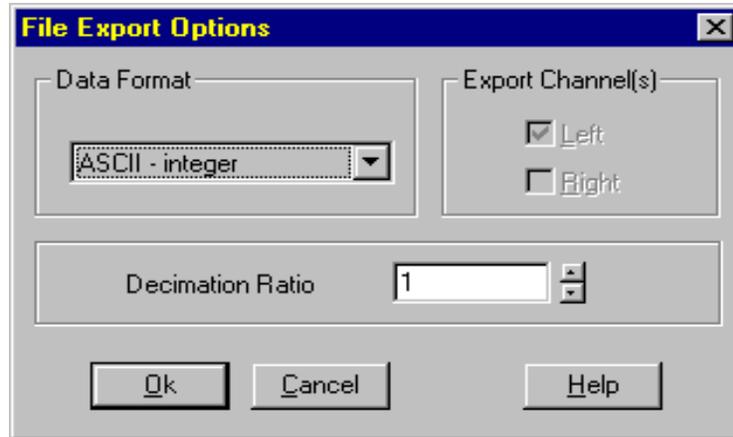
Decimation - this allows you to "down-sample" the wave data prior to exporting.

Destination File Formats:
- ASCII Integer - File is readable with any text editor. For mono files, each line contains a single sample value. For stereo files, each line contains both the left and right sample values separated by a comma.
- ASCII Float - File is readable with any text editor but the values are converted to floating point format (e.g. 21.034). For mono files, each line contains a single sample value. For stereo files, each line contains both the left and right sample values separated by a comma.
- Binary 8 bit - 8 bit integer values.
- Binary 16 bit - 16 bit integer values.
- Binary 32 bit - 32 bit integer values.
- Binary floating point - 32 bit floating point values.

Output Sample Values:
- 8 bit WAV data: 0...255
- 16 bit WAV data: -32767...+32767
- 24 bit WAV data: -8388607...+8388607

Notes:
- When exporting stereo files, the channels are interleaved starting with the left channel.
- Binary files are much more efficient than text files; ASCII text format is not recommended for large files.

- If you wish to export a small time segment from within a large .WAV file, use the time series view to specify the selected segment and then use the <Edit> <Copy> menu item to place the segment on the clipboard. Next, close the file and create a new file by pasting in the selected time segment using the <Edit> <Paste Insert> menu command. Then export this smaller WAV file.
- Exporting a 16 bit WAV file to either a ASCII or Binary 8 bit file will rescale the data to fit within the 8 bit sample range.

File Options

File options allow you to control how .WAV files are stored and accessed.
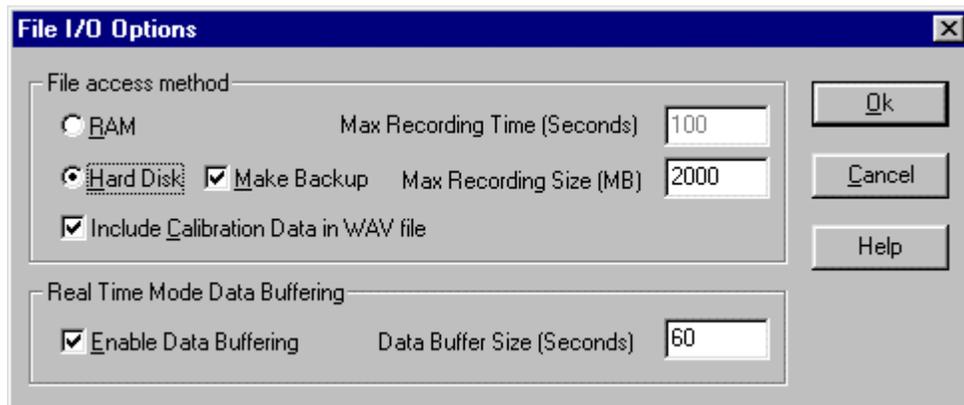
Access modes:
- **RAM** - All Wave data must fit in the available RAM. You can also specify the recording length
- **Hard Disk** - Wave data is recorded and played back directly from your hard disk. This allows much greater recording times. You can specify the maximum recording size in megabytes (wave files can store up to 2 GB of data in a single file - the corresponding time depends upon the sampling format).

Make Backup:
This option applies to hard disk access mode only. When this option is checked, a temporary backup file will be created so that original file contents can be restored if necessary. Use the <File><Set Paths> menu command to specify the directory to be used for these temporary files. All backup files are removed when the application closes.

Notes:

- When working with small files, RAM access will provide faster performance. Hard Disk access is preferred for larger files.
- During hard disk recording sessions, do not perform other disk intensive operations (such as opening a major application). Although Windows is a multitasking operating system, it relies on the cooperation of individual applications and cannot guarantee immediate access to system resources.

Include Calibration Data in WAV file:

If this option is enabled, the calibration parameters will be included in the WAV file and reloaded when the file is opened again. This is enabled by default. The calibration parameters will not add significantly to the file size. Most applications that work with WAV files wil not have a problem with this additional data but many will not preserve the cal data when saving the file. The only reason you should need to disable this option is if you encounter an application that has problems reading WAV files with the calibration data.

Real Time Mode Data Buffering

This option allows you to control whether data is buffered in the real time mode or not. Data buffering will allow the spectrogram or 3-D surface views to be redrawn to correspond to changes in scaling parameters or plot size and allows you to use the right click action menus to perform additional processing. In addition, you can convert this data buffer to a WAV file for further processing in the post-processing or recorder modes. This option is enabled by default and should prove useful in most situations. Turning this option off will result in a slight performance gain but this gain is minimal with adequate memory and hard disk space. The data buffering will use the same file access method (RAM or Hard Disk) as selected above.. The default buffer length is 60 seconds but can be changed as necessary.

Continuous Recording Capability
When Hard Disk recording is used and the maximum file size limit is reached, the program will momentarily stop recording, save the file and then start recording again in a new file. If the file is does not yet have a name, the program will give it a name using the computer date and time. The file will be stored in the \wave folder as specified in the <File><Set Paths> dialog. This feature applies to the hard disk recording mode only.

Example: c:\specplus\wave\recording_2002_11_26_102011.wav
(in this example the date and time were 10:20:11 Nov, 26 2002 when the file was saved)

File Decimate
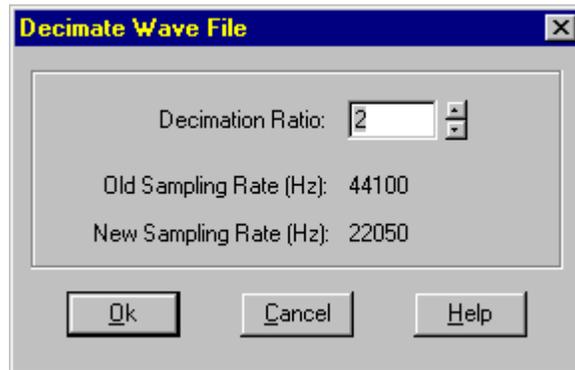
This feature allows you to reduce the sampling rate (decimate) of the current .WAV file by a user specified ratio. Enter the decimation ratio you wish to use; the new sampling rate will be displayed. You will then be prompted for the .WAV filename in which to save the results.

Note:

You may wish to apply a low-pass filter to the data prior to decimating.

File Merge

This feature allows you to combine two single channel .WAV files (mono) into a dual channel (stereo) .WAV file. This feature is useful for performing dual channel processing operations on tests made at different times.

You must close any open .WAV files and be in either the Recorder or Post Processing mode for the Merge command to be available. You will be prompted for the files to be used for the left and right channels. The resulting stereo .WAV file will be created and opened for you.

The source files for each mono file must have identical sampling rates and precision.

Chapter 5 - Editing

Cut/Copy/Paste

The <Edit> <Cut>/<Copy>/<Paste> menu commands are used to place the contents of the active view onto the clipboard. The format of the resulting clipboard data type differs depending on the active view:

View	Cut	Copy	Paste
Spectrum	N/A	ASCII table	N/A
Phase	N/A	ASCII table	N/A
Time Series	.WAV format	.WAV format	.WAV format
Spectrogram	.WAV format	.WAV format	.WAV format
3-D Surface	.WAV format	.WAV format	.WAV format

When using the Time Series, Spectrogram, or 3-D Surface views, you must first select the time segment which you want to edit. To do this, simply click the selection arrow tool on the left edge of the view toolbar. Next, click and drag across the plot area with the left mouse button to highlight the time segment you wish to edit. At this point the Edit commands will be enabled to allow you to select the desired operation. These commands are always disabled when in the real-time mode since no underlying data exists.

The .WAV format is compatible with any application which supports .WAV files.

The ASCII table format produces two columns of numbers with the bin frequency in the left column and the amplitude or phase in the right column. The columns are separated by a tab. You can then paste this data into a spreadsheet or text editor for further analysis. It is also compatible with both the spectral overlay and microphone compensation file formats.

The <Undo> <Edit> command allows you to reverse the most recent editing action. This includes the: <Edit> <Cut, <Edit <Paste Over>, <Edit <Paste Insert> and <Edit Filter> commands.

Play/Play Special

The <Edit> <Play> and <Play Special> menu commands allow you to playback a selected time segment so you can hear it through your speakers.

You must first select the time segment which you want to edit. This can be done from either the Time Series, Spectrogram, or 3-D Surface views. First, click the selection arrow tool on the left edge of the view toolbar. Next, click and drag across the plot area with the left mouse button to highlight the time segment you wish to edit. At this point the Edit commands will be enabled and allow you to select the desired operation.

The <Edit> <Play> menu item plays the selected segment through the speaker port on your sound card; it performs the same action as clicking on the tool containing the small speaker icon on the view toolbar.

The <Edit> <Play Special> menu has three choices:

- **Loop** - continuously plays the selected time segment (click the Stop button to cancel).
- **Half speed** - plays the selected time segment at 1/2 the normal sampling rate.
- **Double speed** - plays the selected time segment at 2 times the normal sampling rate.
- **Arbitrary speed** – plays the selected time segment at the user specified sampling rate.

Filtering

To filter a .WAV file, you must first select a time segment. This is done from either the Time Series, Spectrogram, or 3-D Surface view. Click the selection arrow located on the view toolbar and then click and drag the mouse across the desired time segment. This segment will then be displayed in inverse video. Once the time segment has been selected, press the filter button on the toolbar or select the <Edit><Filter> menu command. The filtering parameter dialog box will then allow you to select the frequencies to be filtered.
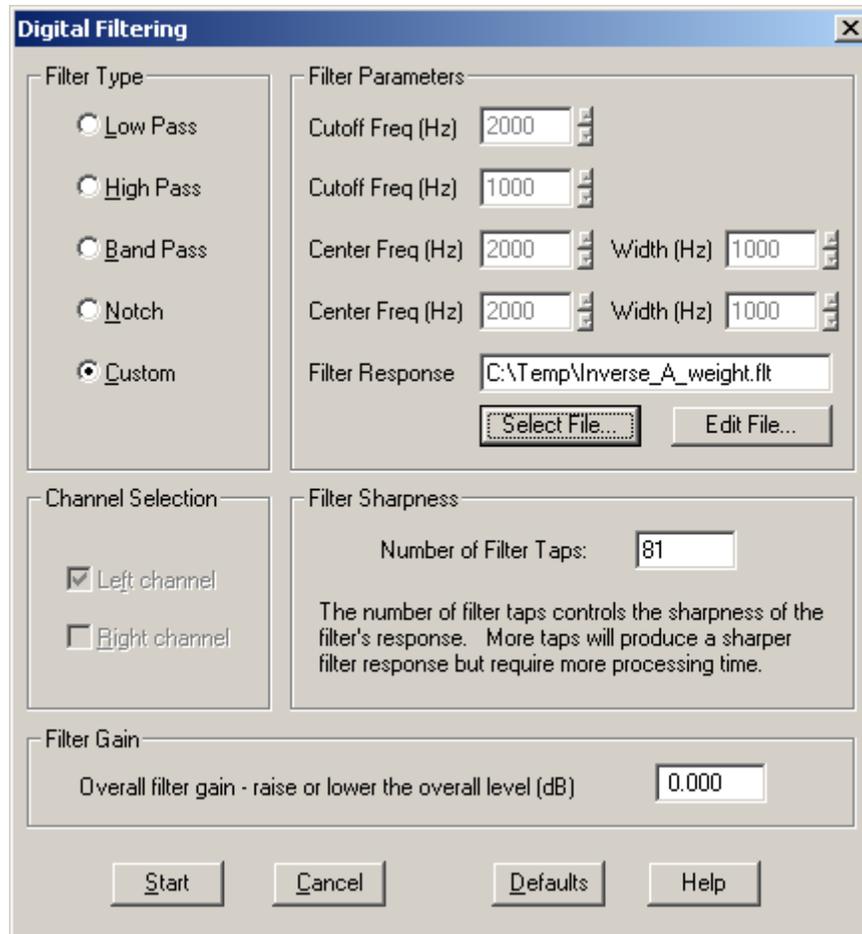
There are five filter types:

- **Low Pass** - passes all frequency components below the specified cutoff frequency and attenuates all components above it.
- **High Pass** - passes all frequency components above the specified cutoff frequency and attenuates all components below it.
- **Band Pass** - passes the specified band of frequencies and *attenuates all others.*
- **Notch** - rejects the specified band of frequencies and passes all others.
- **Custom** - applies the specified filter response file to the data.

Filter Sharpness - the number of filter taps (coefficients) directly affects the sharpness of the filter response and how closely it matches the ideal response. More taps produce a sharper filter response but require more processing time.

Filter Gain - allows the overall gain to be adjusted. Enter a value in dB; use a positive value to increase the gain, negative to decrease.

Building a filter response file:

Run the NOTEPAD.EXE application, spreadsheet or word processor that can produce a standard ASCII text file.
Enter two columns of numbers representing the filter response curve.
The left column contains the frequency in Hertz.
The right column contains the offset in decibels
Enter the information in ascending frequency order.
Separate the columns with a tab character.

Example:
```
---------------------------
100.0      -10.0
500.0       -5.0
1000.0       0.0
2000.0       2.0
10000.0     -5.0
15000.0    -15.0
```

Enter as many as32,768 pairs of numbers; at least 3 pair are required. The analyzer will use a cubic spline algorithm to interpolate between these data points for each frequency in the spectrum. Any line with a semicolon in the first column will be treated as a comment line and ignored. By convention, filter response files have an extension of .FLT.

Note:

Filter Response files, Microphone Compensation files, and Overlay files have identical formats and can be interchanged. This is useful if you want to verify that you have entered the filter response table correctly. To do this, simply select load the filter response file into one of the overlays for display.

The default folders used for storing the various files used by SpectraPLUS can be customized by clicking on the <File><Set Paths> menu item.

Mute

Muting a time segment will set the values to zero.

To mute a segment of a .WAV file, you must first select a time segment. This is done from either the Time Series, Spectrogram, or 3-D Surface view. Click the selection arrow located on the view toolbar and then click and drag the mouse across the desired time segment. This segment will

then be displayed in inverse video. Once the time segment has been selected, select the <Edit><Mute> menu command. The samples corresponding to the selected time segment will be set to zero.

Notes:

The <Edit><Undo Edit> menu command will restore the data to its original values.

Gain Adjust
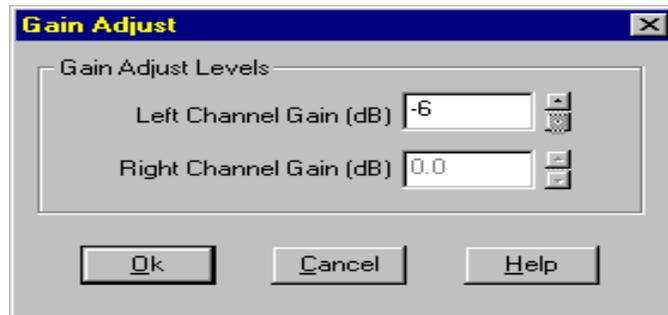
To adjust the gain of a WAV file, you must first select a time segment. This is done from either the Time Series, Spectrogram, or 3-D Surface view. Click the selection arrow located on the view toolbar and then click and drag the mouse across the desired time segment. This segment will then be displayed in inverse video. Once the time segment has been selected, select the <Edit><Gain Adjust> menu command. You will then be prompted enter the amount of gain in decibels (dB) to add or subtract. Click the OK button to perform the adjustment.

Notes:

The <Edit><Undo Edit> menu command will restore the gain to its previous value.

DC Offset

To adjust the DC Offset of a WAV file, you must first select a time segment. This is done from either the Time Series, Spectrogram, or 3-D Surface view. Click the selection arrow located on the view toolbar and then click and drag the mouse across the desired time segment. This segment will then be displayed in inverse video. Once the time segment has been selected, select the <Edit><DC Offset> menu command. You will then be prompted enter the amount of DC Offset add or subtract. The offset is entered in units according to the current calibration settings. Click the OK button to perform the adjustment.

Notes:

The <Edit><Undo Edit> menu command will restore the DC Offset to its previous value.

Select All

The Select All menu command will cause the contents of the time series, spectrogram and 3-D Surface views to be selected (highlighted) for additional editing.

Notes:

 If you wish to select the entire .WAV file for editing, first use the Zoom Out Full button on the time series toolbar. You may also need to increase the maximum zoom limit in the time series options dialog.

Chapter 6 - Modes

Real-Time

In the real-time mode, the program acquires an FFT sized block of digitized sound data directly from the sound card, computes the spectrum and displays the results. The program will continuously acquire new data, average it with previous data, and display the results until stopped.   The toolbar contains controls to allow you to modify the averaging block size and peak hold while running.

By default, the last 60 seconds of digitized audio data is stored in a temporary buffer. Switching to the Recording or Post Processing modes will allow you to convert this buffer into a WAV file for further processing.

Single step operation:

Pressing the <Enter> key will perform a single FFT measurement and then automatically stop the analyzer. The current average is not reset during single step operations.

Notes:

Depending upon the speed of your CPU, the selected sampling rate and FFT size, the computer may not be able to perform an FFT on the current data block and display the results before the next block is available. In this case, there will be gaps in the processed data. To perform gap free analysis, simply record the audio segment and analyze the data using the post-processing mode. Often it is useful to run the analyzer in the Real Time mode while making preliminary measurements and then switching over to the Recorder mode before making the final measurements.

Post-Processing

The post-processing mode allows you to analyze a previously recorded WAV file. The priority is given to spectral updates and the data is not played back through the speakers. For playback, use the recorder mode, or the Play/Play special commands.

This mode provides greater control over the processing than the Recorder or Real-Time modes.   There will be no gaps in the processed data, and you can "stretch" the time axis on the Spectrogram and 3-D Surface Plots by performing overlap processing.   In addition you can edit and playback selected time segments which allows in-depth analysis not possible in the real-time mode.

When a .WAV file is opened, the sampling rate and format are changed to match the rate at which the file was recorded - the Sampling Rate must remain constant within a single .WAV file. When the .WAV file is closed, these settings will revert to their previous values. The current sampling rate is always displayed in the status bar along the bottom of the application.

Single step operation:

Pressing the <Enter> key will perform a single FFT measurement and then automatically stop the analyzer (single step measurement). The current average is not reset during single step operations.

Recorder

This mode is similar to the Windows "Sound Recorder" utility in the Accessories group in that it allows you to record and playback sound files. Unlike "Sound Recorder" however, you have control over the sampling format and sampling rate. The program will also display the spectrum of the signal while recording or playing but priority is given to the recording process.

The maximum possible recording time is limited by the amount of available hard disk space or memory (depending upon the selected File Options).

When using the RAM file access mode, the program will attempt to allocate enough memory for the recording and will reduce the recording length if not enough memory is available. The recording length is specified in the File Options dialog box.
.
When using the Hard Disk file access mode, the program will record and playback directly from your hard disk. Temporary files are created during hard disk recording. Use the <File><Set Paths> menu command to specify the directory to be used for these temporary files. All temporary files are removed when the application closes.

The amount of memory required for each minute of audio varies widely depending upon the sampling rate and sampling format. Some typical values for mono recording are shown below (multiply the values by 2 for stereo recording):

Sampling Rate (Hz)	Sampling Precision (bits)	Bytes/Minute
11,025	8	661,500
11,025	16	1,323,000
22,050	8	1,323,000

22,050	16	2,646,000
44,100	8	2,646,000
44,100	16	5,292,000
44,100	24	7,938,000

When a .WAV file is opened, the sampling rate and format are changed to match the rate at which the file was recorded - the Sampling Rate must remain constant within a single .WAV file. When the .WAV file is closed, these settings will revert to their previous values. The current sampling rate is always displayed in the status bar along the bottom of the application.

Continuous Recording Capability

Wave files can store up to 2 GB of data in a single file (the corresponding time depends upon the sampling format). When this limit is reached, the program will momentarily stop recording, save the file and then start recording again in a new file. If the file is does not yet have a name, the program will give it a name using the computer date and time. The file will be stored in the \wave folder as specified in the <File><Set Paths> dialog. This feature applies to the hard disk recording mode only.

Example: c:\specplus\wave\recording_2002_11_26_102011.wav
(in this example the date and time were 10:20:11 Nov, 26 2002 when the file was saved)

Important:

During hard disk recording sessions, do not perform other disk intensive operations (such as opening a major application). Although Windows is a multitasking operating system, it relies on the cooperation of individual applications and cannot guarantee immediate access to system resources.

Chapter 7 - Views

Time Series
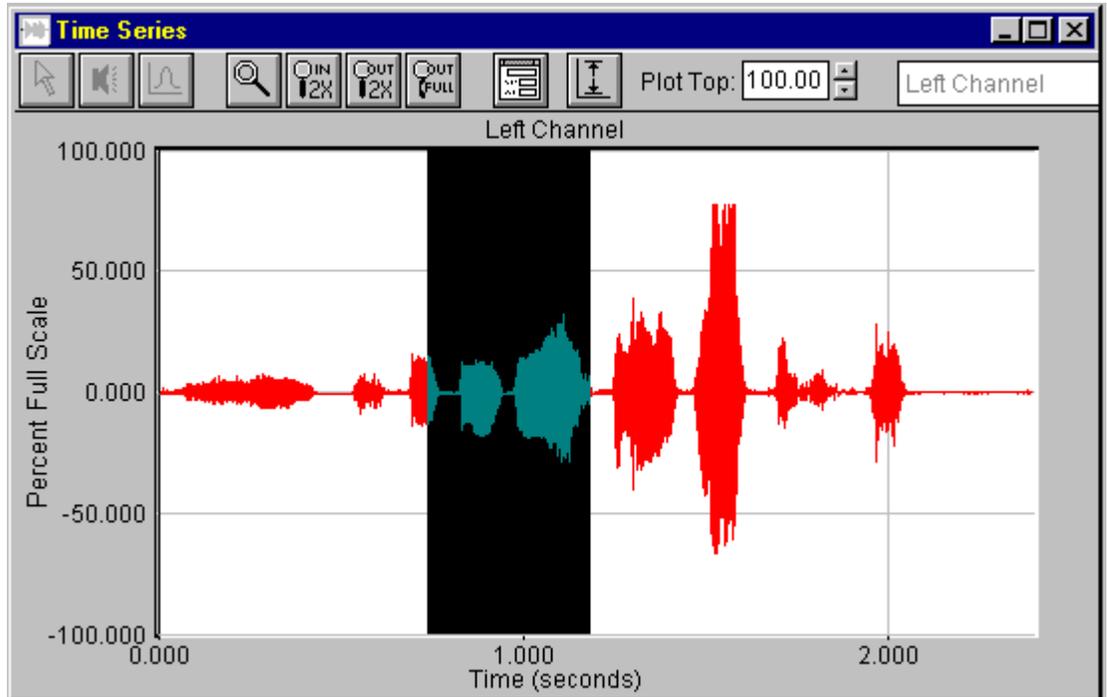
This view displays the raw digitized audio data and is similar to an oscilloscope display with the amplitude shown on the vertical axis and time on the horizontal axis.

When you are using the Real-Time mode, the plot will display the lastest 60 seconds of acquired data.   In the Recorder or Post-Processing modes you may zoom and scroll through the entire data file using the toolbar and scrollbar controls. The horizontal scaling is in seconds relative to the start of the file. Double clicking with the left mouse button anywhere in the plot will move the current file position to the pointer location. In addition, the spectrum corresponding to the new pointer location will be computed and displayed in the spectrum view (if the spectrum display is open).

Time Series Display Options

The time series display options dialog box can be activated by selecting the <Options><Time Series> menu or by clicking on the options dialog icon button in the view toolbar.
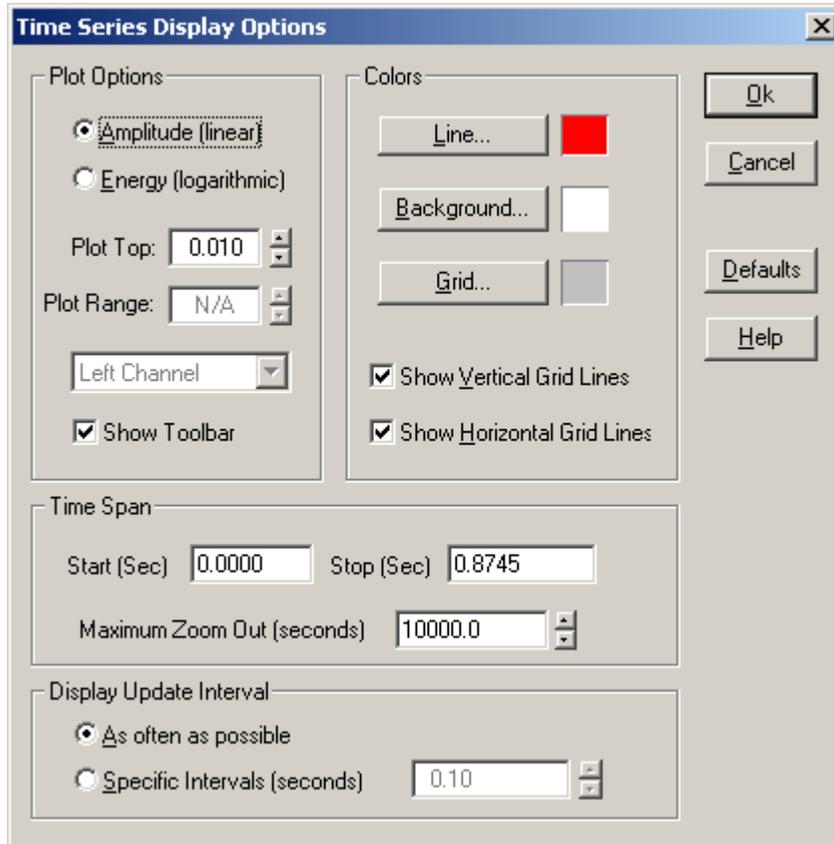
Plot Type: If Amplitude is selected, the display is similar to an oscilloscope (the data is scaled linearly around the centerline). When the Energy option is selected, the data is scaled in Decibels with 0 dB representing 100% full scale. The actual scale values can be modified by using the calibration utility.

Plot Top/Range: The Plot Top and Range values determine the amplitude axis scale values. When the amplitude option is selected, the Plot Top value determines the upper and lower plot limits and the Plot Range value is not used. When the Energy option is selected, the Plot Top is the value at the upper edge of the plot and the lower edge of the plot is equal to the Top value minus the Range value.

Channel Selection: This control allows you to change the display mode from single to dual channel configuration. In addition, it allows you to display the difference between the left and right channels. If Mono mode is selected (see Settings), this control is disabled and only the left channel can be displayed.

Colors: The color of each plot feature can be changed. Only solid colors can be selected. The default button will restore the initial colors settings.

Grid Lines: Show or hide the grid lines.

Show Toolbar: Show or hide the time series plot toolbar.

Pull Cursor to Trace: If this option is enabled, the measurement cursor will automatically "pull" to the time series trace along the amplitude axis.

Maximum Zoom Out Time: Because .WAV files are often very large, displaying the entire file at one time can take an unacceptable amount of time (the entire data file must be read). This value controls how many seconds of time will be displayed when the Zoom Out Full button is pressed.

Display Update Interval: Controls how often the display is updated. This is useful on some laptop computers with slow display refresh rates to reduce flicker.

Note:

For maximum refresh rates, close all other views - no FFT will be performed.

Spectrum
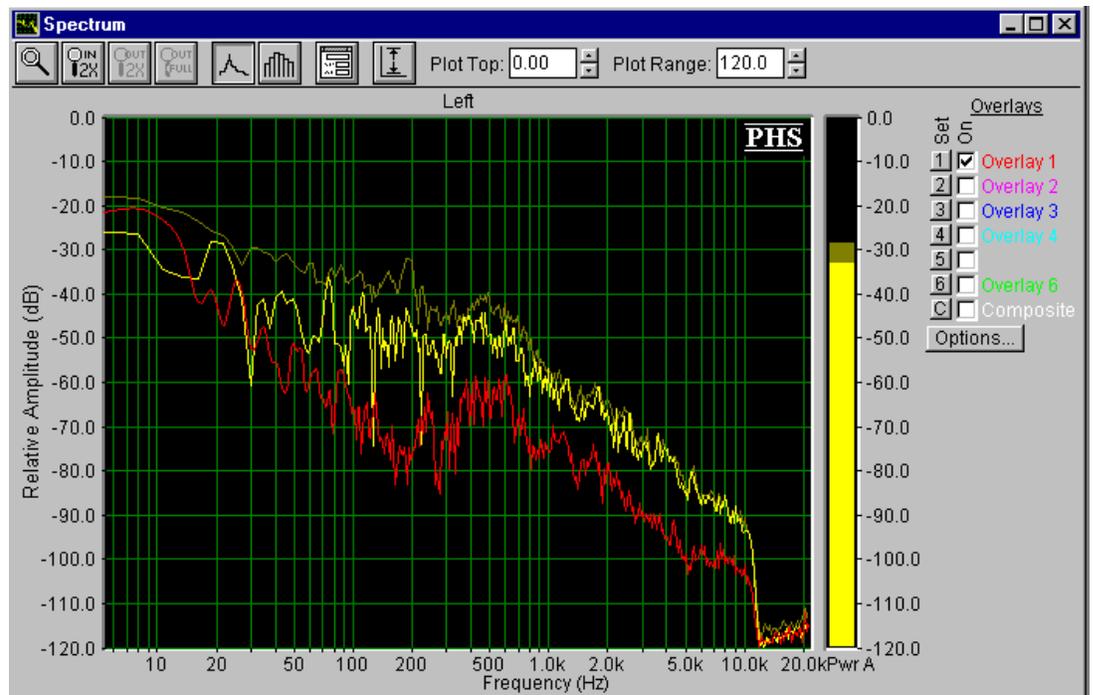
This view is a 2 dimensional plot of the spectrum. The horizontal axis shows the frequency in Hertz (Hz). The vertical axis shows the amplitude

of each frequency line. The amplitude scale can be modified using the calibration utility

The total number of available spectral lines is always equal to the FFT size / 2. Thus for a 1024 point FFT, there will be 512 spectral lines. The measurement will always start at 0 Hz and extend to 1/2 of the selected Sampling Rate divided by the Decimation Ratio.

The maximum displayed spectrum is limited on the low and high ends. The lowest 3 bins contain a DC component and are not displayed; the top 1% of the span often contains alias components and therefore is not displayed.

The zoom buttons on the toolbar allow you to easily adjust the displayed span within the available measurement span; however, the FFT algorithm requires that all frequency lines must be computed regardless of whether or not they are displayed.

The mouse can be used to directly measure the frequency and amplitude information directly off the plot. In addition, the <Edit><Copy> menu item will place the current spectral data on the clipboard in tabular format. This can be directly pasted into any spreadsheet or text file.

Spectrum Display Options

The spectrum display options dialog box can be activated by selecting the <Options><Spectrum> menu or by clicking on the options dialog icon button in the view toolbar.
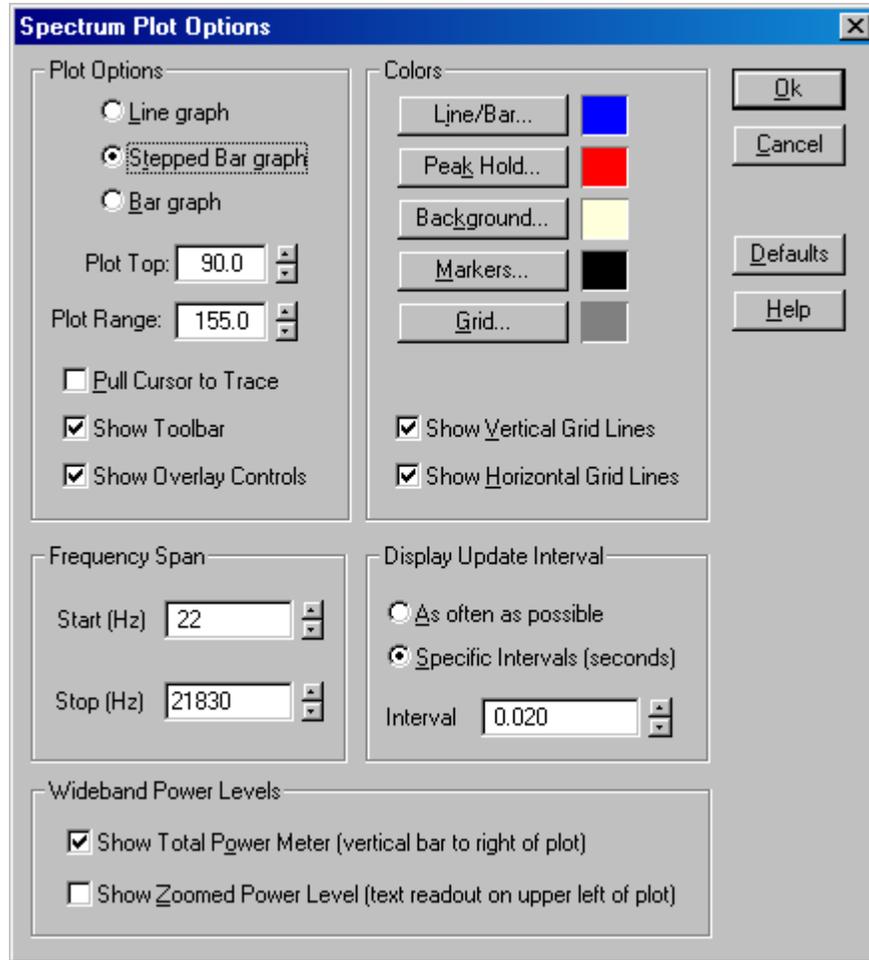
Plot Type: the line or stepped bar graph option is recommended when using large FFT sizes. The bar graph option is recommended when Octave scaling is selected.

Plot Top/Range: The Plot Top and Range values determine the amplitude axis scale values. Plot Top is the value at the upper edge of the plot. The lower edge of the plot is always equal to the Top value minus the Range value. When linear scaling is selected for the amplitude axis, the Plot Range value is set to zero and cannot be changed.

Show Total Power Meter: the total power meter is located along the right side of the spectrum plot and shows the "Total" power of the spectral data. This value is equal to the Total Power utility. A different weighting curve can be applied to this value (this is set in the Scaling dialog box).

Show Zoomed Power Level: If enabled, the RMS Power level will be computed and displayed above the top left corner of the plot. This power level is computed for the currently displayed span and will change as you zoom in and out. Use this feature to compute the power within any

segment of the spectrum. The weighting used for this calculation is the same as the spectrum weighting parameter in the scaling dialog box.

Pull Cursor to Trace: If this option is enabled, the measurement cursor will automatically "pull" to the spectrum trace along the amplitude axis.

**Colors**: The color of each plot feature can be changed. Only solid colors can be selected. The "Default" button will restore the initial colors settings.

Grid Lines: Show or hide the grid lines.

Show Toolbar: Show or hide the plot toolbar

Frequency Span: Although the displayed frequency span can easily be selected using the zoom buttons on the toolbar, you can also manually enter the span yourself. The span is limited to the total available span. The actual span displayed may vary slightly from the requested values because the values are adjusted to the nearest bin.

Display Update Interval: Controls how often the display is updated. This is useful on some laptop computers with slow display refresh rates to reduce flicker.

Spectral Overlays

The overlay feature allows you to display multiple spectral traces on the same plot for quick comparisons with previous measurements. Controls for creating and clearing overlays are located along the right edge of the spectrum view. Up to 6 overlays can be displayed for each channel. In addition, a "Composite" overlay can be set which is computed from other overlays.

Operation
Clicking one of the numbered "Set" overlay buttons will create an overlay using the contents of the current spectrum. The overlays can be turned On and Off by clicking the checkbox labeled "On".

Clicking the "Options" button brings up a dialog box with additional controls:
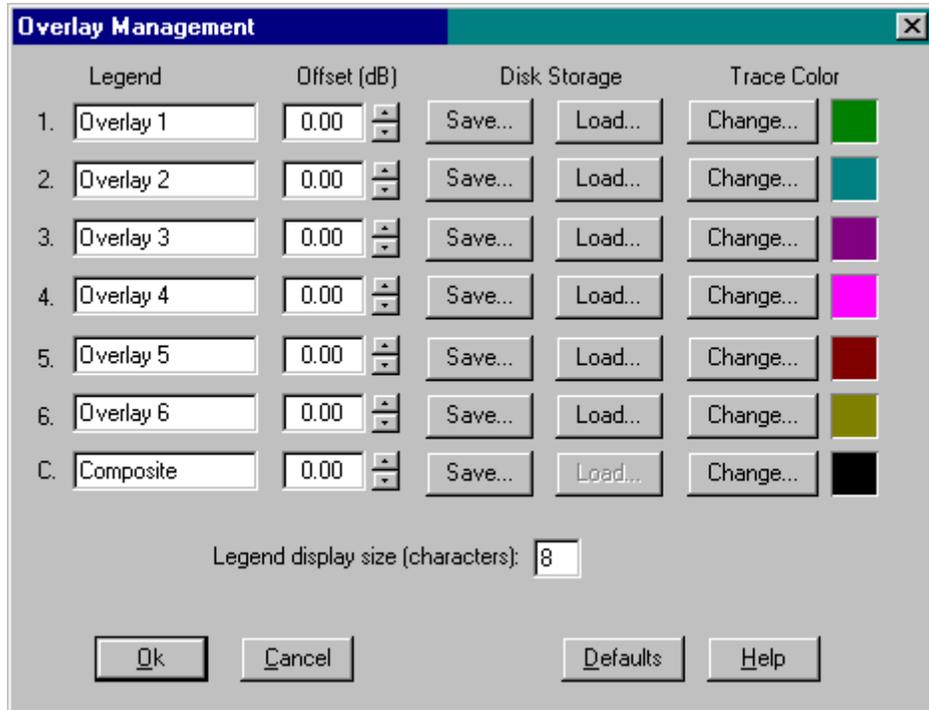
Legend: You may enter a descriptive legend up to 24 characters long. This legend is displayed on the spectrum view and printed on any printouts. The legend is saved as part of the overlay file format and restored whenever you load the file from disk.

Offset: You can apply an offset in dB to each overlay; this is useful for aligning overlays for comparison purposes.

Disk Storage: Use these controls to save and load the overlay traces from disk. Temporary overlay files are used for storing the overlay data between sessions. These files are named L1.OVL, L2.OVL, L3.OVL, L4.OVL, L5.OVL, R1.OVL, R2.OVL, R3.OVL, R4.OVL, R5.OVL, and R6.OVL. Do not use these filenames for saving your own data as they will be overwritten. Any offset is NOT included in the saved overlay file contents.

Trace Colors: these controls allow you to select the display colors for each of the overlay traces and legends. These colors are also used when printing the spectrum view on a color printer. If you print to a monochrome printer, each of the overlays will have a unique line style.

Legend Display Size: this value will adjust the width of the rights border of the spectrum plot to show the specified number of legend characters. Space may be reserved for up to 24 characters.

Composite Overlay

Pressing the overlay button labeled "C" will allow you to setup a "Composite" overlay. The composite overlay can be either an average of up to 6 selected overlays or the difference between any 2 selected overlays. Any offset selected for the dependent overlays is applied when the composite values are computed.   An offset can also be applied to the composite overlay itself.

Overlay File Format:

Overlay files can be created manually. Overlay files are simple ASCII text files containing two columns of numbers separated by a tab. The left column contains the frequency; the right column contains the corresponding amplitude value. If a semicolon is placed on the first column, the entire line will be treated as a comment and ignored. Legends are the first line in the file and have a semicolon in the first column. At least 3 data points are required and up to 32768 points can be used.

A cubic spline algorithm is used to interpolate between each data point. This file format is interchangeable with the microphone compensation file format. When manually creating an overlay file the results can have unexpected excursions due to the cubic spline interpolation algorithm. You can turn off this algorithm by adding a line to the beginning of the file as follows:

;SPLINE OFF

Notes:

If you regularly interchange overlay and microphone compensation files, you may want to set their file paths (See Set Paths command) to the same directory. You can then easily select whether to use .MIC or .OVL file types from the file open/save dialog boxes.

Phase
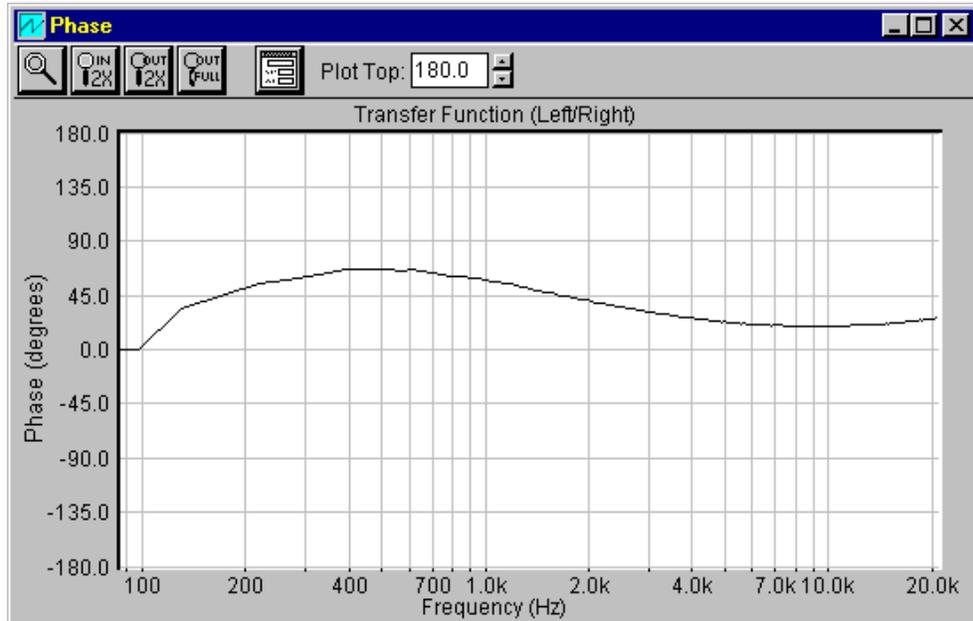
This view displays the phase of the signal versus frequency. It is most useful when examining the phase difference between the right and left channels. To do this, select the "stereo" option in the settings dialog box and then choose the "Transfer Function" option.

The frequency scaling is governed by the current scaling options. The zoom controls located on the toolbar can be used to adjust the displayed frequency span anywhere within the measurement range.

The mouse can be used to measure the frequency and phase information directly off the plot. In addition, the <Edit><Copy> menu command will place the current phase data on the clipboard in tabular format. This can then be directly pasted into any spreadsheet or text file.

The phase is greatly affected by the choice of smoothing window. Selecting "Uniform" will produce the cleanest phase response.

Phase Display Options

The Phase display options dialog box can be activated by selecting the <Option><Phase> menu or by clicking on the options dialog icon button in the view toolbar.
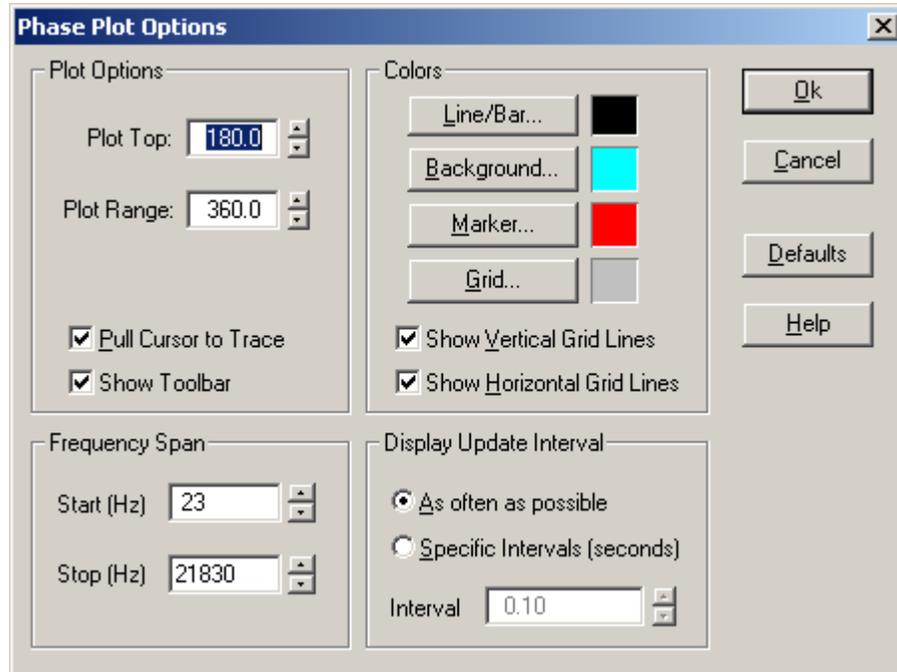
Plot Top/Range: The Plot Top and Range values determine the phase axis scale values. Plot Top is the value at the upper edge of the plot. The lower edge of the plot is always equal to the Top value minus the Range value.

Pull Cursor to Trace: If this option is enabled, the measurement cursor will automatically "pull" to the trace along the phase axis.

Colors: The color of each plot feature can be changed. Only solid colors can be selected. The "Default" button will restore the initial colors settings.

Grid Lines: Show or hide the grid lines.

Show Toolbar: Show or hide the plot toolbar.

Display Update Interval: Controls how often the display is updated. This is useful on some laptop computers with slow display refresh rates to reduce flicker.

Spectrogram
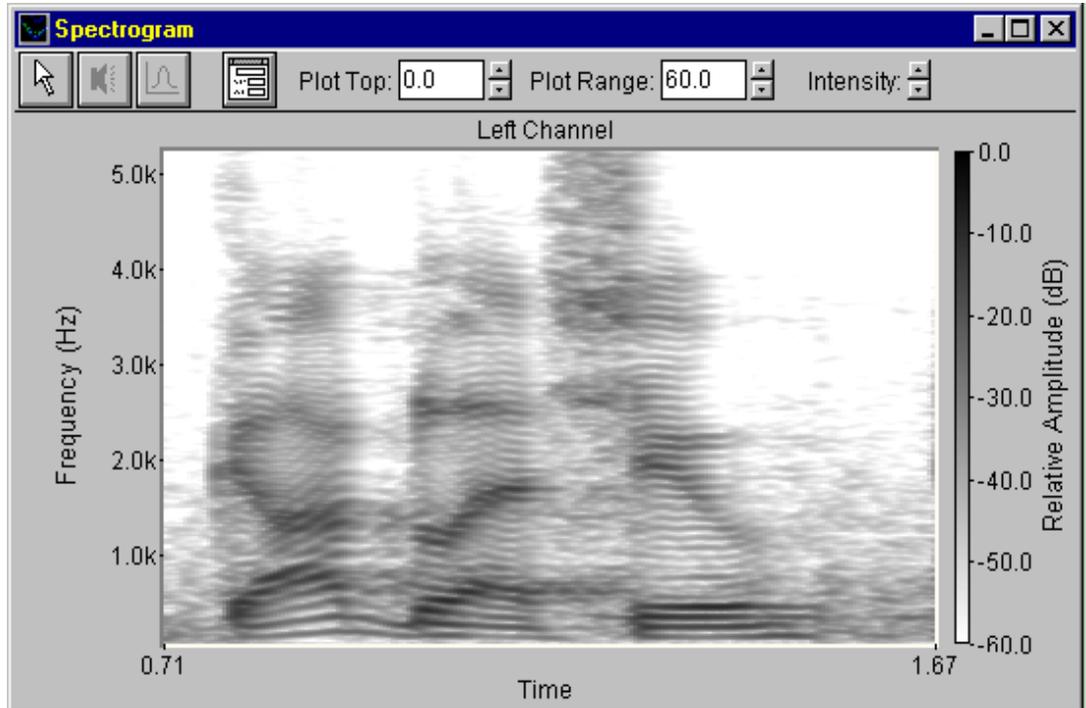
This view displays the spectral data over time with the amplitude shown in color. For some users, this display is difficult to interpret, but with a little practice it is one of the most useful displays available for spectral analysis. Simply think of it as if you were looking down on the 3-D Surface plot and that the amplitude is shown in color. The color bar to the right of the plot shows the amplitude/color relationship.

Double clicking the left mouse button within the spectrogram will display the corresponding spectral and time series data in their respective views.

Spectrogram Display Options

The spectrogram display options dialog box can be activated by selecting the <Options><Spectrogram> menu or by clicking on the options dialog icon button in the view toolbar.
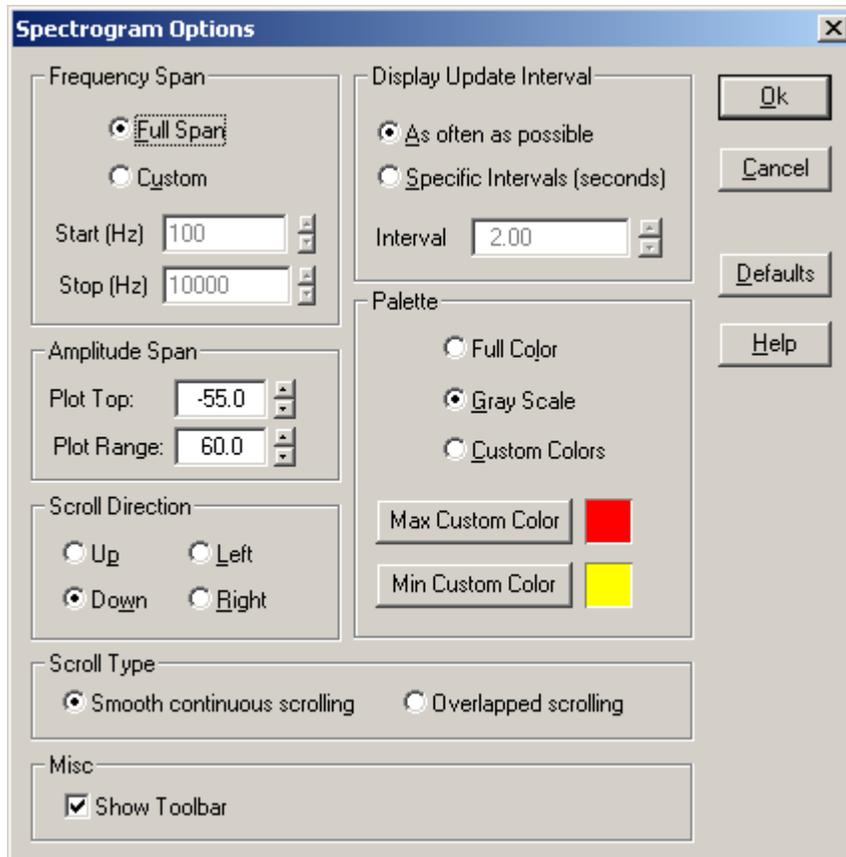
Frequency Span: If "Full Span" is selected, the spectrogram will show the entire available frequency range. When "Custom Span" is selected, the frequency values can be directly entered. When 1/3 Octave scaling is used, Custom span information is ignored.

Display Update Interval: Normally a new line is added to the spectrogram whenever a new FFT is computed. You may elect to only update the spectrogram every N seconds. This is most useful in the Real-Time mode. In this way, long term frequency -vs- time studies can be conducted.

Palette: There are three choices of palettes. The "Color" palette transitions from black to red, the "Grayscale" palette changes from white to black, and the "Custom" palette lets you choose the minimum and maximum colors. If you are using 256 color video mode, palette changes are immediate and the Intensity control will operate (see Video Display Modes for more information).

Intensity Control: The Intensity control "slides" the palette values up or down within the color/grayscale limits. The intensity control is only located on the toolbar and not within the options dialog box. This is

because the intensity control is meant to be used while working with the spectrogram where visual feedback is immediately available. The video display mode must be set to 256 colors for the intensity control to function.

Scroll Direction: Specifies the direction in which the spectrogram will scroll with increasing time.

Scroll Type: Specifies the type of scrolling. Smooth continuous scrolling scrolls the display with new information added at the edge. Overlapped scrolling does not scroll the display but overlaps when the end of the display is reached.

Plot Top/Range: The Plot Top and Range values determine the amplitude (color bar) values. Plot Top is the value at the upper edge of the color bar. The lower edge of the color bar is always equal to the Top value minus the Range value. When linear scaling is selected for the amplitude axis , the Plot Range value is set to zero and cannot be changed.

Show Toolbar: Show or hide the plot toolbar.

Notes:
- Any changes to the display options (with the exception of Direction) will cause the spectrogram to automatically rebuild itself with the new settings (press the Escape key to cancel rebuilding). This behavior will not occur when the analyzer is in the Real-Time mode because the underlying time series data is no longer available.
- The display can also be cleared by using the Clear All command under the View menu.

3-D Surface

This display is a perspective view of the spectral data over time. Frequency is shown on the X-axis and time on the Y-axis. This view is useful for visualizing the dynamics of the spectral data.

Double clicking with the left mouse button within the plot will display the corresponding spectral and time series data in their respective views (if they are open).
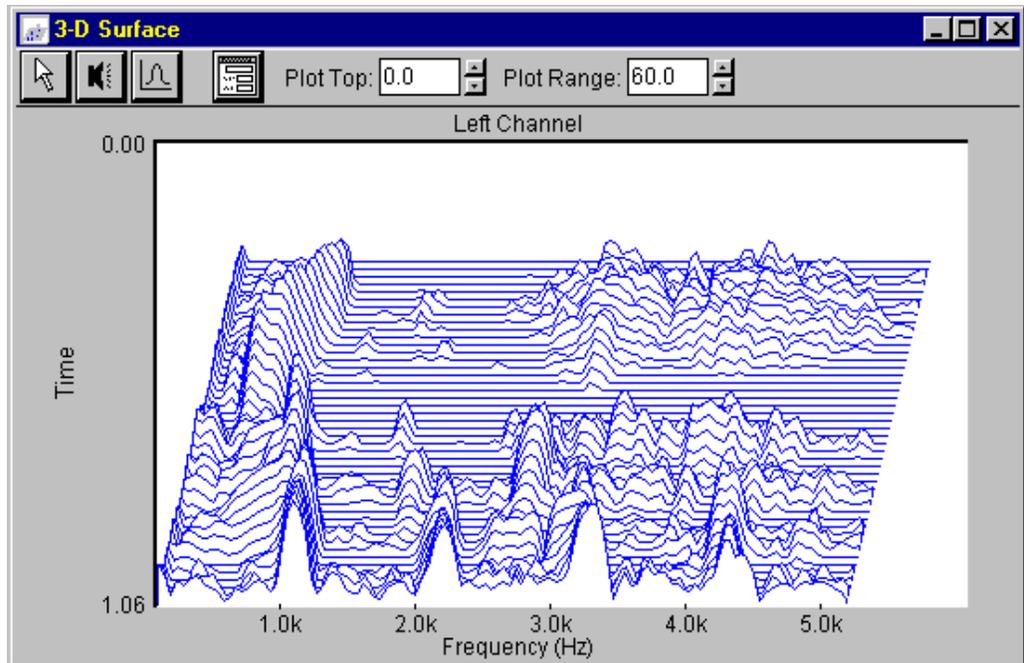
3-D Surface Display Options

The 3-D Surface display options dialog box can be activated by selecting the <Options><3-D Surface> menu or by clicking on the options dialog icon button in the view toolbar.
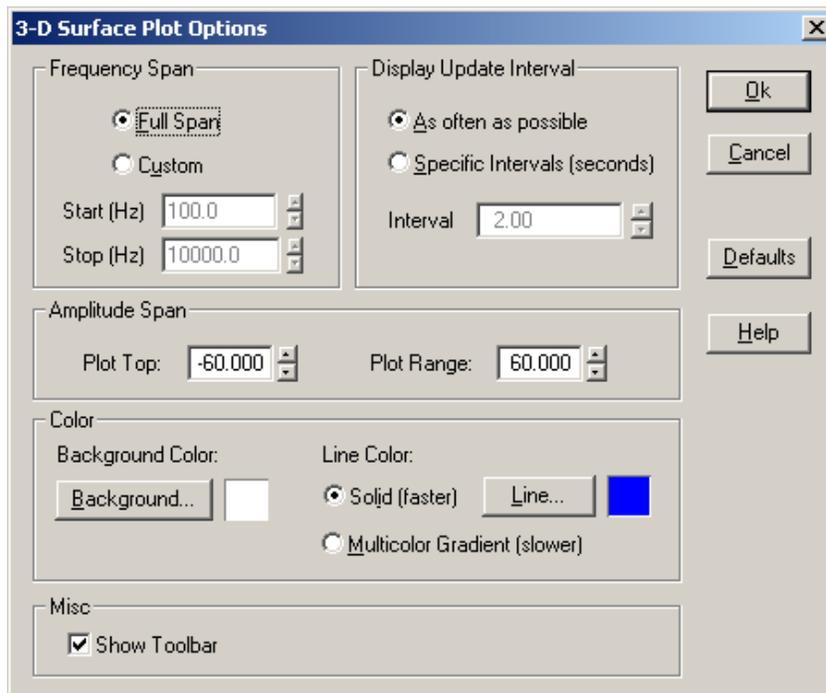
Frequency Span: If "Full Span" is selected, the display will show the entire available frequency span. When "Custom Span" is selected, the frequency values can be directly entered.

Display Update Interval: Normally a new line is added to the plot whenever a new FFT is computed. You may elect to only update the plot every N seconds. This is most useful in the Real-Time mode. In this way, long term frequency -vs- time studies can be conducted.

Plot Top/Range: The Plot Top and Range values determine the amplitude values. Plot Top is the value at the upper edge of the plot. The lower edge of the plot is always equal to the Top value minus the Range value. When linear scaling is selected for the amplitude axis, the Plot Range value is set to zero and cannot be changed.

Colors: The color of each plot feature can be changed. Only solid colors can be selected. The "Default" button will restore the initial colors settings.

Show Toolbar: Show or hide the plot toolbar.

Notes:

Any changes to the display options will cause the 3-D Surface plot to automatically rebuild itself with the new settings (press the Escape key to cancel the rebuild). This behavior will not occur when the analyzer is in the Real-Time mode because the underlying time series data is no longer available.

Chapter 8 - Options

Settings
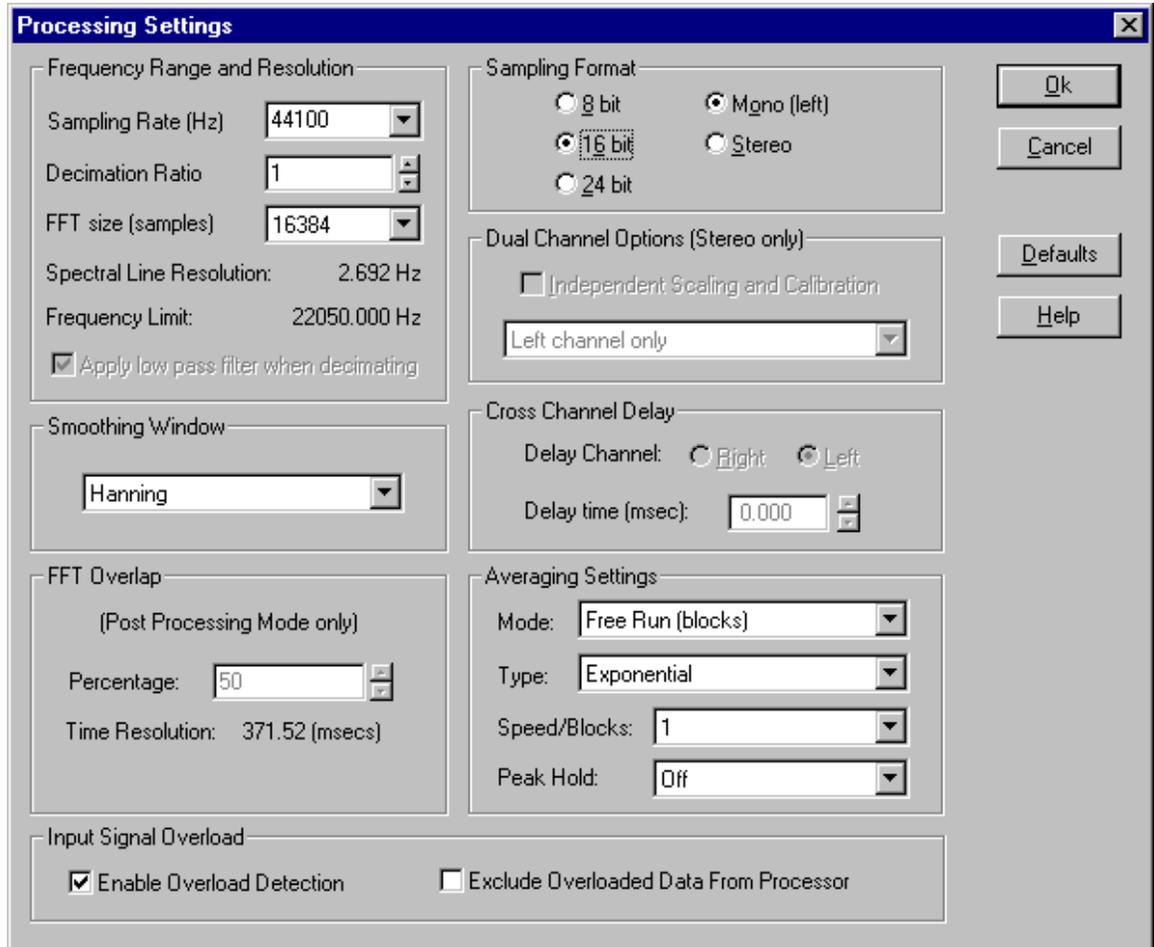
Sampling Rate

SpectraPLUS allows you to select any Sampling Rate supported by your sound card. The measured frequency range is always 0 to 1/2 the Sampling Rate divided by the decimation ratio.

The Sampling Rate determines how many times a second that the analog input signal is "sampled" or digitized by the sound card. A sampling rate of 10,000 Hz will sample the signal every .0001 second. The possible range of sampling rates are limited by the capability of your sound card.

An important principal from digital signal processing is the "Nyquist Sampling Theorem" which states that any signal can be represented if

sampled at least twice the rate of the highest frequency of interest. This means that if you wish to measure a 3,000 Hz signal, the sampling rate must be greater than 6,000 Hz.

When a .WAV file is opened, the Sampling Rate and Format is changed to match the rate at which the file was recorded and the sampling rate edit box is disabled - the Sampling Rate must remain constant within a single .WAV file. When the .WAV file is closed, these settings will revert to their previous values. The current sampling rate is always displayed in the status bar along the bottom of the application.

Note:

To determine if your card supports a rate other than the displayed choices, simply enter the value in the edit control and press OK. The analyzer will attempt to use the specified rate and a message box will appear if it is unsuccessful.

Decimation Ratio

The decimation ratio value controls the ratio with which the digitized audio is down-sampled". This feature is used to achieve a lower sampling rate than that supported by the sound card. This will allow greater frequency resolutions at low frequencies for vibration measurements and similar applications. For instance, if your sound card's lowest supported sampling rate is 11025 Hz, decimating by a factor of 10 will effectively reduce this to 1102.5 Hz. The resulting measurement range will then be 0 to 551.25 Hz.

A value of 1 will not resample the data at all.
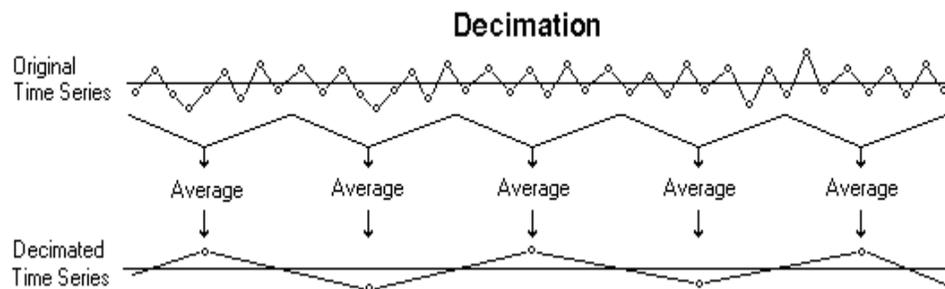
Decimation is actually a very simple process. Let's say we want to down-sample by a factor of 8:1, all we need to do is to throw away 7 out of every 8 samples.

You need to be careful when decimating to make sure that the original signal contains no energy above the "new" Nyquist frequency otherwise

aliasing will occur. If this is not the case you should either use an external low pass filter or check the box labeled "Apply Low Pass Filter when Decimating".

Note:

The Spectral Line Resolution and Frequency Limit values will update to automatically reflect the current Sampling Rate, FFT size, and Decimation ratio settings.
The Time Series view always displays data prior to the decimation process.

FFT Size

The Fourier Transform is a mathematical tool developed and named after Jean Baptiste Fourier (1768 - 1830) and is commonly used to convert a signal from the time domain (amplitude-vs-time) to the frequency domain (amplitude-vs-frequency). A frequency domain plot is also known as a spectra plot.

The Fast Fourier Transform (FFT) is a computationally efficient implementation of the Fourier Transform developed by J.W. Cooley and J.W. Tukey in 1965. The Fast Fourier Transform is limited to block sizes which are even powers of 2.

For example, if a FFT is performed on a pure sinusoidal signal, the resulting spectra would be single peak (line). Real world signals are a composite of many sinusoidal signals; examining the signal's spectra clearly shows the frequency tones which are present.

The selected FFT size directly affects the resolution of the resulting spectra. The number of spectral lines is always 1/2 of the selected FFT size. Thus a 1024 point FFT produces 512 output spectral lines.

The frequency resolution of each spectral line is equal to the Sampling Rate divided by the FFT size. For instance, if the FFT size is 1024 and the Sampling Rate is 8192, the resolution of each spectral line will be:

$$8192 / 1024 = 8 \text{ Hz.}$$

Larger FFT sizes provide higher spectral resolution but take longer to compute.

Notes:
- The Spectral Line Resolution and Frequency Limit values will update to automatically reflect the current Sampling Rate, FFT size, and Decimation ratio settings.
- The current FFT size is always displayed in the status bar along the bottom of the application.

Smoothing Windows

If a sinewave is passing through zero at the beginning and end of the time series, the resulting FFT spectrum will consist of a single line with the correct amplitude and at the correct frequency. If, on the other hand, the signal level is not at zero at one or both ends of the time series record, truncation of the waveform will occur, resulting in a discontinuity in the sampled signal. This discontinuity causes problems with the FFT process, and the result is a smearing of the spectrum from a single line into adjacent lines. This is called "leakage"; energy in the signal "leaks" from its proper location into the adjacent lines.

Leakage could be avoided if the time series zero crossings were synchronized with the sampling times, but this is impossible to achieve in practice. The shape of the "leaky" spectrum depends on the amount of signal truncation, and is generally unpredictable for real signals.

In order to reduce the effect of leakage, it is necessary that the signal level is forced zero at the beginning and end of the time series. This is done by multiplying the data samples by a "smoothing window" function, which can have several different shapes. The difference between each smoothing window is the way in which they transition from the low weights near the edges to the higher weights near the middle of the sequence. If there is no windowing function used, this is called "Rectangular", "Flat", or "Uniform" windowing.

While the smoothing window does a good job of forcing the ends to zero, it also adds distortion to the time series which results in sidebands in the spectrum. These sidebands, or side lobes, effectively reduce the frequency resolution of the analyzer; it is as if the spectral lines are wider. The measured amplitude of the weighted signal is also incorrect because a portion of the signal level is removed by the weighting process. To make up for this reduction in power, windowing algorithms give extra weight to the values near the middle of the sequence.

The figure shown below compares the spectrum of a signal using a Uniform window and a Blackman window. Remember that a Uniform window is actually no window at all.
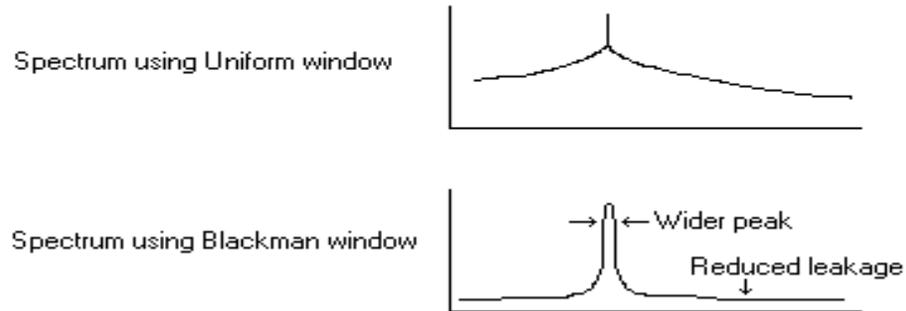
Characteristics of various smoothing windows

Window Type	Frequency Resolution	Amplitude Resolution	Leakage Suppression	Application
Bartlett	Fair	Fair	Moderate	
Blackman	Fair	Good	Excellent	Distortion Measurements
Flattop	Poor	Excellent	Moderate	High Accuracy Amplitude Measurements. Swept Sine testing
Hamming	Fair	Fair	Fair	
Hanning	Fair	Excellent	Excellent	Distortion testing, noise measurements
Kaiser	Fair	Fair	Poor	
Parzen	Fair	Fair	Poor	
Uniform	Excellent	Poor	Poor	High resolution frequency testing

Notes:

- The Time Series View displays the signal prior to the application of any smoothing window.
- The effects of these functions are most noticeable when using Logarithmic scaling.

Averaging Mode

Free Run (blocks) - this is the default mode and updates the analyzer each time a new FFT block of data is computed. In this mode the average block size directly determines how many spectra blocks are averaged together to compute a moving average. For instance, if the averaging block size is set to 4, the spectrum currently displayed is a moving average of the previous 4 traces.

If the signal you are measuring is rapidly changing in frequency, a low averaging block size should be used. Use a high block size to "dig" out a steady state signal buried in a noisy background.

Setting the averaging size to "infinite" causes the program to average successive spectral traces continuously. Stopping and restarting the processor will reset the average. A single left click on the main toolbar will also reset the average.

Sound Level Meter (SLM) - this mode sets the averaging algorithm to match that of a standard Sound Level Meter (SLM). The average speed controls the spectral averaging period and decay time as follows:

- Off - no averaging is performed.
- Fast - computes a running average with a decay rate of 32 dB/sec.
- Medium - computes a running average with a decay rate of 20 dB/sec.
- Slow - computes a running average with a decay rate of 4 dB/sec.
- Forever - accumulates the data for as long as the analyzer is running.

Exponential averaging should be used when the Sound Level Meter mode is selected.

Averaging Type

Exponential - this method produces an average where the most recently acquired spectra have more influence on the average than older ones. If you were to suddenly shut off the input signal, the decay rate will follow an exponential curve. Exponential averaging should be used when the Sound Level Meter mode is selected.

Linear - this method is a straight linear average of the spectra over time. Each spectra block contributes equally to the average. This type of averaging is also known as "stable averaging"

Vector - this method performs a complex (vector) average of successive spectra over time. Because vector averaging includes the phase component in the computation, triggering must be employed if meaningful results are to be expected. This type of averaging is also known as "synchronous averaging".

Exponential averaging is the default as it is more computationally efficient.

Notes:
In both the Real-Time and Post-Processing modes, the averaging block size can be modified while processing by using the "Avg" control on the toolbar.

Peak Hold

When Peak Hold is used, the analyzer will display a secondary trace showing the highest amplitude values for each frequency bin. Selecting Fast, Medium or Slow will cause the peak hold values to temporarily hold before resetting. Forever will hold the values without releasing them. Stopping and restarting the processor will reset the peak hold values. A single left click on the main toolbar will also reset the peak hold buffer.

Peak Hold can be controlled from either the settings dialog box or from the main application toolbar in the Real-Time and Post-Processing Modes.

Overlap Percentage

This powerful feature allows the time axis on the Spectrogram and 3-D Surface plot to be "stretched" by reprocessing portions of the time series. The setting determines what percentage of the FFT size block of data points is reused for the next trace. High overlap percentages (90+%) and low FFT block sizes provide the highest time resolution but take longer to process the data file. This feature is only used in Post-Processing mode.

Large FFT sizes produce high frequency resolution but poor time resolution while low FFT sizes have the opposite effect. Overlap processing can be used to provide high resolution in both the frequency and time axis. For example, if a signal contains two discrete tones separated in frequency, only one of which is on at a time, it is difficult to discern the relative timing of each tone because both frequencies are often contained within a single FFT size time series block. If we perform an FFT and, instead of moving forward a whole FFT size block in the series,

move forward only a small amount then we can isolate the pulse timing for each tone. The following figure illustrates the effect of overlap processing
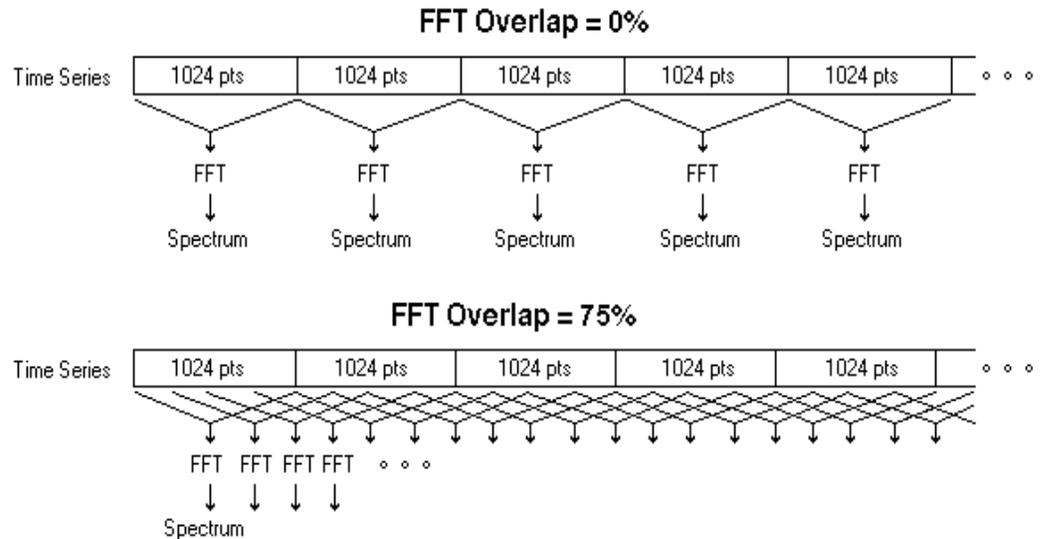
Notice that 4 times as many FFT's are performed in the second case; this will increase the time resolution by a factor of 4.

The software can automatically compute and use the optimum overlap percentage for you. Using either the Time Series, 3-D Surface or Spectrogram views, simply select a time segment for editing and right click anywhere in the plot. An Action Menu will appear with choices such as "Compute and Display Spectrogram". The analyzer will automatically compute the overlap percentage necessary to fill the plot from the selected time segment. See the Right Click Action Menu topic for more details.

Notes:

High overlap percentages will require greater processing times.

Sampling Format

An important factor in determining the quality of the sampled sound is the number of bits per sample. This specifies the number of discrete levels that are used in the analog to digital (A/D) conversion process and directly affects the dynamic range of the measurement. The theoretical dynamic range is shown below:

- 8 bit sampling precision = 48 dB
- 16 bit sampling precision = 96 dB
- 24 bit sampling precision = 144 dB

In practice, the achievable dynamic range is less that the theoretical maximum due to internal noise and other factors. Refer to your sound card documentation for the actual performance figures. It is recommended that you use the highest sampling precision supported by your card with the real-time mode since the data is not being stored (the actual FFT process is floating point and requires the same computational time regardless of the sampling resolution).

When "Mono" is selected, you should connect the test signal to the left channel on your sound card. This is normally the tip on a standard 1/8" stereo connector. Consult your sound card documentation for further details.

When "Stereo" is selected, the "dual channel processing options" will be enabled. This will allow you to display either the left, right, both, or a variety of advanced cross channel measurements.

Dual Channel Processing Options

Dual channel spectrum analysis provides several very powerful advanced measurements.

You must select the Stereo format before these options are available. Note that once you have opened or recorded a mono file, you cannot change to stereo; you must first close the file. If you are in the Real Time mode, you can change between mono or stereo at any time. It is also possible to combine two mono files into a single stereo file using the <File><Merge> command.

Independent Scaling and Calibration

This option allows each channel to be scaled and calibrated independently. When selected each channel will be displayed in a separate window instead of in a split screen view. In addition the Scaling {linkID=360}and Calibration {linkID=750}dialogs will allow you to choose separate parameters for each channel.

This powerful feature allows you to analyze two completely different types of signals at once. For example you can configure the analyzer to show accelerometer data scaled in G's in one channel and 1/3 octave data scaled in SPL from a microphone in the other.

Definitions

Auto-spectrum - calculated by multiplying a spectrum by its complex conjugate (opposite phase). The auto-spectrum is real and is identical to the magnitude response. The auto-spectrum is what is normally displayed in the single channel spectrum view.

Cross-spectrum - calculated by multiplying one spectrum by the complex conjugate of a second spectrum. The cross-spectrum is a complex quantity (has both real and imaginary components). The magnitude of the cross-spectrum represents the amount of power common to both channels, and the phase represents the phase difference between channels. For example, If two identical signals are 180 degrees out of phase, the magnitude of the cross spectrum is zero. If two channels are identical, their cross-spectrum is equal to the auto-spectrum of either channel alone.

Transfer Function - is the ratio between two channels and can be computed in one of two ways:

- **Real** - this is the simplest method and is simply the ratio between the auto-spectrum (magnitude) of each channel. The relative phase of each channel does not contribute to the result.
- **Complex** - this is the ratio between the cross-spectrum of the two channels and the auto-spectrum of the reference channel. The relative phase of each channel contributes to the result.

The transfer function is a very powerful feature. It allows you to quickly and accurately compute the frequency response of a device under test. See the topic "Transfer Function Example" for a typical test setup.

Coherence Function - is the ratio of the squared magnitude of the cross-spectrum and the product of the auto-spectrum of both channels. It measures the degree of linearity between the channels and is analogous to the squared correlation coefficient used in statistics. Two perfectly coherent signals have a coherence value of 1.0 (0 dB).

Dual Channel Processing Options:

- **Left Channel Only** - processes and displays the auto-spectrum (magnitude) of the left channel only.
- **Right Channel Only** - processes and displays the auto-spectrum (magnitude) of the right channel only.
- **Both Left and Right** - processes and displays the auto-spectrum (magnitude) of both the left and right channels. The results are displayed in independent windows.
- **Average Left and Right** - Averages the auto-spectrum of the left and right channels.
- **Real Transfer Function** - computes and displays the real transfer function between the two channels.
- **Complex Transfer Function** – computes and displays the complex transfer function between the two channels.
- **Cross Spectrum** - computes and displays the cross-spectrum of both channels.
- **Coherence Function** - computes and displays the coherence function between the two channels

Notes:

These are spectral processing options and do not apply to the time series view.

The Cross-Spectrum, Complex Transfer Function and Coherence Functions operations are very sensitive to phase delays between channels. Use the Cross Channel Delay feature to delay one channel with respect to the other.

Cross Channel Delay

The Cross-Spectrum, Coherence Function and Complex Transfer Function operations are very sensitive to phase delays between channels. This feature allows you to delay one channel with respect to another to compensate for this delay.

All delay values are rounded off to the nearest sampling interval (1/Sampling Rate). Clicking the up/down arrows to will change the delay value by exactly one interval.

Notes:

This delay only applies when you are in stereo mode.

Overload Detection

Overload is a condition where the input signal strength exceeds the full scale sensitivity of the sound card input. Known in the analog world as "clipping", it will cause severe distortion to occur.

If overload checking is enabled, the analyzer will test each incoming data block to determine if an overload condition is present and display a message in the status bar along the lower edge of the application. This message is shown in bright red and is very noticeable.

If you have chosen to exclude overloaded data from processing, the analyzer will throw away any overloaded data.

If an overload condition occurs, you will need to either reduce the input signal strength or decrease the input sensitivity of the sound card using the sound card's mixer utility.

Scaling

When the "Independent Scaling and Calibration" option is selected in the Settings dialog box, the scaling control dialog box will change to display separate parameters for each channel and each channel will be plotted in a separate window. This allows you to use two completely different types of scaling at once. For example you can configure the analyzer to show accelerometer data scaled in G's in one channel and 1/3 octave data scaled in SPL from a microphone in the other.
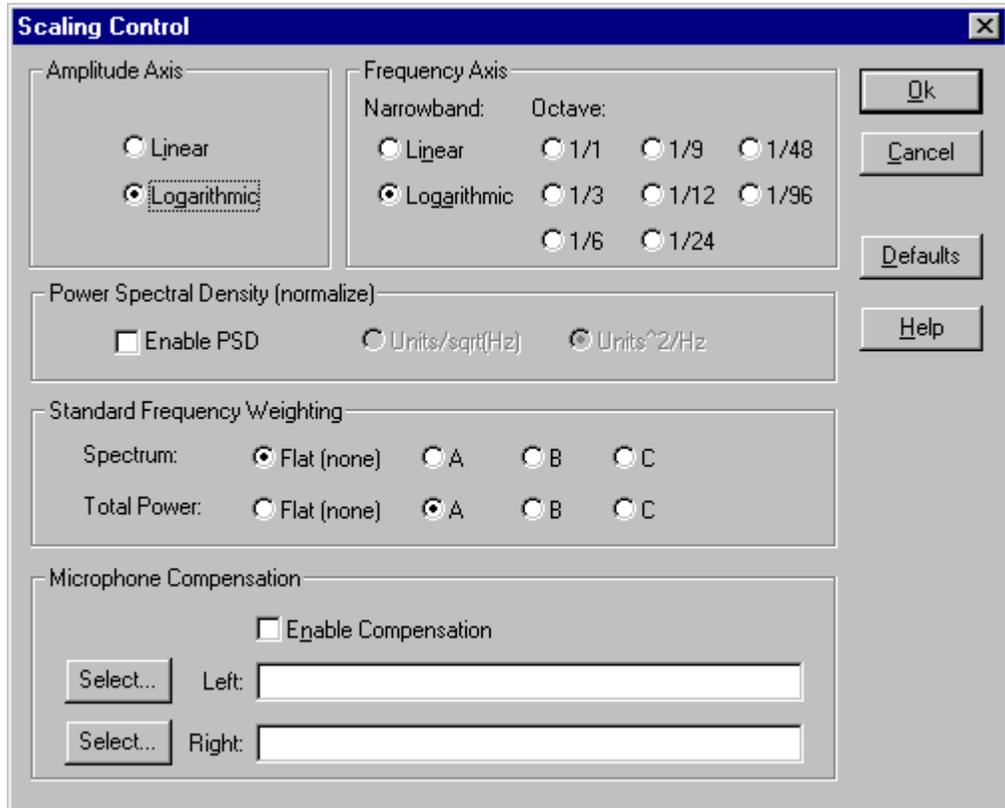
You can control the scaling method for both the frequency and amplitude axis. These selections apply to all displays except the Time Series display and the Y axis on the Phase display.

Amplitude Axis Scaling

Two basic amplitude scaling types are linear and logarithmic. They are related by the following equations:

Linear value = 10 ^ (Log value/20)

Log value = 20 * Log10 (Linear value)

Linear amplitude scaling makes the largest components in a spectrum very easy to see and to evaluate, but very small components may be overlooked completely. Linear scaling may be adequate in cases where the components are all about the same size. Logarithmic amplitude scaling clearly shows the low level signals in the presence of stronger signals. Logarithmic scaling is the default.

Power Spectral Density

The Power Spectral Density option causes the analyzer to normalize the spectrum to a 1 Hz band. This is used for making noise power

measurements. For instance, when measuring a noise signal without the PSD option selected, the amplitude will change with FFT size and sampling rate. Using the PSD option, the resulting amplitude will be independent of FFT size and sampling rate. You can select between Units / Sqrt(Hz) or Units^2/Hz. Units^2/Hz is most often used. You should select Linear amplitude axis scaling and narrowband frequency axis scaling when using the PSD option.

Frequency Axis Scaling

There are two basic types of frequency axis scaling: Narrowband and Octave. Narrowband scaling can be displayed in either Linear or Logarithmic format; Octave scaling is always displayed in Logarithmic format.

Narrowband Scaling - the spectrum produced by an FFT algorithm is also known as a narrowband spectrum. Each spectral line is a constant bandwidth and typically quite narrow. This narrowband spectral data can be displayed on either a linear or logarithmic scale but the frequency resolution is determined by the sampling rate and FFT size and remains unchanged.

Octave Scaling - produces a set of bands with constant proportional bandwidth. You can select from a range of octave formats (1/1, 1/3, 1/6, 1/9, 1/12, 1/24 etc). 1/1 and 1/3 octave are the most commonly used formats. See the Octave Scaling topic for more details.

Octave Scaling

The octave is such an important frequency interval to the ear that so-called octave band analysis has been defined as a standard for acoustic analysis. This type of spectrum is called constant percentage band (CPB) because each frequency band is a constant percentage of its center frequency. In other words, the analysis bands become wider in proportion to their center frequencies.   1/3 octave spectra are frequently used in acoustical measurements - three 1/3 octave bands encompass one octave.

It can be argued that the frequency resolution in 1/3 octave band analysis is too poor to be of much use, but a major advantage of constant percentage band analysis is that a very wide frequency range can be displayed on a single graph and the frequency resolution at the lower frequencies can still be fairly narrow. The naturally occurring ambient spectra will be found to slope downwards at the highest frequencies, and a constant percentage band (CPB) spectrum of the same data will usually be more uniform in level over a broad frequency range. Pink noise slopes downward at 3dB per octave and produces a perfectly flat 1/3 Octave spectrum.

1/3 Octave measurements are widely used in the audio industry. For instance, professional equalizers utilize 1/3 Octave bands. Standard 1/3 Octave frequencies have been agreed upon by the international community. These are listed below and are the set used by SpectraPLUS.

Most 1/3 Octave analyzers are analog instruments and use analog filters for each band. SpectraPLUS is a digital analyzer and utilizes the FFT algorithm to compute the narrowband (linear) frequency spectrum. This linear frequency spectrum is then apportioned among the 1/3 Octave bands to produce the corresponding 1/3 Octave spectrum.

As you would expect, the FFT size directly determines the number of 1/3 Octave bands that can be computed. At the low frequencies, there are only a small number of narrowband bins which fall within the corresponding 1/3 Octave band limits. The number of 1/3 Octave bands produced for various FFT sizes are listed below:

FFT size	Number of 1/3 Octave Bands
32	3
64	6
128	10
256	13
512	16
1,024	19
2,048	22
4,096	25
8,192	28
16,384	31
32,768	34
65,536	36

The actual band frequencies are determined by the sampling rate and decimation ratio.

Standard Frequency Weighting

ANSI standard A, B and C weighting curves can be applied to the spectral data and/or to the total power value. The total power setting affects the values shown in the total power utility and on the total power bar along the right side of the spectrum view (if enabled).

Flat is equivalent to no weighting at all. Weighting functions emphasize frequencies in the 500 to 10,000 Hz range, which is the area of greatest sensitivity to the human ear.

Standard Frequency Weighting Functions
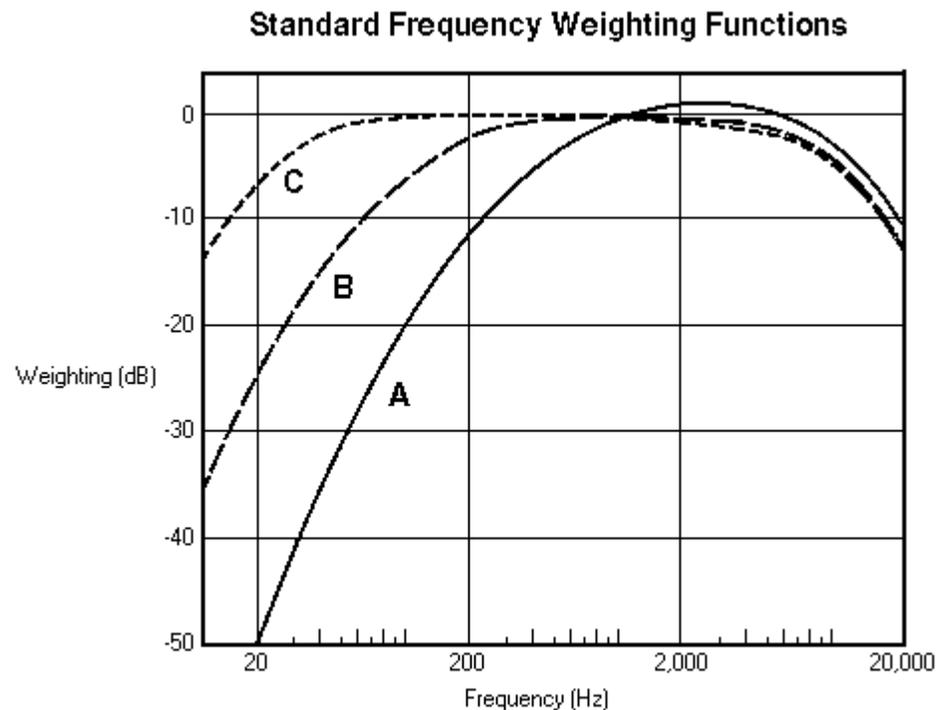
Microphone Compensation

An ideal microphone for acoustic measurements would have a perfectly flat frequency response (all frequencies detected with equal amplitude). In practice, most microphones do not meet this criteria. Microphone compensation curves solve this problem.

Any microphone manufacturer worth their salt provide a frequency response curve for their products. This curve shows both the total frequency range covered and the frequency response of the microphone within this range. By creating a simple text file containing this information, we can compensate for its response.

Building a microphone compensation file:
- Run the NOTEPAD.EXE application, spreadsheet or word processor that can produce a standard ASCII text file.
- Enter two columns of numbers representing the microphone response curve.
- The left column contains the frequency in Hertz.
- The right column contains the offset in decibels (this offset is subtracted from the logarithmic spectrum value).
- Enter the information in ascending frequency order.

- Separate the columns with a tab character.

Example:

100.0   -10.0
500.0   -5.0
1000.0   0.0
2000.0   2.0
10000.0   -5.0
15000.0   -15.0

Enter as many as 32768 pairs of numbers; at least 3 pair are required. SpectraPLUS will use a cubic spline algorithm to interpolate between these data points for each frequency bin in the spectrum. Any line with a semicolon in the first column will be treated as a comment line and ignored. By convention, microphone compensation files have an extension of .MIC.

Note:

Microphone Compensation files, Filter Response Files and Spectral Overlay files have identical formats and can be interchanged. This is useful if you want to verify that you have entered the microphone response table correctly. To do this, simply select the microphone compensation file as an overlay.

When manually creating an compensation file the results can have unexpected excursions due to the cubic spline interpolation algorithm. You can turn off this algorithm by adding a line to the beginning of the file as follows:
;SPLINE OFF

Markers

Markers are used to highlight specific frequencies of interest.
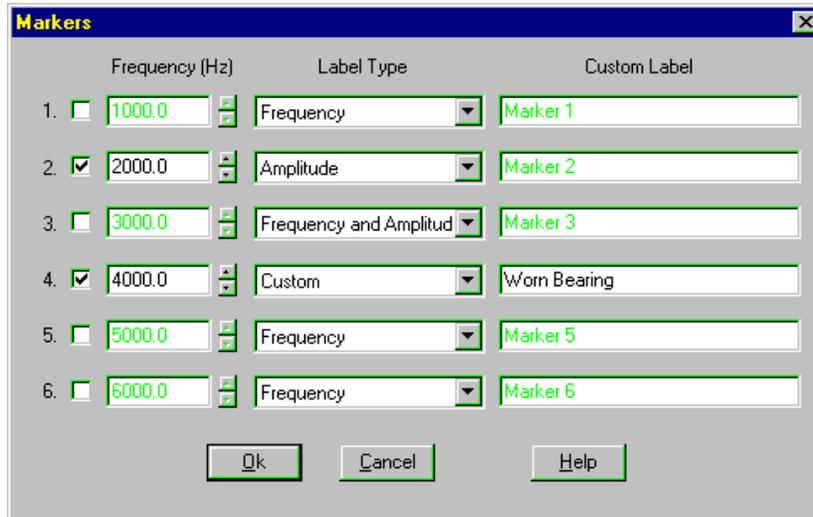
The selected frequencies are marked differently depending upon the view:

- Spectrum - marked frequency is shown with an arrow.
- Phase - marked frequency is shown with a diamond.
- Spectrogram - red arrow marks are shown along the frequency axis of the plot.
- 3-D Surface view - red arrows are displayed on the frequency axis.

Labels can be applied to the spectrum view. If custom labeling is selected, you may enter up to 30 characters of text.

Markers can also be set graphically by clicking the right mouse button in the spectrum view. This will bring up the Right Click Action Menu. Choosing one of the "Set Marker #" commands will set the marker frequency corresponding to the mouse click position. The "Clear All Markers" option will simply hide all markers; they can be reactivated by enabling them in the <Options><Markers> dialog box.

Marker frequencies can also be set from an external program sending a DDE command. Marker amplitudes can be transferred to an external program using the similar DDE command. See the DDE topic for more information.

Triggering

When triggering is enabled, the processor will test for a time series data value which exceeds the specified threshold before performing an FFT.
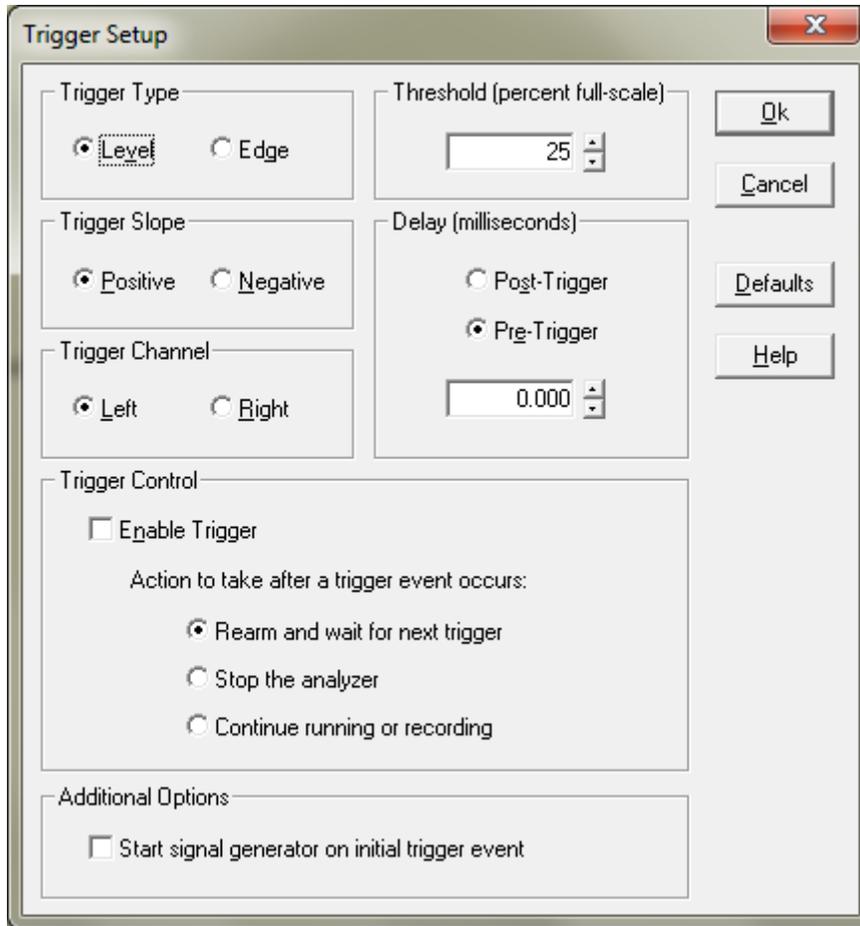
The triggering mode can be set to either level or edge:

- **Level triggering** - the analyzer will test the current data block for a time series data value which exceeds the specified threshold.
- **Edge triggering** - the analyzer will test the current data block for a sharp transition (difference in level between any two samples exceeds the threshold setting). This is useful for triggering on pulse type signals.

Trigger delay can be used to adjust the acquisition window within the FFT block. Pre-delay will move the trigger point to the right, post-delay to the left. If the delay is set to zero, the trigger point is positioned at the start of the FFT data block.

Action to take after a trigger event occurs:

Rearm and wait for next trigger - the analyzer will pause and continue to examine the incoming data stream for the presence of a trigger condition. This is the default behavior.

Stop the analyzer - the analyzer will stop after the first trigger has been detected and processed.

Continue running or recording - the analyzer will continue to run normally after the first trigger condition has been detected and processed. If the analyzer is stopped and restarted, it will once again wait for another trigger event.

Start signal generator on initial trigger event - this will delay the start of the signal generator{linkID=620} until the first trigger event is detected. The signal generator utility must be open when the analyzer is started for this option to take effect. Please note that due to operating system latencies the exact time between the trigger event and the start of the generator cannot be precisely controlled.

Triggering can be used in any operating mode but the behavior is slightly different for each:

- Real-Time - Each FFT block of input data is tested against the threshold. An FFT is performed if any sample value exceeds the threshold.
- Recorder - The recording/playback process examines one second chunks of data for the presence of a sample value which exceeds the specified threshold. If the threshold is exceeded, the entire one second of data is recorded/played.
- Post-Process - Each FFT block of wave data is tested and an FFT is performed if any sample value exceeds the trigger threshold.

Notes:

External triggering can be performed by utilizing one of the channels for data and the other for the trigger pulse. Select the appropriate trigger channel in the trigger dialog box and then setup the analyzer to only display the spectrum of the data channel.

Caution!

When triggering is enabled, the processor may appear not to work if the signal stays below the threshold. Check the status bar along the bottom edge of the program for the current mode.

Devices
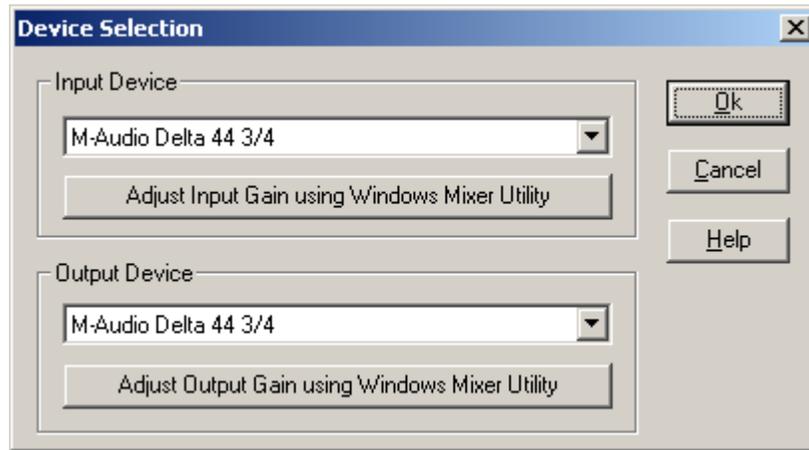
This allows you to select a specific sound card if more than one is installed in your system. The maximum sampling rate and precision are shown for the selected device.

The Input Device is used by the analyzer when Recording or running in the Real Time mode. The Output Device is used by the analyzer to Playback a .WAV file or to generate a test signal using the Signal Generator Utility.

Adjust Input/Output gain - this will launch the Mixer utility provided with the operating system. This allows you to adjust the sensitivity of the line in or mic inputs. The output gain settings control the playback/generator volume. Not all sound cards allow the gain/volume to be controlled.

Calibration

By default, the analyzer is calibrated to show the relative power levels where 0dB is the strongest possible 8 or 16 bit signal. The calibration option allows you to change the reference point to a known external value.

We will do this by connecting a known signal level to the input of the sound card and examining its corresponding digital level. With this information, we can compute the correct scale factors to calibrate the analyzer.

Select the <Options><Calibration> menu to display the calibration dialog box.

When the "Independent Scaling and Calibration" option is selected in the Settings dialog box, the scaling control dialog box will change to display separate parameters for each channel and each channel will be plotted in a separate window. This allows you to use two completely different types of scaling at once. For example you can configure the analyzer to show accelerometer data scaled in G's in one channel and 1/3 octave data scaled in SPL from a microphone in the other.

Procedure:

- Connect a steady state reference tone to the input of the sound card (do not use a DC level or noise signal).
- Enter the reference signal level in the edit control (you can use an oscilloscope or voltmeter to measure the signal level at the input to the sound card).
- Select "Peak" or "RMS" as appropriate (e.g. voltmeters produce RMS values).
- Select the applicable units for the input reference signal or enter custom labels.
- Select the desired Display mode: "Peak" or "RMS".

- Press the "Measure Input Signal" button. The analyzer will acquire 1 second of input data and compute the corresponding scale factors for each channel.

If you are calibrating to an accelerometer, enter the reference signal in G's (ft/sec^2). The "Convert Spectrum To:" field will activate and allow you optionally convert the spectrum to velocity or displacement. These conversions do not apply to the time series display.

You may customize the labels by directly modifying them in the edit box.

If the "Enable Calibration" button is checked, the scale factors will be applied to the displayed values and the amplitude axis labels will reflect the specified units.

You may save the calibration settings in a file and reload them as needed. The current calibration settings are always saved in configuration files and restored when loaded.

Note: The "Measure from Wave File" button will compute the calibration scale factors from an existing wave file. This is useful in cases where you only have a recording of the reference signal - please note that this recording must have been made with the same input gain for the results to be meaningful.

Very Important!
Most sound cards provide a mixer utility to allow you to set the input gain of the sound card. If the input gain is changed, the calibration is no longer valid. Many of these mixer utilities allow you to save the current gain settings for later recall. Unfortunately the multimedia API does provide any means for application programs such as SpectraPLUS to directly control the input gain level.

Chapter 9 - Utilities

Signal Generator
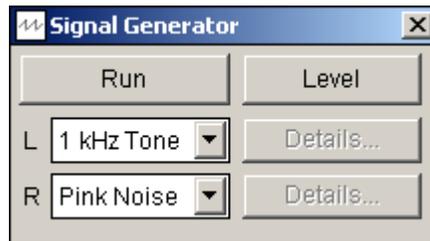
The signal generator utility operates in conjunction with the playback (D/A) channel of any Windows compatible sound card to produce a variety of test signals. Selecting the <Utilities><Signal Generator> menu command or pressing the F11 key will launch the signal generator.

Basic Operation

- Select the type of waveform you wish to generate by clicking on the list of waveforms.
- Click the "Details" button to modify the specified waveform (not all waveform types have user options).
- Click the "Level" button to specify the generator output level.
- Click the "Run" button to start the generator.
- When the generator is running, the window will change to show only a single "Stop" button.

Generator Output Level

Clicking on the "Level" button on the signal generator will allow you setup and control the generator output level.

In order to accurately control the generator output level we need to know the "full scale" signal level that your sound card can produce. This is the signal level produced when a 100% full scale sine wave is generated. Because most sound cards have the ability to control the output volume, "full scale" is not necessarily the same thing as the "maximum possible" signal level unless the volume control is set to its maximum value.

If you don't know this full scale level, you can still control the relative output level. This value can be entered in either dB or Percent Full Scale.

If you have a volt meter or scope handy, you can easily measure the full scale output level. Simply set the output level to 0 dB Full Scale and

generate a 1kHz tone. Now measure this value and enter the level in Volts RMS.

Some sound cards such as Digital Audio Labs CardD allow only fixed gains. We have included some predefined values for this and other selected cards.

Keep in mind that if your sound card has a volume control, you will need to reset the output level if the volume control is adjusted. Most sound cards save the volume settings between sessions.

Refer to the CardD user's guide for information on setting the jumpers.

Because of variations between individual sound cards, you may wish to measure the output level of your specific card for more accurate results.

When a single tone is generated, the output level is as specified above. When two tones are generated, the level will drop by 6dB so that clipping will not occur when the tones are in phase.

White Noise

A commonly used random noise waveform which has uniform amplitude for all frequencies. Its averaged spectrum is flat when viewed on a narrowband spectrum analyzer. The cutoff frequency is governed by the sampling rate used by the analyzer.

Pink Noise

A commonly used random noise waveform in acoustics. It has more energy at the lower frequencies. It's amplitude decreases at a rate of 3dB per octave. Its averaged spectrum is flat when viewed on an octave spectrum analyzer. The cutoff frequency is governed by the sampling rate used by the analyzer.

Noise Burst

This option allows you to generate a burst of noise. The "Burst Time" defines the length of the noise period. The "Cycle Time" specifies the repetition period. For instance, if the Burst Time is set to 1000 msec and the Cycle Time is set to 2000 msec, the generator will output 1 second of silence followed by 1 second of noise. This pattern will repeat until the generator is stopped.

1 kHz Tone

This 1,000.0 Hz sinewave test tone is widely used for distortion measurements.

Multiple Tones

This signal generator options allows you to configure your own set of tones. The tone waveform consists of up to 10 user defined tonals. You can select the number of active tones, their frequencies and relative amplitudes. The check box to the left of the frequency setting determines whether or not the tone is enabled. If the box is not checked, the associated tone will not be generated. The frequency setting determines the tone frequency in cycles per second. The relative power setting determines the amplitude of the tone compared to the other tones being generated. The power is specified in decibels (dB).

Tone Burst

This option allows you to generate a tone burst. The "Burst Time" defines the length of the tone period. The "Cycle Time" specifies the repetition period. For instance, if the Burst Time is set to 1000 msec and the Cycle Time is set to 2000 msec, the generator will output 1 second of silence followed by 1 second of tone signal. This pattern will repeat until the generator is stopped.

IMD Test Tones

InterModulation Distortion (IMD) testing requires two independent tones at different amplitudes. The amplitude of the higher of the two tones is typically 12dB (4:1 ratio) lower than the low frequency tone. Several standard tone pairs are provided as well as the ability to select your own custom tones.

Frequency Sweep (Swept Sine)

The swept sine waveform produces a sine wave which changes frequency over time. The start and stop frequencies determine the limits of the sweep signal. The start frequency can be less than the stop frequency in which case the sweep frequency will decrease over time.

The sweep type can be either Linear or Logarithmic. This determines the rate of change of the sweep frequency.

Jump Back to Start Frequency - when the stop frequency is reached the signal will immediately start again at the start frequency

Sweep Back to Start Frequency - when the stop frequency is reached the signal will sweep back to the start frequency.

Stop After a Single Sweep - the signal generator will stop after the sweep completes; otherwise it will repeat continuously until the generator is stopped.

Sweep Time - specifies how long the frequency takes to change from the start to the stop limits. This value is specified in milliseconds.

Swept sine waveforms are useful for frequency response testing. Set the averaging block size to 1 and use the Peak Hold option to capture the peak amplitude as the signal is swept across the frequency span.

Note: Sweeping too fast with large FFT sizes will "smear" the amplitude as the signal is changing frequency withing the underlying time series segment. The larger the FFT size used, the longer the required sweep time that should be used.

Frequency Step (Stepped Sine)

The frequency step waveform produces a sine wave which changes frequency over time in discrete steps.   The start and stop frequencies determine the limits of the signal. The start frequency can be less than the stop frequency in which case the frequency will decrease over time.

The step size can be either Linear or Octave (logarithmic).

Linear Steps - specify the step size in Hz. This works best when narrowband scaling is used on the analzyer.

Octave Steps - specify the fractional octave steps size to use. This works best when the octave scaling is used on the analyzer.

Jump Back to Start Frequency - when the stop frequency is reached the signal will immediately start again at the start frequency

Sweep Back to Start Frequency - when the stop frequency is reached the signal will step back to the start frequency.

Stop After a Single Pass - the signal generator will stop after the sequence completes; otherwise it will repeat continuously until the generator is stopped.

Dwell Time - specifies how long the signal stays at each step in the sequence. This value is specified in milliseconds.

Stepped sine waveforms are useful for frequency response testing. Set the averaging block size to 1 and use the Peak Hold option to capture the peak amplitude as the signal is stepped across the frequency span.

Level Sweep

The level sweep waveform produces a sine wave which changes amplitude over time. You select the start and stop levels in decibels, the tone frequency, and sweep time. The start level can be less than the stop level in which case the signal level will decrease over time.

The sweep time specifies how long the level takes to change from the start to the stop limits. This value is specified in milliseconds. The sweep will repeat continuously until the generator is stopped.

Sawtooth

The Sawtooth waveform produces a continuous train of ramps. The frequency and slope of the sawtooth can be adjusted by clicking on the "Details" button on the main screen. The Frequency setting specifies how many ramp cycles are generated each second. The Slope setting specifies whether the ramp rises or falls over time.

Triangular

The Triangular waveform produces a continuous train of triangular signals. The frequency of the triangular waveform can be adjusted by clicking on the "Details" button on the main screen. The Frequency setting specifies how many triangular cycles are generated each second.

Pulse

The Pulse waveform produces a single pulse. The width of the pulse and the repetition interval can be adjusted by clicking the "Details" button. If the "One Shot Pulse" option is checked, the generator will automatically stop after the pulse has been emitted. Press the "Run" button to generate another pulse.

Square

The Square waveform consists of a continuous series of pulses. The frequency and duty cycle of the square wave can be adjusted by clicking the "Details" button on the main screen. The Frequency setting determines the number of pulses generated each second. The Duty Cycle setting determines the pulse width as a percentage of the period.

User Defined

The user defined waveform option allows you to use a WAV file as the signal source. The signal will be looped continuously until you press the Stop button. Simply click on the "Pick" button to choose the .WAV file. Since the signal generator loads this file into memory, you must have sufficient RAM on your computer. SpectraPLUS allows you to build your own .WAV files from an ASCII text or binary file of sample data. See the <File><Import> topic for more details.

DTMF

Produces a series of dual tone pulses (touch tones) for the specified key string. You may also adjust the pulse width, spacing and twist. Twist is entered in dB. A positive twist value means that the amplitude of the higher frequency tone will be greater than the low frequency tone by this amount. Negative twist values produce the opposite effect. Please note that the full extended DTMF character set is supported (1234567890*#ABCD).

Digital Zero

Generates a signal containing all zero's or silence.

Notes

The accuracy of the signal is partly determined by the sound card itself. The accuracy of the sampling rate clock in the card will govern the accuracy of the frequency characteristics. In addition, many of the consumer grade sound cards will not generate DC components. These cards are "AC coupled" and as such, intentionally filter out or reject any DC component.

Professional grade sound cards are often capable of generating DC components (DC coupled) and will have no problems in this regard. The key specification governing this is the frequency response; DC coupled cards will specify a range from DC to 20kHz.

Save Signal To Wave File

This menu command creates a .WAV file from the current signal generator waveform. The signal generator must be activated before you can use this command. You will be given the option of modifying the length of the resulting wave file by repeating the signal multiple times; you will then be prompted for the filename that will contain the waveform data.

Lock Signal Generator Channels

This menu option locks the left and right channels so that the selected left channel signal is routed to both the left and right channels.

Peak Frequency

The peak frequency utility will display the frequency of the strongest spectral component in the entire span (not just what is displayed). This

utility window can be resized as required by the user and its contents will update anytime a new spectrum is computed. If the analyzer is configured to display the spectrum of both the left and right channels, this utility window will also show the value for both channels. The left channel value will appear above the right channel value.

Notes: The contents of any open utility window will be printed on the right margin of the spectrum view.

Peak Amplitude

The peak amplitude utility will display the amplitude of the strongest spectral component in the entire span (not just what is displayed). This utility window can be resized as required by the user and its contents will update anytime a new spectrum is computed. If the analyzer is configured to display the spectrum of both the left and right channels, this utility window will also show the value for both channels. The left channel value will appear above the right channel value.

Notes: The contents of any open utility window will be printed on the right margin of the spectrum view.

Total Power

The total power utility will display the total RMS power level for the entire spectrum and is equal to the total power bar on the right side of the spectrum view. The weighting (Flat, A, B or C) used for this utility is set under the Scaling options dialog box. This is equivalent to the readings on a Sound Level Meter with calibrated to units of SPL.

This utility window can be resized as required by the user and its contents will update anytime a new spectrum is computed. If the analyzer is configured to display the spectrum of both the left and right channels, this utility window will also show the value for both channels. The left channel value will appear above the right channel value.

Notes: The contents of any open utility window will be printed on the right margin of the spectrum view.

Total Harmonic Distortion

Real world audio devices introduce unwanted side effects called distortion in addition to the desired response. Total Harmonic Distortion (THD) and (THD+Noise) are two widely used measurements of this quantity. Both measurements are expressed as a percentage. The lower the THD value, the "cleaner" the response.

Total Harmonic Distortion (THD) - the ratio of the harmonic power to the fundamental power. It is computed by searching the entire spectrum to find the peak frequency (fundamental) and then calculating the total power in the harmonic frequencies. The THD level is then computed as the ratio of the total harmonic power to the fundamental power. Residual noise is not included in this calculation.

Total Harmonic Distortion + Noise (THD+N) - the ratio of the harmonic power plus noise to the fundamental power. It is computed by searching the entire spectrum to find the peak frequency (fundamental) and then calculating the total power in the remaining spectrum (harmonics plus noise). The THD level is then computed as the ratio of the total harmonic+noise power to the fundamental power.

THD+N values will almost always be greater than the THD values for the same signal.

A typical THD measurement will use a 1kHz reference test tone. This reference signal is passed through the audio device under test and its spectrum computed. It is often useful to employ long term averaging to cancel the random noise components and enhance the measurement accuracy.

It should be noted that the purity of both the sound card and the reference signal must exceed the device being measured if meaningful results are to be expected. The purity of the signal generator is directly dependent on the THD of the sound card used.

This utility window can be resized as required by the user and its contents will update anytime a new spectrum is computed. If the analyzer is configured to display the spectrum of both the left and right channels, this utility window will also show the value for both channels. The left channel value will appear above the right channel value

Notes:
- Hanning or Blackman smoothing window is recommended because of its low noise quality.

- An FFT size of 2048 or greater is recommended in order to provide adequate spectral resolution.
- The contents of any open utility window will be printed on the right margin of the spectrum view.

THD+N vs Frequency

The THD+N versus Frequency utility will generate a stepped frequency signal using the selected output device and compute and display a THD+N vs Frequency plot. This utility allows you to quickly and conveniently measure the distortion characteristics of your device over a range of frequencies.
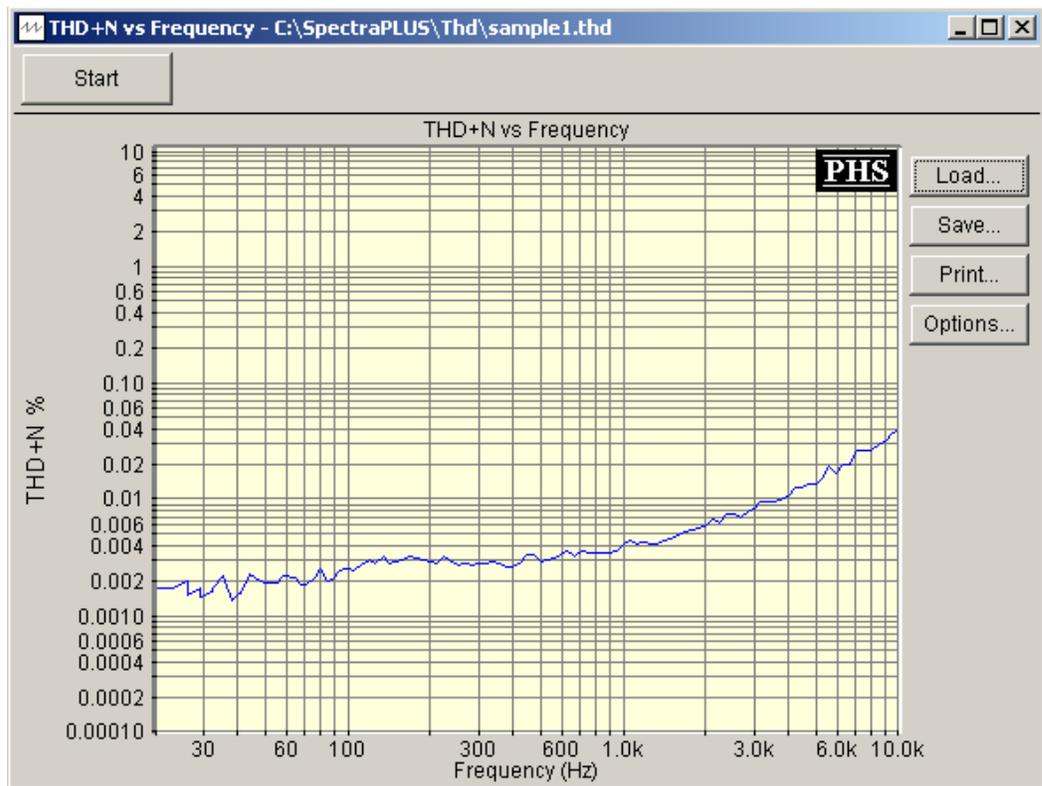
To start the utility click on the <Utilities><THD+N vs Frequency> menu or click on the "THD Freq" button on the toolbar. The utility will produce a stepped frequency sweep which provides the stimulus for the measurement. It must be connected from the output of the sound card to the device under test (DUT). The output of the device is then connected to the input of the sound card. During the test, no operator intervention is

required. The computer performs all the ncessary steps and plots the results across the audio band.

Once the results have been analyzed, the plot can be viewed, printed and stored to disk.   Right clicking on the plot will also popup a menu of options allowing you to place the data or plot image on the clipboard for use in other applications.

Options
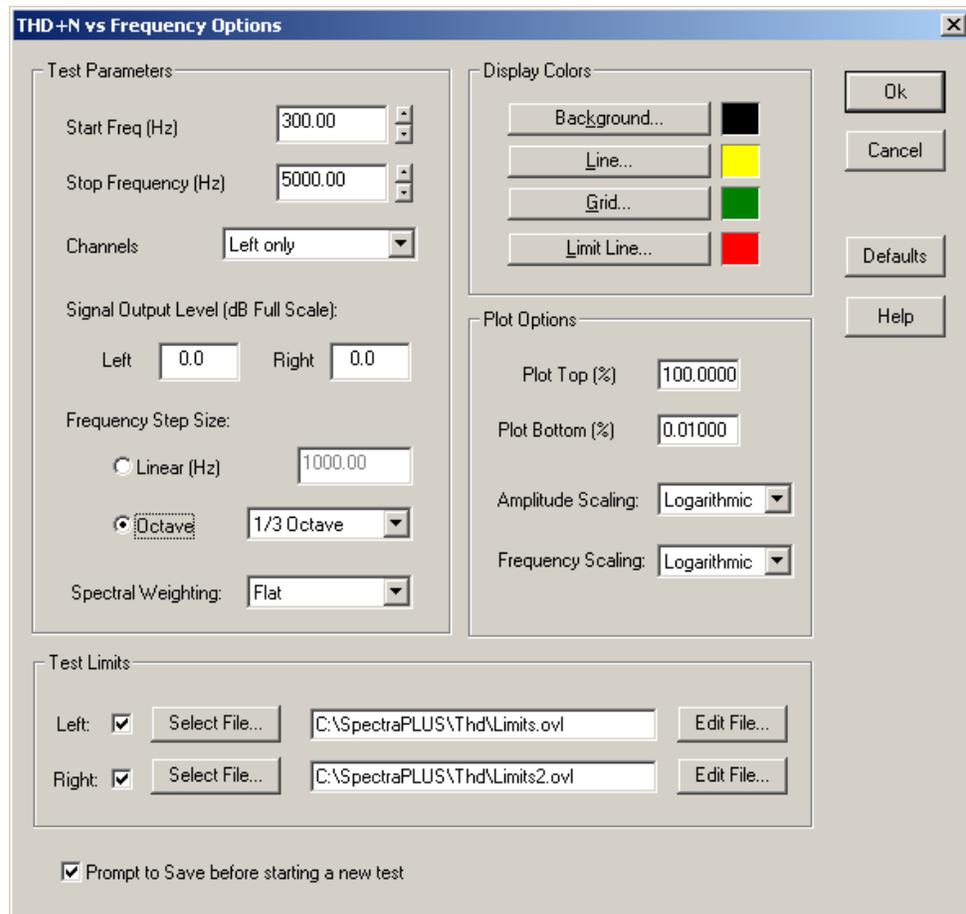
The THD+N vs Frequency options dialog box can be activated by clicking on the "Options" button along the right side of the plot area. This dialog box allows you to change the frequency span, display color scheme and a number of other useful options:

Start and Stop Frequencies - sets the frequency limits of the measurement. The stop frequency must be less than 1/2 of the current sampling rate used on the main analyzer settings.

Channels - select left only or both left and right.

Signal Output Level - sets the test signal output level in dB full scale. 0.0 dB is the signal level produced when a 100% full scale sine wave is generated. Because most sound cards have the ability to control the output volume, "full scale" is not necessarily the same thing as the "maximum possible" signal level unless the volume control is set to its maximum value.

Frequency Step Size - select a specific linear step size in Hz or octave scaling (1/1 thru 1/96). The stepped sine takes between 1 and 2 seconds per frequency step - decreasing the step size will increase the time required to perform the test.

Spectral Weighting – sets the spectral weighting used for the computations. See the topic on Scaling for more information on spectral weighting.

Plot Top - this sets the uppermost point for the amplitude axis. The value is entered as a percentage.

Plot Bottom - this sets the lowermost point for the amplitude axis. The value is entered as a percentage.

Frequency Axis Scaling - sets the scaling format for the frequency axis.

Amplitude Axis Scaling - sets the amplitude axis scaling format.

Test Limits - this lets you specify a file containing limits for each channel. The limit data will show as a separate trace. The limit file format is the same as the overlay and mic files and is a simple ASCII text file with two tab delimited columns of numbers; the first column contains the frequency and the second column contains the THD limit. The values must be in increasing order by frequency.

Prompt to Save – controls whether or not you will be prompted to save the file before starting a new test.

Notes:
You must use the internally generated stepped frequency signal as the source.

The only main analyzer settings that are used for this utility are:
        - Device selection

- Sampling Rate
- Sampling Precision

InterModulation Distortion (IMD)

InterModulation Distortion (IMD) is a measure of the distortion caused by the interaction (mixing) of two tones.

When multiple signals are injected into a device, undesired modulation or mixing of these two signals can occur. The IMD level is calculated by first computing the frequencies and amplitudes of the two strongest tones in the spectrum. The total power in each of the intermodulation product frequencies is then computed. IMD is the ratio of the intermodulation power to the rms sum of the tone power.

A typical IMD measurement will use tones at 250 Hz and 8020 Hz with the higher frequency tone reduced by 12dB (4:1 ratio) below the low frequency tone. It is often useful to employ long term averaging to cancel the random noise components and enhance the measurement accuracy.

This utility window can be resized as required by the user and its contents will update anytime a new spectrum is computed. If the analyzer is configured to display the spectrum of both the left and right channels, this utility window will also show the value for both channels. The left channel value will appear above the right channel value.

Notes:
- Hanning or Blackman smoothing window is recommended because of its low noise quality.
- An FFT size of 4096 or greater is recommended in order to provide adequate spectral resolution for the IMD measurements
- The contents of any open utility window will be printed on the right margin of the spectrum view.

Signal to Noise Ratio (SNR)

The Signal to Noise Ratio (SNR) is the ratio of the signal peak power level to the total noise level and is expressed in decibels (dB).

The SNR is computed by searching the entire spectrum to find the peak frequency and then calculating the total noise power in the remaining spectrum. The SNR is then computed as the ratio of the noise power to the peak power and expressed in decibels.

This utility window can be resized as required by the user and its contents will update anytime a new spectrum is computed. If the analyzer is configured to display the spectrum of both the left and right channels, this utility window will also show the value for both channels. The left channel value will appear above the right channel value.

Notes:
- Hanning or Blackman smoothing window is recommended because of its low noise quality.
- An FFT size of 2048 or greater is recommended in order to provide adequate spectral resolution.
- The contents of any open utility window will be printed on the right margin of the spectrum view.

Signal To Noise and Distortion (SINAD)

SINAD is a common sensitivity measurement normally associated with FM receivers. It is an acronym for Signal plus noise and distortion and is equal to (S+N)/N in our system. It is expressed in dB. If the signal is much stronger than the noise then SINAD value approaches the SNR value. Otherwise SINAD will be greater than the SNR.

This utility window can be resized as required by the user and its contents will update anytime a new spectrum is computed. If the analyzer is configured to display the spectrum of both the left and right channels, this utility window will also show the value for both channels. The left channel value will appear above the right channel value.

Notes:
- Hanning or Blackman smoothing window is recommended because of its low noise quality.
- An FFT size of 2048 or greater is recommended in order to provide adequate spectral resolution.
- The contents of any open utility window will be printed on the right margin of the spectrum view.

Noise Figure (NF)

The Noise Figure (NF) is a measure of the signal-to-noise degradation by a component under test. It is expressed in dB.

The Noise Figure is a two channel measurement similar to a Transfer Function where the input is compared with the output. In the case of the Noise Figure, it is the difference between the SNR at the output and the SNR at the input of the device. An ideal device has a Noise Figure of 0 dB; practical devices always have a larger Noise Figure than this.

A sinusoidal input test signal such as a 1kHz tone should be used as the source. The Noise Figure can be setup as either "Left vs Right" or "Right vs Left". If the input signal is on the right channel and the output on the left, you should use "Left vs Right" otherwise the Noise Figure will be negative.

This utility will only be available when the analyzer is setup for dual channel operation (stereo) under the <Options><Settings> dialog box.

This utility window can be resized as required by the user and its contents will update anytime a new spectrum is computed.

Notes:
- Hanning or Blackman smoothing window is recommended because of its low noise quality.
- An FFT size of 2048 or greater is recommended in order to provide adequate spectral resolution.
- The contents of any open utility window will be printed on the right margin of the spectrum view.

Delay Finder

The delay finder utility will compute the delay between the left and right channels. This delay can be displayed in milliseconds, feet or meters. Pressing the button labeled "Apply to Processing Delay" will then use this computed delay value as the new cross channel processing delay. This processing delay is the same value as displayed in the <Options><Settings> dialog box.

Positive values mean that the left channel is delayed relative to the right channel.
Negative values mean that the right channel is delayed relative to the left channel.
The cross channel processing delay setting does not influence the value computed with the delay finder.

Options

Pressing the "Options" button on the delay finder utility panel will bring up additional options:

Display Delay Values In - You can choose to display the delay value in milliseconds, feet or meters.

Speed of Sound - The speed of sound is used to convert the delay value between milliseconds and feet or meters. Because the speed of sound varies with temperature and pressure, you may want to customize this value to meet your conditions. Typically the speed of sound at room temperature is 1130 ft/sec - this is the default value used by the program.

You can measure the actual speed of sound by making an initial measurement in milliseconds (T1) and then moving the microphone exactly one foot away and taking a second measurement (T2). The speed of sound can then be computed as follows:

$$\text{Speed of Sound (ft/sec)} = 1000 / (T2 - T1)$$

Maximum Delay - This parameter controls the maximum resolvable delay. Longer delays take more processing time.

Typical Operation
- Connect a microphone to the left input channel.
- Connect a sample of the mixer output signal to the right input channel.
- Play a pink or white noise signal though the mixer.
- Run the analyzer to start the delay finder.
- Allow the measurement to settle for a moment. The delay value should remain relatively constant.
- Press the "Apply" button to use this delay value as the current processing setting.

Notes:

The delay finder will only operate in the Stereo mode; it will shut itself down if you are running in the Mono mode.
Program material (such as music) can be used instead of white or pink noise; however best results are obtained with a white or pink noise source.

Reverberation Time (RT-60)

The Reverberation Time utility allows you to measure and analyze the reverberation decay time and spectral characteristics of an acoustic environment. Reverberation is the sound you hear after an initial burst of energy. This measurement, called RT-60 is a standard method of characterization for concert halls and auditoriums and is the time it takes the sound to decay to 60 dB below its original level.

The Reverberation utility includes an internal pink noise generator which provides the stimulus for the RT-60 measurement. It must be connected from the output of the sound card through a power amplifier to a high-quality loudspeaker in the room. Typically only one loudspeaker is used for the test (rather than an array of speakers). The amplifier output level should be adjusted so that it is loud enough to "saturate" the environment because the louder the initial sound, the more accurate the resulting RT-60 curves; however, the level should not be so high as to clip the power amplifier or produce distortion from the loudspeakers.

During the test, no operator intervention is required. The computer performs all the ncessary steps to produce a family of decay curves across the audio band. First it activates the noise source long enough to saturate the environment (that is, an equilibrium is reached between direct and reverberant sound throughout the room). Then the sound is turned off and the decay of the signal captured and the reverberation response analyzed in detail.

To use an signal source other than the internal pink noise such as a ballon pop or gunshot follow the steps below:

1) Use the Recorder mode to capture the signal and subsequent decay.

2) Examine the waveform using the Time Series view and graphically select the signal and the decay; to do this click the selection arrow and then click and drag the mouse across a range of time. When the mouse button is released, the selected time segment will be shown with inverse video.

3) Right click on the plot and select the "Compute and Display RT-60" menu item.

4) The Reverb Time utility will open and the results computed and displayed.

5) You can fine tune the time at which the utility will start the decay time calculation by changing the value in the Start Offset control on the toolbar or by right clicking on the decay plot at the desired time offset and selecting the "Set Decay Start Time" menu. This will recompute the results based on the new decay start point.   The offset can also be set in the reverberation options dialog box.

Once the reverberation signal has been captured and analyzed, a series of plots can be viewed, printed and stored to disk.   Right clicking on the plot will also popup a menu of options allowing you to place the data or plot image on the clipboard for use in other applications.

Reverberation Plots

Reverb Times – this is a bar graph of reverberation time versus frequency band. 1/3 Octave bands from 80 Hz to 20 kHz are available. In addition a "wideband" or broadband value is shown on the right side. The analyzer defaults to displaying the RT-60 decay time; however, you can change this to display the decay time for RT-10 through RT-60 by using the associated control on the toolbar.   In addition double clicking on one of the bars will show the Reverberation Decay plot for the selected frequency band.
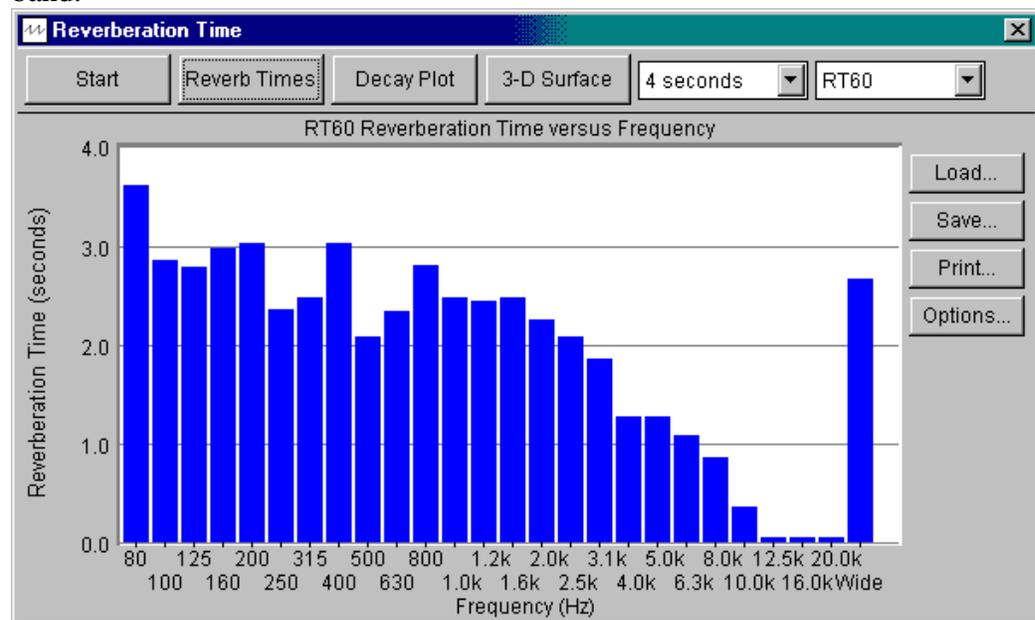
Reverb Decay – this is a graph of the reverberation decay versus time for a specific frequency band. You can change the selected frequency band using the associated control on the toolbar and you can change the displayed time limit by using the associated control on the toolbar; this allows you to zoom in to view the early decay time for instance.
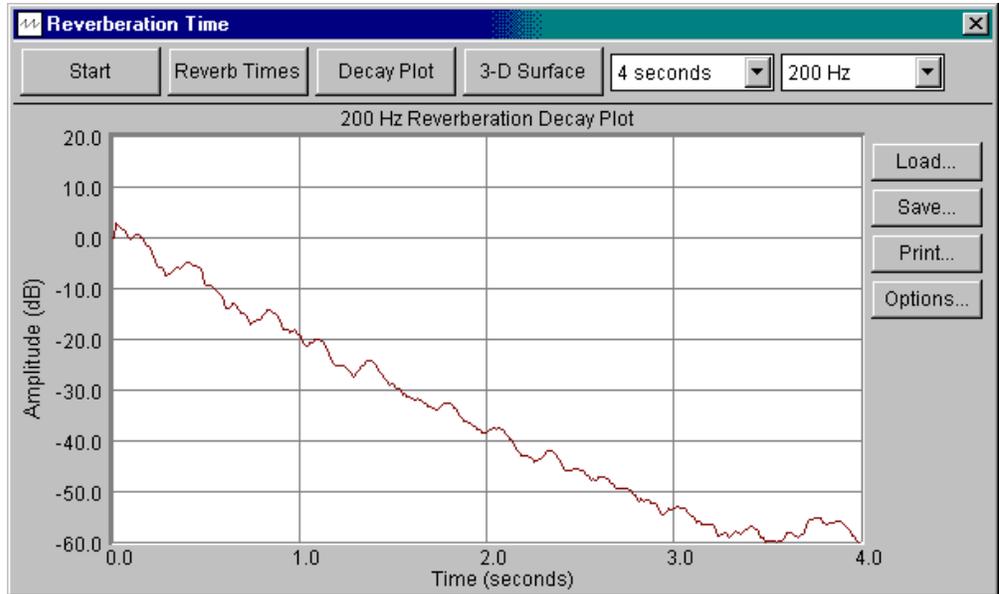
3-D Surface – this is a 3 dimensional surface plot of the reverberation decay; it shows the reverberation response for all frequencies in a single plot. Double clicking on the plot will display the decay plot for that particular band. You can also adjust the displayed time limit by using the associated control on the toolbar.
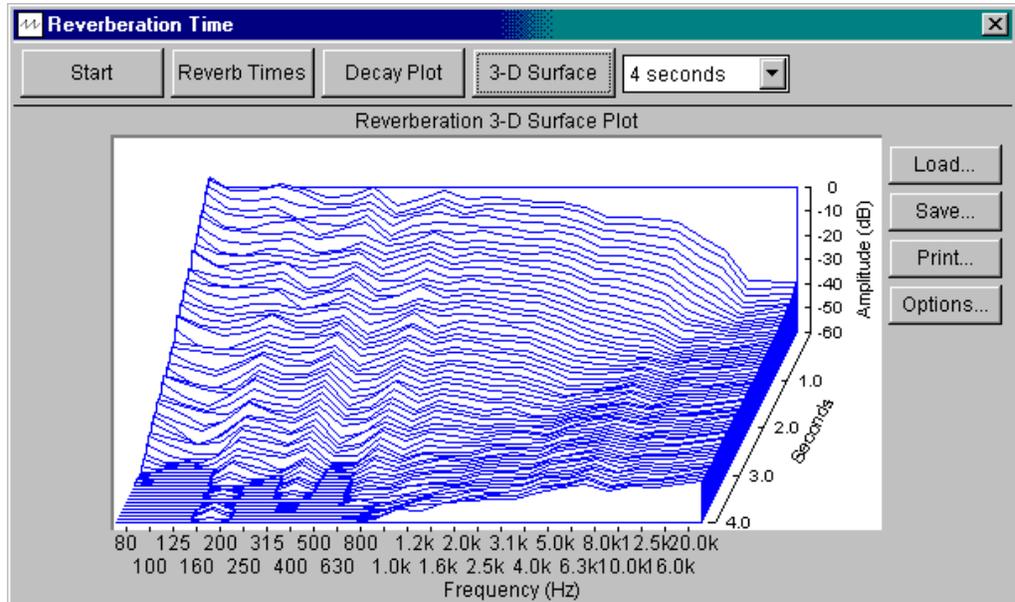
Reverberation Options
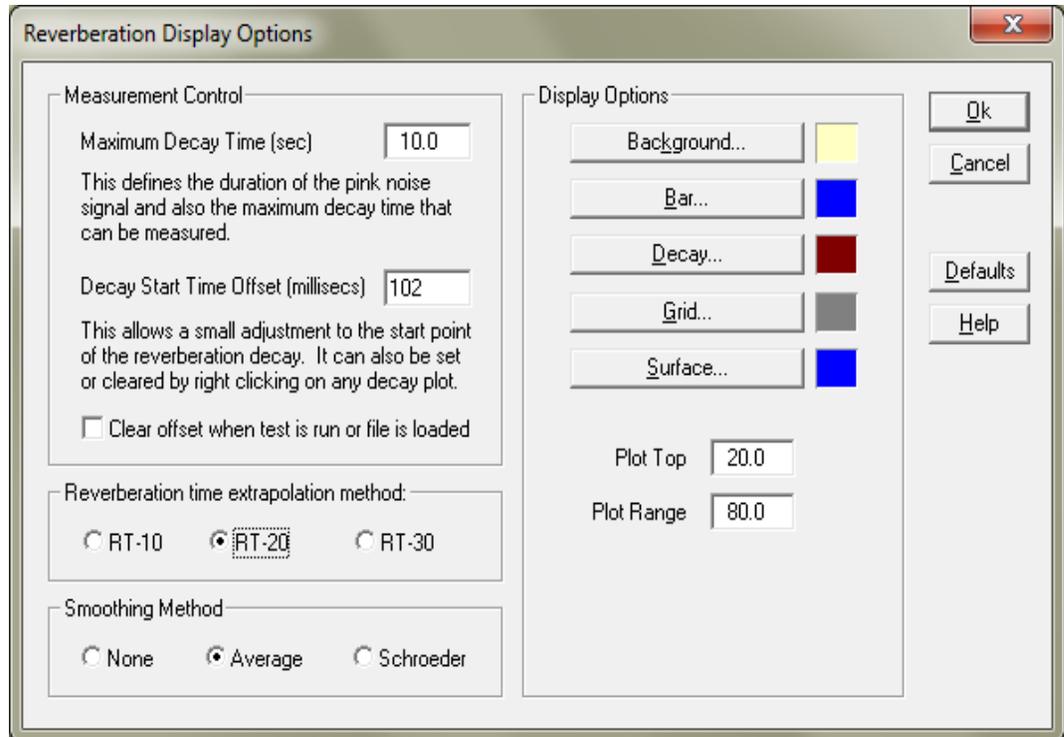
The reverberation options dialog box can be activated by clicking on the "Options" button along the right side of the plot area. This dialog box allows you to can change the color scheme used for the various plots as well as a number of other useful options:

Maximum Decay Time – this controls how long the analyzer will "listen" for the reverberation decay signal. This value should be set so that it is greater than the maximum expected decay time (you can always zoom in to display a shorter time period). Setting it to a longer time than necessary will not cause a problem but will require additional processing time and also increase the reverb file size.

Decay Start Time Offset - this allows you to make small adjustments to the initial start point for the decay calculation. Some technicians prefer to ignore the first 100 milliseconds or so of the initial decay. This value can be set directly using the associated toolbar control, in the reverb options dialog box or by right clicking on one of the Decay plots at the desired time offset and selecting the menu command "Set Decay Start Time". If the offset is non zero you can right click at any time and select "Clear Decay Start Time Offset" to reset the offset back to zero.

Clear offset when test is run or file is loaded - automatically sets the decay start time delay to zero whenever a new test is run or existing reverb file is loaded.

Extrapolation Method – in a typical acoustic test, it is rare that the difference between the pink noise level and the ambient background level is 60dB or more and because of this we must extrapolate the actual RT-60 time from the earlier decay times. This allows you to change the extrapolation method between RT-10, RT-20, and RT-30. RT-20 is the default setting. If you change this setting, the reverb times will be recomputed and the Reverb Time plot updated.

Smoothing Method – because of the complex nature of reverberation, the response often contains both diffuse reverberation as well as strong reflection components from hard surfaces. The resulting decay plot may appear quite irregular and it is useful to employ smoothing techniques to achieve a better estimation of the reverberation time. You can disable smoothing by choosing "None" or choose between "Average" and "Schroeder". The Average method is a 100 msec moving window that is applied to the decay data. The Schroeder method applies a reverse integration to the decay data starting at the RT-60 time and working backwards in time. The analyzer defaults to using the Average method. If you change this setting, the decay data will be recomputed and the displays will be updated.

Plot Colors – adjust as desired.

Plot Top - this sets the uppermost point for the amplitude axis in both the Reverb Decay Plot and the 3-D Surface Plot.

Plot Range - this sets the overall range for the amplitude axis in both the Reverb Decay Plot and the 3-D Surface Plot

Typical Reverberation Times – the following plot shows the typical RT-60 times for various environments.

Typical reverberation times
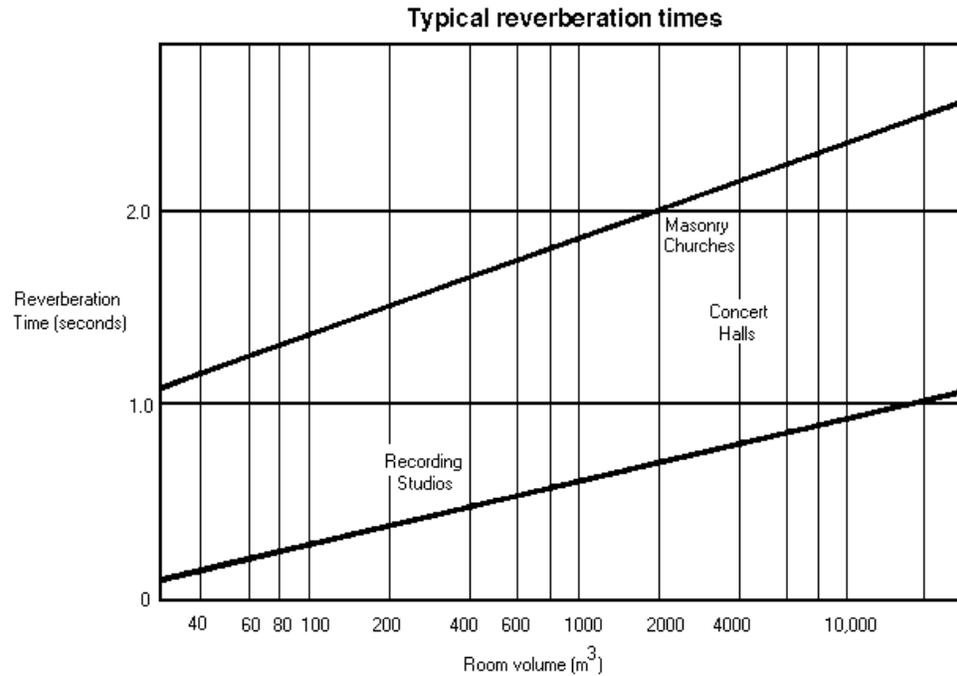
Notes:

The reverberation data is "normalized" so that the actual starting amplitudes are all adjusted to 0 dB. You may find that if you do not have sufficient signal amplitiude at certain frequencies, the decay curve for these frequencies will appear almost as a flat line with little or no decay.

The analysis method used in this utility has a 10 millisecond resolution. This resolution is not affected by the maximum decay time parameter or any other setting.

Equivalent Noise Level (LEQ)

The Equivalent Noise Levels are calculated as the analyzer runs.

Leq(t) - the equivalent noise level converted to the user specified "reference time period". For example, selecting a 3.6 second time segment of 90 dBA noise and specifying a reference time period of 1 hr (3600 sec) produces an equivalent noise level of 60 dBA.

Leq - the equivalent noise level for the selected time segment. Uses Forever SLM averaging.

Lsel - the equivalent noise level converted to a 1 second period.

Lpk - the peak level for the selected time segment. No averaging is used for this parameter.

Lmax - the maximum level for the selected data.

Lmin - the minimum level for the selected data.

L10 - the level exceeded 10% of the time.

L50 - the level exceeded 50% of the time.

L90 - the level exceeded 90% of the time.

Click the "Options" button to set the options for the calculations. You can select the averaging and weighting as needed.

Calculation Interval: determines the time period for recomputing the noise values.

Specific Intervals: Leq values will be recalculated at the specified interval
When the Analyzer Stops: Leq values will only be recalculated when the analyzer "Stop" button is pressed. All data contributes to the Leq values from when the analyzer starts to when it stops. (no values will be available until the analyzer stops).

Notes:
The current Calibration settings and Microphone Compensation settings (if enabled) are used for these calculations.

Macro Command Processor

This utility allows you easily to automate measurements, collect data, save files and output results using a simple programming language. The macro command processor uses the underlying DDE (Dynamic Data Exchange) syntax and capabilities but does not require a third party application such as Visual Basic. It provides more flexibility than the data logging utility but without the complexity of Visual Basic. This is not intended to be a comprehensive programming language - Visual Basic or other external application should be used if more complex control logic is needed.

The macro utility window contains a built in text editor; however, macro files are simple ASCII text files. The file size is limited to 32KB.

Quick Run Buttons
This optional toolbar allows you to define and label up to 6 buttons that will automatically launch an associated macro file. Very useful for quick test access.

Macro Control Statements:

1) **Send** - the Send statement must be followed by one of the DDE commands; it will send this DDE command to the analyzer. The square brackets are required . For example,

Send [Run]

will send the [Run] command to start the analyzer.

2) **Output** - the Output statement must be followed by one of the DDE request items; it will then write this item to the current macro output text file. For example,

Output Peak Frequency

will write the current peak frequency value to the output file.

3) **SetOutputFile** - this statement will set the name of the current output file; all subsequent Output statements will be written to this file until this statement is used again. For example,

SetOutputFile c:\temp\my_spectral_data.txt

If you do not use this command in your macro script, the Output data will be written to the default output filename as specified in the macro options dialog box.

4) **OutputComment** - this statement will simply write the text that follows it to the output file. For example,

OutputComment Starting Data Collection

will write the string "Starting Data Collection" to the output file.

5) **Wait** - this statement will cause the macro to wait until the specified time has elapsed before continuing. There are several variations of this command:

WaitSeconds 4

will wait for four seconds. The wait time can be a floating point number such as 0.25 (250 milliseconds).

WaitTopOfHour

will wait for the top of the next whole hour.

WaitClock 1415

will wait until the system clock reaces 1415 (14:15 or 2:15 PM). Leading zeros are required e.g. (0903)

WaitDate 12/04/2002

will wait until December 4, 2002. Leading zeros are required.

If you need to wait until a specific hour on a specific day simply use the WaitDate command followed by the WaitClock and WaitSeconds commands as needed.

6) **Loop, LoopEnd** - this statement pair allows you to iterate through a series of statements. For example,

Loop 10
   Send [Run]
   WaitSeconds 30
   Output Spectrum
   Send [Stop]
   WaitSeconds 3600
LoopEnd

will execute the commands between the two Loop statements 10 times; in this case it will run the analyzer for 30 seconds, output the spectral data, then stop and wait one hour. This will repeat 10 times.

Nested loops are not supported.

7) **Label, GoTo** - this statement pair allows you to jump to a specific labeled line in the macro file. For example,

Label StartPoint
.
. more macro statements...

.

GoTo StartPoint

The above statements will cause the macro command processor to jump to the line labelled "StartPoint".

GoTo CollectData

Label CollectData

.

. more macro statements

.

Return

This will cause the macro command processor to jump to the line labelled "CollectData". The "Return" statement will cause the macro to jump to the line just after the GoTo CollectData call. This allows you to implement simple functions. Up to 10 levels of nested GoTo/Return statements (functions) are supported

8) **;** - if the first character on a line is a semicolon, that line will be ignored. Use this for adding comments to your macro script.

9) **AutoDateTime** - if this command is embedded in any command that requires a filename, it will be expanded to the current date and time. For example, if the following command was used on Nov, 26 2002 10:20:11,

SetOutputFile c:\temp\SPL_data_AutoDateTime.txt

will set the output file to c:\temp\SPL_data_2002_11_26_102011.txt

This feature also works with the DDE commands that use a filename such as [File Save]. It also works with the OutputComment statement (e.g. OutputComment Starting at AutoDateTime).

10) **MacroEnd** - will cause the macro command processor to stop.

Notes:
- Leading or trailing spaces are ignored.
- Passing in a macro filename on the command line when starting the program will automatically start the macro. To do this create a shortcut to SpectraPLUS and edit the command line in the properties dialog for the shortcut. For example c:\specplus\bin\specplus.exe c:\specplus\macro\MyTest.mac

Data Logging

Data logging allows you to produce an output text file containing selected spectral parameters along with a timestamp showing when the event occurred.

Logging Options

Output Data Type:

Determines the contents to be written to the data file:

- Peak Frequency - writes the peak frequency within the specified frequency span.
- Peak Amplitude - writes the peak amplitude within the specified frequency span.
- Peak Frequency and Amplitude - writes both the peak frequency and amplitude within the specified frequency span.
- Marker Levels – writes the level of each enabled marker
- Total Power - writes the total RMS power level for the specified frequency span using the total power weighting selected in the scaling dialog box.

- Total Power Flat, A, B and C - writes the total RMS power level for the specified frequency span for all weighting options.
- Spectrum Values - writes all spectrum values for the specified frequency span.
- Total Harmonic Distortion (THD) - writes the THD value for the entire span.
- Total Harmonic Distortion + Noise (THD+N) - writes the THD+N value for the entire span.
- InterModulation Distortion (IMD) - writes the IMD value for the entire span.
- Signal To Noise Ratio (SNR) - writes the SNR value for the entire span.

Output Threshold: Determines the threshold above which the specified data type data will be output to the file; if no data within the specified frequency span exceeds the threshold, no data is output to the file. If you wish to output the data regardless of the threshold, simply set the threshold to -999 dB.

Logging Interval: Determines how often the spectral data is tested against the specified threshold. If the threshold is exceeded, a new line containing the selected output data is written to the data log file.

Frequency Span: Determines the frequency range over which data logging will occur. If "Full Span" is selected, logging is applied to the entire available frequency range. When "Custom Span" is selected, the frequency values can be directly entered. The custom span option is disabled whenever THD, THD+N, IMD or SNR values are selected as the output data type since these values are computed using the entire spectrum.

Output Filename: Output data file name. New data will be appended to the existing file.

Viewer Program: Select the name of the text editor you wish to use for viewing the data logging output files. The output data is always written in ASCII text. Although Notepad is the default viewer, it is unable to read files containing more than about 53KB of data. You may wish to select a word processor, MS-Write or other application if you will be creating large data files.

File Optionss: Select whether you want each logging session to overwrite the previous data or append new data to the end of the file. You can also control whether or not a header and timestamp are written to the file.

Notes:

Use caution when logging the entire spectrum values for large frequency ranges. The resulting data file will be very large.
Use the <Utilities><Data Logging><View> menu command to activate the text viewer program and read the contents of the current logging file.

Output example:

Data Logging On: Fri Sep 30 10:43:41 1995
Time   Freq(Hz)   Amplitude(dB)

10:43:41   527.56   -45.60
10:43:42   495.26   -54.03
10:43:43   495.26   -49.83
10:43:44   1399.66   -59.01
10:43:48   1399.66   -59.14
10:43:49   516.80   -56.38
10:43:50   549.10   -53.37
10:43:51   624.46   -43.66
10:43:52   559.86   -37.96

Chapter 10 - Configuration Files

The <Config> menu contains commands which allow you to save the current analyzer configuration to disk and load previously stored configurations from disk.

Configuration files contain all analyzer options as well as position and size information for each open view. The default file extension for configuration files is .CFG. The file is in same standard ASCII text format used for WIN.INI and SYSTEM.INI files.

Note:

If a configuration filename is passed as a parameter on the command line, it will automatically load when started.

Chapter 11 - Licensing

This software is copy protected using one of the following methods:

- A "Software Key" contained on the hard drive
- A "Hardware Key" which is plugged into the computer's USB or Parallel Port
- A "Network Hardware Key" which is plugged into this or another computer on the network

You can examine the licensing method and the license status at any time by selecting the <License><Status and Authorization> menu command. A list of all available options and their current license status will be displayed.

The program defaults to using the Software Key method; Hardware Keys can be purchased as a option for users who need to frequently transfer the license between computers or who simply prefer having a tangible physical key.

Software Key Method

This method uses a "Site Code" and "Authorization Key" technique which restricts the operation to a specific computer. No hardware key is required.

The "Site Code" is a number which is unique to your computer and is used to generate the matching "Authorization Key" to enable program operation. Because the Site Code is computer specific, we cannot provide you the corresponding Authorization Key until you send us your Site Code. This Site Code is easily obtained by selecting the <License><Status and Authorization> menu command clicking the "Authorize" button.

Click the "Authorize" button to display the "Site Code" unique to your computer. Click the "Print Site Code" button to print a page containing the site code which you can fax to your software vendor. When you receive your authorization key, return to this dialog box and enter the key. Please note that the Site Code will not change when you exit the program.

Adding additional options to the Software Key
Additional product options can be added to the Software Key in the field. Contact your software vendor to purchase the options or upgrade and provide them with the current "Site Code"; you will then be given a new

"Authorization Key". Go to the <License><Status & Authorization> menu and click the "Authorize" button and enter the Authorization Key; this will update the Software Key license information with the new options.

Transferring the Software Key to another computer
To transfer the Software Key, Go to the <License><Status & Authorization> menu and click the "Transfer Key" button and follow the instructions. There are two basic transfer methods: 1) directly over a local network connection, and 2) using a floppy diskette or USB memory stick. Both methods require that you first install a copy of SpectraPLUS on the computer to which will be receiving the license.   Please note that method 2) requires that the floppy disk or USB memory stick be "registered" using the UNLICENSED copy of SpectraPLUS. After registering, the Software Key can be transferred "Out" using the LICENSED copy of SpectraPLUS and then transferred "In" using the UNLICENSED copy.

Because transferring the Software Key requires access to both computers involved in the transfer, this is not always a practical solution. If you regularly transfer the license between computers, or if the computers are at different geographical locations, a Hardware Key may offer greater convenience. Please contact your software vendor to obtain a Hardware Key.

Hardware Key Method

This method uses a physical hardware device also known as a "dongle". This hardware key is delivered with the software and is preprogrammed with the options you have purchased.

- The Hardware Key can be either a USB or Parallel port type.
- You must install the Hardware Key device driver before the program will detect the key.
- The Hardware Key device driver is provided on the installation CD-ROM. You must login as administrator when installing the driver; once installed you can login using your normal user account.

Adding additional options to the Hardware Key
Additional product options can be added to the Hardware Key in the field. Contact your software vendor to purchase the options or upgrade; you will then be given a "Field Activation Key". Go to the <License><Status & Authorization> menu and click the "Authorize" button and enter the Activation Key; this will update the Hardware Key license information with the new options..

Network Hardware Key Method

This method also uses a USB or Parallel Port hardware key device but contains the ability to manage multiple licenses over the network. The Network Hardware Key allows multiple users to be running the software simultaneously and only one Network Hardware Key is required. The Key is preprogrammed with a user limit that, once reached, will not allow additional copies to run until other copies are shutdown. If a machine running SpectraPLUS crashes or is detached from the network, it will not have a chance to release its license; however, the license server will automatically release the license if communications are lost for more than about 5 minutes or so.

The Network Hardware Key can be installed on any machine on the local network but requires the installation of a "License Server" device driver. This "License Server" component is provided with the Network Key and only needs to be installed on the machine containing the Network Hardware Key. All other machines that will be running SpectraPLUS only need to have the normal Hardware Key device driver installed.

- The SpectraPLUS software must be installed on each local machine
- The standard Hardware Key driver must be installed on each local machine
- One of the machines in the network must have the Network Hardware Key installed and is designated the "license server".
- The "license server" does not have to be a true "server" but can simply be one of the computers in a peer to peer network.
- The "license server" hardware key device driver must be installed on the machine with the Network Hardware Key.

Additional network options allow you to specify a specific server and protocol but in most cases selecting the option labelled "Automatically find the license server on the network" should be used.

A Monitoring Tool is provided with the License Server to allow the administrator to check the number of copies in use.

Chapter 12 - Performance Issues

Processing Speed

The processing speed is affected by many factors. Foremost among these is the FFT size. Select a lower FFT size for faster performance but with reduced frequency resolution.

Video speed is a major factor. The more views that are displayed, the greater the demands placed on the video subsystem. The Spectrum plot is the fastest.

The 3-D Surface and Spectrogram views are significantly slower than the Time Series or Spectrum views. You may want to close these views if you are not using them. Alternatively they will run slightly faster if minimized to an icon along the bottom of the display.

Notes:

If only the Time Series plot is active, the system will not perform an FFT and will function much like a digital storage scope

Measurement Accuracy

Frequency Accuracy:

The frequency accuracy depends directly on the accuracy of the sampling clock on the sound card. This is typically a fraction of a Hertz (Hz). You may wish to check your sound card by using a known signal source for comparison.

Amplitude Accuracy:

By default, the analyzer is calibrated to show the relative power levels where 0dB is the strongest possible 8, 16 or 24 bit signal. It is possible to calibrate the amplitude axis by equating a known reference signal level with a measured sampling level; however, the frequency response of your sound card still comes into play (the frequency response of your sound card ultimately determines the ultimate accuracy of the measurement). It is possible to use the mic compensation feature to compensate for the frequency response of your sound card. By measuring the response of a known reference signal (e.g. white noise generator or swept sine measurement) you can determine the frequency response of the input

circuitry of your card. This can be placed on the clipboard using the <Edit> <Copy> command, pasted into a text editor and saved as a mic compensation file (you may want to use a spreadsheet to bias the values so that they center around 0 dB).

Octave Scaling Accuracy:
Historically, spectrum analyzers have utilized analog filters for each band and standards (ANSI S1.11-1986) have been developed which specify the performance of these filters . SpectraPLUS uses a Fast Fourier Transform (FFT) to compute the spectral data and then derives the octave data. This algorithm uses the same ISO center frequencies and bandwidths as analog instruments; however, the FFT filtering method produces much steeper "shoulders". These digital filters meet or exceed the performance of traditional analog filters.

Dynamic Range:
The theoretical dynamic range of the system is as follows:
- 8 bit sampling precision = 48 dB
- 16 bit sampling precision = 96 dB
- 24 bit sampling precision = 144 dB

The actual dynamic range achieved is dependent upon the performance of your sound card.

Notes:
Some older 8 bit sound cards utilize Automatic Gain Control (AGC). If this is the case, you will not be able to calibrate the amplitude axis.

Aliasing

Aliasing is a situation in which the signal being sampled contains frequencies which are greater than 1/2 the sampling rate. If this is the case, the signal will be detected by the analyzer but the frequency will be incorrect. For instance, if the sampling rate is 6,000 Hz and a 3,500 Hz signal is applied to the input, the analyzer will show the signal at 2,500Hz.

All sound cards provide low pass filtering to prevent aliasing; however, when decimation is used you should either use an external low pass filter or check the box labeled "Apply Low Pass Filter when Decimating".

Chapter 13 - Dynamic Data Exchange (DDE)

Dynamic Data Exchange (DDE) allows other applications to "talk" to SpectraPLUS; applications can send commands to the analyzer and request data in return. This powerful feature allows you to build custom signal processing solutions using SpectraPLUS.

Overview:

DDE is supported by all major spreadsheet programs, databases and programming languages. We recommend using Visual Basic for Applications (VBA) which Microsoft has incorporated into the latest versions of all their major applications. This provides a very flexible yet easy to learn macro language which is consistent across their entire product line.

DDE support consists of two major types of operations - Commands and Data Requests.

- Commands are actions messages such as "Run" and "Stop" which are sent to SpectraPLUS for control purposes.
- Data Requests are messages such as "THD" or "Spectrum" which cause SpectraPLUS to send the corresponding data values to the requesting application.

Establishing a DDE Hot Link automatically:

It is possible to set up what is known as a "hot link" wherein SpectraPLUS will send an "advise" message whenever it has computed new data values. When the application receives this advise message it should then request the corresponding data.   A hot link can be quickly established using the external application's <Paste Special> or <Paste><Link> menu command. For example, to establish a hot link with Excel, perform the following steps:

- Highlight the Spectrum View in SpectraPLUS.
- Click on the <Edit><Copy> menu command in SpectraPLUS.
- Go to Excel, highlight a destination cell and click on the <Edit><Paste Special> menu command.
- Click "Paste Link" as the source type and press Ok.
- At this point a "DDE hot link" will be established. Whenever the spectral data changes, the spreadsheet contents will update immediately. Any cells computed from these spectral values will be recomputed.

Establishing a DDE link manually:

Before an application can "talk" to SpectraPLUS it must first initiate what is known as a "DDE conversation". When initiating a conversation, the program must know the name of the DDE server and the topic. When requesting data, the name of the item must be known. All available Item names for SpectraPLUS are documented as follows:

DDE Server name

"Specplus"

Topic names Supported

"Data"

DDE Commands

Analyzer Control

[Run]	Starts the Analyzer
[Run ##]	Runs the Analyzer and stops after ## FFT's are performed
[Single Step]	Performs a single FFT and stops.
[Go To ##]	Go to the specified time within the WAV file (## = time in seconds)
[Go To Sample ##]	Go to the specified sample index within the WAV file (0 based index)
[Stop]	Stops the Analyzer
[Rewind]	Rewinds to the start of WAV file
[Fast Forward]	Jumps to the end of the WAV file
[Record]	Records new WAV data at current file position
[Pause Recorder]	Pauses recording but allows analyzer to continue running
[Resume Recorder]	Resumes a paused recording
[Play]	Plays WAV file from current position.
[Play Segment # &]	Plays WAV segment (#=start time in seconds, &=stop time in seconds)
[Apply Filter # &]	Apply filter to wave file (#=start time in seconds, &=stop time in seconds)

[Set Filter Channel #]	Set which channels to filter (0=Left only, 1=Right only, 2=Both left and right)

Modes

[Mode Real Time]	Changes to Real Time mode
[Mode Recorder]	Changes to Recorder mode
[Mode Post Processing]	Changes to Post Processing mode

File Operations

[File Open FILENAME.WAV]	Opens specified WAV file
[File Open FILENAME.CFG]	Opens specified configuration file
[File Open FILENAME.CAL]	Opens specified calibration file
[File Open FILENAME.MAC]	Opens and runs the specified macro
[File Open FILENAME.RVB]	Opens the specified reverb file.
[File Open FILENAME.THD]	Opens the specified thd vs freq file
[File Open FILENAME.FLT]	Sets the specified filter response file as the custom filter file
[File Save FILENAME.WAV]	Saves current WAV file with specified filename
[File Decimate FILENAME.WAV RATIO]	Decimates and saves WAV file with specified filename and decimation ratio
[File Merge LEFT.WAV RIGHT.WAV]	Merges two mono WAV files into a single Stereo WAV file
[File Close]	Closes .WAV file
[File Delete #]	Deletes the specified filename (use caution - cannot be undone)

ClipboardOperations

[Cut Time Segment # &]	Cut time segment to clipboard (#=start time in seconds, &=stop time in seconds)
[Copy Time Segment # &]	Copy time segment to clipboard (#=start time in seconds, &=stop time in seconds)
[Paste Time Segment #]	Paste time segment from clipboard (#=paste time in seconds)
[Mute Time Segment # &]	Mute time segment (#=start time in seconds, &=stop time in seconds)

Settings

[Set Sampling Rate #]	Sets the sampling rate where # is the value in Hertz

[Set FFT Size #]	Sets the FFT size
[Set Decimation Ratio #]	Sets the decimation ratio
[Set Channel #]	Sets the channel processing where # is one of the following: Left, Right, Both, Average, Coherence, Transfer LR, Transfer RL, Transfer LR+C, Transfer RL+C
[Set Delay #]	Sets the cross channel delay where # is the delay time in milliseconds
[Window #]	Sets the Smoothing Window type where # is one of the the window names (e.g. Blackman, Flat Top, Hamming).
[Set Average Mode #]	Sets the averaging type where # is Blocks or Meter.
[Set Average Type #]	Sets the averaging type where # is Exponential, Linear or Vector
[Set Average Size ##]	Sets the averaging block size or speed. If Block mode, ## = 1...1000; 1001 = Infinite. If Meter mode 0=Off, 1=Fast, 2=Medium, 3=Slow, 4=Forever
[Reset Average]	Resets running average
[Reset Overload]	Resets overload status
[Set Peak Hold #]	Set the Peak Hold Type where # is: 0=Off, 1=Fast, 2=Medium, 3=Slow, 4=Forever
[Set FFT Overlap ##]	Sets the FFT overlap percentage (## = 0...99) Post Processing mode only

Scaling

[Set Amplitude Scaling Type #]	Sets the amplitude axis scaling type where # is Linear or Log
[Set Left Amplitude Scaling Type #]	Sets the left channel amplitude axis scaling type where # is Linear or Log (independent channel scaling only)
[Set Right Amplitude Scaling Type #]	Sets the right channel amplitude axis scaling type where # is Linear or Log

	(independent channel scaling only)
[Set Frequency Scaling Type #]	Sets the frequency scaling axis type where # is Linear, Log or Octave
[Set Left Frequency Scaling Type #]	Sets the left channel frequency scaling axis type where # is Linear, Log or Octave (independent channel scaling only)
[Set Right Frequency Scaling Type #]	Sets the right channel frequency scaling axis type where # is Linear, Log or Octave (independent channel scaling only)
[Set Octave #]	Sets the Octave number where # is 1, 3, 6, 9, 12 etc.
[Set Left Octave #]	Sets the left channel Octave number where # is 1, 3, 6, 9, 12 etc. (independent channel scaling only)
[Set Right Octave #]	Sets the right channel Octave number where # is 1, 3, 6, 9, 12 etc. (independent channel scaling only)
[Set Spectrum Weighting #]	Sets the spectrum weighting where # is Flat, A, B or C
[Set Left Spectrum Weighting #]	Sets the left channel spectrum weighting where # is Flat, A, B or C independent channel scaling only)
[Set Right Spectrum Weighting #]	Sets the right channel weighting where # is Flat, A, B or C (independent channel scaling only)
[Set Total Power Weighting #]	Sets the Total Power weighting where # is Flat, A, B or C
[Set Left Total Power Weighting #]	Sets the left channel Total Power weighting where # is Flat, A, B or C independent channel scaling only)
[Set Right Total Power Weighting #]	Sets the right channel Total Power weighting where # is

	Flat, A, B or C (independent channel scaling only)
[Set PSD Type #]	Sets the PSD type where # is either 0 or 1. 0 is units^2/Hz 1 is units/sqrt(Hz)
[Set Left PSD Type #]	Sets the left channel PSD type where # is either 0 or 1. (independent channel scaling only) 0 is units^2/Hz 1 is units/sqrt(Hz)
[Set Right PSD Type #]	Sets the right channel PSD type where # is either 0 or 1. (independent channel scaling only) 0 is units^2/Hz 1 is units/sqrt(Hz)
[Enable PSD]	Enables PSD
[Enable Left PSD]	Enables PSD for the left channel (independent channel scaling only)
[Enable Right PSD]	Enables PSD for the right channel (independent channel scaling only)
[Disable PSD]	Disables PSD
[Disable Left PSD]	Disables PSD for the left channel (independent channel scaling only)
[Disable Right PSD]	Disables PSD for the right channel (independent channel scaling only)
[Set Mic Compensation File #]	Sets the microphone compensation filename to use for both channels
[Set Left Mic Compensation File #]	Sets the microphone compensation filename to use for the left channel
[Set Right Mic Compensation File #]	Sets the microphone compensation filename to use for the right channel
[Enable Mic Compensation]	Enables mic compensation
[Enable Left Mic Compensation]	Enables left channel mic compensation
[Enable Right Mic Compensation]	Enables right channel mic compensation
[Disable Mic Compensation]	Disables mic compensation

[Disable Left Mic Compensation]	Disables left channel mic ompensation
[Disable Right Mic Compensation]	Disables right channel mic compensation

Signal Generator

[Open Generator]	Opens the signal generator utility
[Close Generator]	Closes the signal generator utility
[Start Generator]	Starts the generator
[Stop Generator]	Stops the generator
[Set Generator Type Left #]	Set the left channel generator signal type where XYZ is the name of the signal type to use. Use any of the names shown in the generator's drop down list box.
[Set Generator Type Right #]	Set the right channel generator signal type where XYZ is the name of the signal type to use. Use any of the names shown in the generator's drop down list box.
[Set Generator Tone Left Freq AmpDB]	Sets the left channel generator type as "multiple tones" and configures it to produce a single tone at the specified frequency and relative amplitude. The amplitude value must be between 0.0 and -96 dB.
[Set Generator Tone Right Freq AmpDB]	Sets the right channel generator type as "multiple tones" and configures it to produce a single tone at the specified frequency and relative amplitude. The amplitude value must be between 0.0 and -96 dB.
[Set Generator Wave File Left #]	Sets the left channel generator type to "User Defined" and sets the wave filename to (# = filename)

[Set Generator Wave File Right #]   Sets the right channel generator type to "User Defined" and sets the wave filename to (# = filename)

 [Set Generator Level Left &]   Sets the left channel generator output level where & is the level value.

[Set Generator Level Right &]   Sets the right channel generator output level where & is the level value.

Triggering

[Trigger]   Forces trigger to occur
[Trigger On]   Enable Triggering
[Trigger Off]   Disable Triggering
[Trigger ReArm On]   Enables ReArm after trigger setting.

[Trigger ReArm Off]   Disables ReArm after trigger setting.

Markers

[Set Marker # xxx]   Sets frequency of marker # to XXX Hz.
[Show Marker #]   Shows marker #
[Hide Marker #]   Shows marker #

Overlays

[Set Overlay #]   Sets both left and right overlays from the current spectral data where # is the overlay number (1-7)

[Set Left Overlay #]   Sets the left channel overlay from the current spectral data where # is the overlay number (1-7)

[Set Right Overlay #]   Sets the right channel overlay from the current spectral data where # is the overlay number (1-7)

[Set Overlay From Peak Hold #]   Sets both left and right overlays from the current peak hold spectral data where # is the overlay number (1-7)

[Set Left Overlay From Peak Hold #]Sets the left channel overlay from the current peak hold spectral data where # is the overlay number (1-7)

[Set Right Overlay From Peak Hold #]   Sets the right channel overlay from the current peak hold spectral data where # is the overlay number (1-7)

[Show Overlay #]   Shows both left and right overlays where # is the overlay number (1-7)

[Show Left Overlay #]	Shows the left channel overlay where # is the overlay number (1-7)
[Show Right Overlay #]	Shows the right channel overlay where # is the overlay number (1-7)
[Hide Overlay #]	Hides both left and right overlays where # is the overlay number (1-7)
[Hide Left Overlay #]	Hides the left channel overlay where # is the overlay number (1-7)
[Hide Right Overlay #]	Hides the right channel overlay where # is the overlay number (1-7)
[Set Overlay Offset # &]	Set the offset for both channels where # is the overlay number and & is the offset in dB
[Set Left Overlay Offset # &]	Set the left channel offset where # is the overlay number and & is the offset in dB
[Set Right Overlay Offset # &]	Set the right channel offset where # is the overlay number and & is the offset in dB
[Set Overlay Legend # &]	Set the legend for both channels where # is the overlay number and & is the legend.
[Set Left Overlay Legend # &]	Set the left channel legend where # is the overlay number and & is the legend
[Set Right Overlay Legend # &]	Set the right channel legend where # is the overlay number and & is the legend
[Setup Composite Overlay # &]	Setup the composite overlay for both channels where # is the composite type (AVG or SUB). If AVG the & parameter is a list of channels to include in the average. For example [Setup Composite Overlay AVG 1 0 0 1 0 0] will include overlays 1 and 4 in the average. If SUB the & parameter is a list of which two channels are to be subtracted. For example [Setup Composite Overlay SUB 4 6] will subtract overlay 4 from 6
[Setup Left Composite Overlay # &]	Same as above but only for the left channel
[Setup Right Composite Overlay # &]	Same as above but only for the right channel
[Save Left Overlay # &]	Save the left overlay channel # to the specified filename &
[Save Right Overlay # &]	Save the right overlay channel # to the specified filename &
[Load Left Overlay # &]	Load the left overlay channel # from the specified filename &
[Load Right Overlay # &]	Load the right overlay channel # from the specified filename &

Configuration

[Save Config #]	Saves the calibration parameters in the specified filename.
[Load Config #]	Loads the calibration parameters from the specified filename

Calibration

[Calibrate]	Calibrates the analyzer to the input signal using the current calibration parameters.
[Save Calibration #]	Saves the calibration parameters in the specified filename.
[Load Calibration #]	Loads the calibration parameters from the specified filename
[Enable Calibration]	Enables calibration
[Disable Calibration]	Disables calibration

Device Selection

[Set Input Device #]	Sets the sound card device used for input where # is either the 1 based index into the device list or the exact device name
[Set Output Device #]	Sets the sound card device used for output where # is either the 1 based index into the device list or the exact device name

Views

[Open Time Series]	Opens the Time Series view
[Close Time Series]	Closes the Time Series view
[Maximize Time Series]	Maximizes the view in the overall application
[Restore Time Series]	Restores the view to its normal size
[Set Time Segment # ##]	Sets the start and stop time to display in the time series plot where # is the start time and ## is the stop time in seconds
[Set Time Series Plot Top #]	Sets the Plot Top parameter value
[Set Time Series Plot Range #]	Sets the Plot Range parameter value
[Autoscale Time Series]	Sets the plot top and range based on the current data
[Clear Spectrum]	Clears the current spectral data
[Clear Peak Hold]	Clears the current peak hold data
[Open Spectrum]	Opens the Spectrum view
[Close Spectrum]	Closes the Spectrum view

[Maximize Spectrum]	Maximizes the view in the overall application
[Restore Spectrum]	Restores the view to its normal size
[Spectrum Freq Span # ##]	Sets the start and stop frequency span to display in the Spectrum view where # is the start frequency and ## is the stop frequency in Hz
[Set Spectrum Plot Top #]	Sets the Plot Top parameter value
[Set Spectrum Plot Range #]	Sets the Plot Range parameter value
[Set Spectrum Bar Graph]	Sets display to bar graph type
[Set Spectrum Stepped Bar Graph]	Sets display to stepped bar graph type
[Set Spectrum Line Graph]	Sets display to line graph type
[Autoscale Spectrum]	Sets the plot top and range based on the current data
[Compute Average Spectrum # ##]	Computes and displays the average spectrum for the specified time segment where # is the start time and ## is the stop time in seconds
[Compute Reverb # ##]	Computes and displays the reverberation (RT-60) for the specified time segment where # is the start time and ## is the stop time in seconds
[Open Phase]	Opens the Phase view
[Close Phase]	Closes the Phase view
[Maximize Phase]	Maximizes the view in the overall application
[Restore Phase]	Restores the view to its normal size
[Phase Freq Span # ##]	Sets the start and stop frequency span to display in the Phase view where # is the start frequency and ## is the stop frequency in Hz
[Open Surface]	Opens the Surface view
[Close Surface]	Closes the Surface view
[Maximize Surface]	Maximizes the view in the overall application
[Restore Surface]	Restores the view to its normal size
[Surface Freq Span # ##]	Sets the start and stop frequency span to display in the 3D Surface view where # is the start frequency and ## is the stop frequency in Hz
[Build Surface # ##]	Builds and fills the Surface view with the spectum for the specified time segment

where # is the start time and ## is the stop time in seconds

[Open Spectrogram]	Opens the Spectrogram view
[Close Spectrogram]	Closes the Spectrogram view
[Maximize Spectrogram]	Maximizes the view in the overall application
[Restore Spectrogram]	Restores the view to its normal size
[Spectrogram Freq Span # ##]	Sets the start and stop frequency span to display in the Spectrogram view where # is the start frequency and ## is the stop frequency in Hz
[Build Spectrogram # ##]	Builds and fills the Spectrogram view with the spectum for the specified time segment where # is the start time and ## is the stop time in seconds
[Copy to Clipboard as Bitmap]	Places the contents of the currently active view on the Windows clipboard in a bitmap format

Utilities

[Open Peak Freq Utility]	Opens the Peak Frequency utility
[Close Peak Freq Utility]	Closes the Peak Frequency utility
[Open Peak Amp Utility]	Opens the Peak Amplitude utility
[Close Peak Amp Utility]	Closes the Peak Amplitude utility
[Open Total Power Utility]	Opens the Total Power utility
[Close Total Power Utility]	Closes the Total Power utility

[Open THD Utility]	Opens the THD utility
[Close THD Utility]	Closes the THD utility
[Open THD_N Utility]	Opens the THD+N utility
[Close THD_N Utility]	Closes the THD+N utility
[Open THDvsFreq]	Opens the THD+N vs Freq utility
[Close THDvsFreq]	Closes the THD+N vs Freq utility
[Start THDvsFreq]	Starts the THD+N vs Freq measurement
[Save THDvsFreq #]	Saves the THD+N vs Freq measurement results where # is the filename
[Print THDvsFreq]	Print the THD vs Frequency plot

[Open IMD Utility]	Opens the IMD utility
[Close IMD Utility]	Closes the IMD utility
[Open SNR Utility]	Opens the SNR utility

[Close SNR Utility]	Closes the SNR utility
[Open DelayFinder]	Opens the Delay Finder utility
[Close DelayFinder]	Closes the Delay Finder utility
[Apply DelayFinder]	Applys the current delay finder value to the processing settings.
[Open PhaseScope]	Opens the Phase Scope utility
[Close PhaseScope]	Closes the Phase Scope utility
[Open Macro]	Opens the macro command processor utility
[Close Macro]	Closes the macro command processor utility
[Open Macro File #]	Opens the specified macro filename
[Start Macro]	Runs the current macro
[Stop Macro]	Stops the current macro
[Open Reverb]	Opens the Reverberation utility
[Close Reverb]	Closes the Reverberation utility
[Start Reverb]	Starts the Reverberation measurement
[Save Reverb #]	Saves the Reverberation measurement results where # is the filename
[Set Reverb View Time]	Sets the Reverberation utility view to show the reverb times plot
[Set Reverb View Decay]	Sets the Reverberation utility view to show the reverb decay plot
[Set Reverb View Surface]	Sets the Reverberation utility view to show the 3-D Surface plot
[Enable DataLogging]	Enables data logging
[Disable DataLogging]	Disables data logging
[Set DataLogging File #]	Sets the name of the data logging output file

Printing

[Set Annotation Line1 XYZ]	Sets the text for annotation line 1
[Set Annotation Line2 XYZ]	Sets the text for annotation line 2
[Enable Comment Block]	Enable printing of comment block
[Disable Comment Block]	Disable printing of comment block
[Set Comment Line1 abc]	Sets the text for comment line 1
[Set Comment Line2 abc]	Sets the text for comment line 2
[Set Comment Line3 abc]	Sets the text for comment line 3
[Set Comment Line4 abc]	Sets the text for comment line 4
[Set Comment Line5 abc]	Sets the text for comment line 5
[Set Comment Line6 abc]	Sets the text for comment line 6
[Set Comment Line7 abc]	Sets the text for comment line 7
[Set Comment Line8 abc]	Sets the text for comment line 8

[Set Comment Line9 abc]	Sets the text for comment line 9
[Set Comment Line10 abc]	Sets the text for comment line 10
[Print Time Series]	Print the time series plot.
[Print Spectrum]	Print the spectrum plot.
[Print Phase]	Print the phase plot.
[Print Spectrogram]	Print the spectrogram plot.
[Print 3D Surface]	Print the 3D surface plot.
[Print THDvsFreq]	Print the THD vs Frequency plot

Window Control

[Minimize Window]	Shrink application to an icon
[Restore Window]	Restore application to normal position
[Maximize Window]	Expand application to full screen
[Cascade Windows]	Cascade all view windows
[Tile Windows Vertical]	Tile all views vertically
[Tile Windows Horizontal]	Tile all views horizontally
[Hide Window]	Hides the application from view
[Show Window]	Shows the application
[Hide Menu]	Hides the main application menu
[Show Menu]	Shows the main application menu
[Hide Toolbar]	Hides the main application toolbar
[Show Toolbar]	Shows the main application toolbar
[Hide Toolbar2]	Hides the secondary application toolbar
[Show Toolbar2]	Shows the secondary application toolbar
[Hide Statusbar]	Hides the main application lower statusbar
[Show Statusbar]	Shows the main application lower statusbar
[Hide Plot Toolbars]	Hides the toolbars of all open views
[Show Plot Toolbars]	Shows the toolbars of all open views
[Exit Application]	Exits the application

Special Functions

[Ambient Compensate]	Creates a mic compensation files AmbientLeft.mic and AmbientRight.mic from the most recent 1 second of data and uses it as the current mic compensation file.
[Ambient Compensate Enable	Enables the ambient compensation
[Ambient Compensate Disable]	Disables ambient compensation
[Set Peak Search Bandwidth # &]	Set the bandwidth to use for the Peak

Frequency and Peak Amplitude data request items (# = start frequency and & = stop frequency).

DDE Data Request Items

<u>Setttings</u>

"Mode"	'Returns the current mode: 1=Real time, 2=Post Processing, 3=Recorder
"FFT Size"	'Returns the FFT size
"Sampling Rate"	'Returns the Sampling Rate
"FFT Overlap" (0...99)	'Returns the overlap percentage
"Decimation Ratio"	'Returns the decimation ratio
"Average Mode"	'Returns the averaging mode: 0=Free Run (blocks), 1=Sound Level Meter (SLM)
"Average Type"	'Returns the averaging type: 0=Exponential, 1=Linear, 2=Vector.
"Average Size"	'Returns the averaging block size/speed (1...1000; 1001 if infinite) (0=off, 1=Fast, 2=Med, 3=Slow, 4=Forever).
"Peak Hold"	'Returns 0=Off, 1=Fast, 2=Medium, 3=Slow, 4=Forever
"Smoothing Window"	'Returns the name of the smoothing window being used

<u>Analyzer Status</u>

"Current Time"	'Returns the current WAV file position in seconds.
"Total Time"	'Returns the total WAV file time in seconds.
"Overload Status"	'Returns current overload status (1=true, 0=false)
"Overload Count"	'Returns count of overload events since last Run command.
"FFT Count"	'Returns count of FFT's performed since last Run command.
"Logging Status"	'Returns 1 if Data Logging is enabled, 0 otherwise
"Last Logging Time"	Returns the timestamp of the last data logging event. returns N/A if logging is off.
"Macro Status"	'Returns the run status of the macro command processor: 0=stopped, 1=running. -1=utility is not open.
"Signal Generator Type Left"	'Returns the left channel generator

	signal type
"Signal Generator Type Right"	'Returns the right channel generator signal type

Scaling

"Amplitude Axis Scaling Left"	'Returns the left channel amplitude axis scaling: 0=Linear, 1=Logarithmic
"Amplitude Axis Scaling Right"	'Returns the right channel amplitude axis scaling: 0=Linear, 1=Logarithmic
"Frequency Axis Scaling Left"	'Returns the left channel frequency axis scaling: 0=Linear, 1=Logarithmic, 2=Octave
"Frequency Axis Scaling Right"	'Returns the right channel frequency axis scaling: 0=Linear, 1=Logarithmic, 2=Octave
"Spectrum Weighting Left"	'Returns the left channel spectrum weighting type: 0=Flat, 1=A, 2=B, 3=C
"Spectrum Weighting Right"	'Returns the right channel spectrum weighting type: 0=Flat, 1=A, 2=B, 3=C
"Total Power Weighting Left"	'Returns the left channel total power weighting type: 0=Flat, 1=A, 2=B, 3=C
"Total Power Weighting Right" power	'Returns the right channel total weighting type: 0=Flat, 1=A, 2=B, 3=C
"PSD Left"	'Returns the left channel PSD status (0=off, 1=on)
"PSD Right"	'Returns the right channel PSD status (0=off, 1=on)
"PSD Type Left"	'Returns the left channel PSD type (0=units^2/Hz, 1=units/sqrt(Hz))
"PSD Type Right"	'Returns the right channel PSD status (0=units^2/Hz, 1=units/sqrt(Hz))
"Mic Compensation Left"	'Returns the left channel mic compensation status (0=off, 1=on)
"Mic Compensation Right"	'Returns the right channel mic compensation status (0=off, 1=on)
"Mic Compensation File Left"	'Returns the left channel mic compensation filename
"Mic Compensation File Right"	'Returns the right channel mic compensation filename

Peak Frequencies

"Peak Frequency"	'Returns the frequency of the peak bin (same as Peak1 Frequency)
"Peak1 Frequency"	'Returns the 1st highest peak frequency.
"Peak2 Frequency"	'Returns the 2nd highest peak frequency.
"Peak3 Frequency"	'Returns the 3rd highest peak frequency.
"Peak4 Frequency"	'Returns the 4th highest peak frequency.
"Peak5 Frequency"	'Returns the 5th highest peak frequency.
"Peak6 Frequency"	'Returns the 6th highest peak frequency.
"Marked Peak Frequency"	'Returns the peak frequency in the span between Marker1 and Marker2

Useful Note: Use the [Set Peak Search Bandwidth # &] command to limit the peak search bandwidth if needed

Peak Amplitudes

"Peak Amplitude"	'Returns the amplitude of the peak bin (same as Peak1 Amplitude)
"Peak1 Amplitude"	'Returns the 1st highest peak amplitude.
"Peak2 Amplitude"	'Returns the 2nd highest peak amplitude.
"Peak3 Amplitude"	'Returns the 3rd highest peak amplitude.
"Peak4 Amplitude"	'Returns the 4th highest peak amplitude.
"Peak5 Amplitude"	'Returns the 5th highest peak amplitude.
"Peak6 Amplitude"	'Returns the 6th highest peak amplitude.
"Marked Peak Amplitude"	'Returns the peak amplitude in the span between Marker1 and Marker2

Marker Values

"Marker1 Amplitude"	'Returns Marker 1 amplitude value
"Marker2 Amplitude"	'Returns Marker 2 amplitude value
"Marker3 Amplitude"	'Returns Marker 3 amplitude value
"Marker4 Amplitude"	'Returns Marker 4 amplitude value
"Marker5 Amplitude"	'Returns Marker 5 amplitude value
"Marker6 Amplitude"	'Returns Marker 6 amplitude value
"Marker7 Amplitude"	'Returns Marker 7 amplitude value
"Marker8 Amplitude"	'Returns Marker 8 amplitude value

Total Power

"Total Power"	'Returns the total power with default weighting
"Total Power A"	'Returns the total power with A weighting
"Total Power B"	'Returns the total power with B weighting
"Total Power C"	'Returns the total power with C weighting
"Total Power Flat"	'Returns the total power with flat weighting

"Marked Total Power"	'Returns the total power in the frequency span between Marker1 and Marker2 with default weighting
"Marked Total Power A"	'Returns the total power in the frequency span between Marker1 and Marker2 with A weighting
"Marked Total Power B"	'Returns the total power in the frequency span between Marker1 and Marker2 with B weighting
"Marked Total Power C"	'Returns the total power in the frequency span between Marker1 and Marker2 with C weighting
"Marked Total Power Flat"	'Returns the total power in the frequency span between Marker1 and Marker2 with Flat weighting

Distortion Values

"THD"	'Returns the THD value
"THD+N"	'Returns the THD+N value
"IMD"	'Returns the IMD value
"SNR"	'Returns the SNR value
"SINAD"	'Returns the SINAD value
"NF"	'Returns the Noise Figure (NF) value
"Delay Finder"	'Returns the delay value from the delay finder utility.
"THD vs Frequency"	Returns the THD+N vs Frequency values as an array**

"Compare Left THD vs Freq to Limits"	-2 if Limit data not available
"Compare Right THD vs Freq to Limits"	-1 if THD data not available
	0 if THD within limits
	1 if THD exceeds limits

<u>Spectral Data</u>

"Spectrum"	'Returns entire spectrum as array**
"Spectrum Left"	'Returns entire left channel spectrum as array**
"Spectrum Right"	'Returns entire right channel spectrum as array**
"Zoomed Spectrum"	'Returns spectrum range corresponding to the current zoom in/out settings on the view**
"Zoomed Spectrum Left"	'Returns left channel spectrum range corresponding to the current zoom in/out

	settings on the view**
"Zoomed Spectrum Right"	'Returns right channel spectrum range corresponding to the current zoom in/out settings on the view**
"Marked Spectrum"	'Returns spectrum range between Marker1 and Marker2 as array**
"Marked Spectrum Left"	'Returns left channel spectrum range between Marker1 and Marker2 as array**
"Marked Spectrum Right"	'Returns right channel spectrum range between Marker1 and Marker2 as array**
"Peak Hold Spectrum"	'Returns entire peak hold spectrum as array**
"Peak Hold Spectrum Left"	'Returns entire left channel peak hold spectrum as array**
"Peak Hold Spectrum Right"	'Returns entire right channel peak hold spectrum as array**
"Zoomed Peak Hold Spectrum"	'Returns peak hold spectrum range corresponding to the current zoom in/out settings on the view**
"Zoomed Peak Hold Spectrum Left"	'Returns left channel peak hold spectrum range corresponding to the current zoom in/out settings on the view**
"Zoomed Peak Hold Spectrum Right"	'Returns right channel peak hold spectrum range corresponding to the current zoom in/out settings on the view**
"Marked Peak Hold Spectrum"	'Returns peak hold spectrum range between Marker1 and Marker2 as array**
"Marked Peak Hold Spectrum Left"	'Returns left channel peak hold spectrum range between Marker1 and Marker2 as array**
"Marked Peak Hold Spectrum Right"	'Returns right channel peak hold spectrum range between Marker1 and Marker2 as array**

Comparing Spectral Data To Overlays

"Compare Left Spectrum To Overlay 1"	'Compares the left channel spectral data to Overlay 1. Frequency Span is between Markers 1 and 2.
"Compare Left Spectrum To Overlay 2"	'Compares the left channel

	spectral data to Overlay 2. Frequency Span is between Markers 3 and 4.
"Compare Left Spectrum To Overlay 3"	'Compares the left channel spectral data to Overlay 3. Frequency Span is between Markers 5 and 6.
"Compare Left Spectrum To Overlay 4"	'Compares the left channel spectral data to Overlay 4. Frequency Span is between Markers 7 and 8.
"Compare Left Peak Hold To Overlay 1"	'Compares the left channel peak hold data to Overlay 1. Frequency Span is between Markers 1 and 2.
"Compare Left Peak Hold To Overlay 2"	'Compares the left channel peak hold data to Overlay 2. Frequency Span is between Markers 3 and 4.
"Compare Left Peak Hold To Overlay 3"	'Compares the left channel peak hold data to Overlay 3. Frequency Span is between Markers 5 and 6.
"Compare Left Peak Hold To Overlay 4"	'Compares the left channel peak hold data to Overlay 4. Frequency Span is between Markers 7 and 8.
"Compare Right Spectrum To Overlay 1"	'Compares the right channel spectral data to Overlay 1. Frequency Span is between Markers 1 and 2.
"Compare Right Spectrum To Overlay 2"	'Compares the right channel spectral data to Overlay 2. Frequency Span is between Markers 3 and 4.
"Compare Right Spectrum To Overlay 3"	'Compares the right channel spectral data to Overlay 3.

	Frequency Span is between Markers 5 and 6.
"Compare Right Spectrum To Overlay 4"	'Compares the right channel spectral data to Overlay 4. Frequency Span is between Markers 7 and 8.
"Compare Right Peak Hold To Overlay 1"	'Compares the right channel peak hold data to Overlay 1. Frequency Span is between Markers 1 and 2.
"Compare Right Peak Hold To Overlay 2"	'Compares the right channel peak hold data to Overlay 2. Frequency Span is between Markers 3 and 4.
"Compare Right Peak Hold To Overlay 3"	'Compares the right channel peak hold data to Overlay 3. Frequency Span is between Markers 5 and 6.
"Compare Right Peak Hold To Overlay 4"	'Compares the right channel peak hold data to Overlay 4. Frequency Span is between Markers 7 and 8.

Return values for "Compare Spectrum To" function set:

    -2 = Spectral data not available
    -1 = Overlay data not available
     0 = Spectrum is completely above Overlay
     1 = Spectrum is completely below Overlay
     2 = Spectrum intersects Overlay (partially above and below)

Phase Data

"Phase"	'Returns entire phase as array**
"Phase Left"	'Returns entire left channel phase as array**
"Phase Right"	'Returns entire right channel phase as array**
"Zoomed Phase"	'Returns phase range corresponding to the current zoom in/out settings on the view**
"Zoomed Phase Left"	'Returns left channel phase range

	corresponding to the current zoom in/out settings on the view**
"Zoomed Phase Right"	'Returns right channel phase range corresponding to the current zoom in/out settings on the view**
"Marked Phase"	'Returns phase range between Marker1 and Marker2 as array**
"Marked Phase Left"	'Returns left channel phase range between Marker1 and Marker2 as array**
"Marked Phase Right" between	'Returns right channel phase range between Marker1 and Marker2 as array**

Time Series Data

"Time Series"	'Returns entire time series values as array**
"Time Series Left"	'Returns entire left channel time series values as array**
"Time Series Right"	'Returns entire right channel time series values as array**
"Zoomed Time Series"	'Returns time series values corresponding to the current zoom in/out settings on the view**
"Zoomed Time Series Left"	'Returns left channel time series values corresponding to the current zoom in/out settings on the view**
"Zoomed Time Series Right"	'Returns right channel time series values corresponding to the current zoom in/out settings on the view**
"Wave Segment"	'Returns the wave data corresponding to [Set Time Segment # ##] command**
"Wave Segment Left"	'Returns the left channel wave data corresponding to [Set Time Segment # ##] command**
"Wave Segment Right"	'Returns the right channel wave data corresponding to [Set Time Segment # ##] command**
"Wave Sample Left"	'Returns the left channel wave file value for the current position. Use [Go To Sample ##] to set position
"Wave Sample Right" for	'Returns the right channel wave file value for the current position. Use [Go To Sample ##] to set position

** Array format:

Data is returned in the same format used by the clipboard:

Spectrum
1st column = Frequency in Hz
2nd column = Amplitude in same units as current display.

Phase
1st column = Frequency in Hz
2nd column = Phase in degrees

Time Series
1st column = Time in seconds
2nd column = Amplitude in same units as current display.

DDE Examples

Microsoft Excel 7.0 DDE Module Example:
(Limit testing - runs THD test and checks against spec.)
Sub LimitTest()
'Start a DDE connection with SpectraPLUS
'You should already have opened the program.
ch = DDEInitiate("Specplus", "Data")
DDEExecute ch, "[File Open c:\spectraplus\config\thd_test.cfg]"
DDEExecute ch, "[Run]"

'wait 10 seconds
newHour = Hour(Now())
newMinute = Minute(Now())
newSecond = Second(Now()) + 10
newTime = TimeSerial(newHour, newMinute, newSecond)
Application.Wait newTime
DDEExecute ch, "[Stop]"

'retrieve the results from the analyzer
Data = DDERequest(ch, "THD")
thd_value = Data(1)

'Check result against specification
If thd_value < 0.05 Then
        MsgBox("Test PASSED")
Else
        MsgBox("Test FAILED")
End If

```
                DDETerminate ch
                End Sub


Microsoft Excel 7.0 DDE Module Example:
(Statistics - Acquires 20 sets of averaged 1/3 Octave data)
Sub ThirdOctaveTest()
'Start a DDE connection with SpectraPLUS
'You should already have opened the program.
ch = DDEInitiate("Specplus", "Data")

'Control variables.  Change these as needed.
MaxAverages = 20
AverageTimeMinutes = 1
CurrentAverage = 0

Do
        'Start the analyzer
        DDEExecute ch, "[Run]"

        'wait N minutes from now
        newHour = Hour(Now())
        newMinute = Minute(Now()) + AverageTimeMinutes
        newSecond = Second(Now())
        newTime = TimeSerial(newHour, newMinute, newSecond)
        Application.Wait newTime

        ' Stop the analyzer
        DDEExecute ch, "[Stop]"

        'Request the spectrum data from the analyzer
        'it will be placed in an array.
        DataArray = DDERequest(ch, "Spectrum")

        'Find the total number of bands in the spectrum.
        'We will have one row per band
        num_bands = UBound(DataArray)

        'Place the spectral data in the worksheet starting
        'from right to left.  'Since two columns of numbers
        'are sent to us (freq, amplitude) we will simply overwrite
        'the frequency column with the next column of amplitude data.
        'Statistical calculations can then be performed in Sheet1
                Worksheets("Sheet1").Range(Cells(2,MaxAverages-
CurrentAverage), Cells(1 + num_bands, MaxAverages - CurrentAverage
+ 1)).Formula = DataArray
```

```
            CurrentAverage = CurrentAverage + 1
            If CurrentAverage >= MaxAverages Then Exit Do
Loop

DDEExecute ch, "[Stop]"
DDETerminate ch
End Sub
```

Chapter 14 - Misc

Keyboard Shortcuts

The main toolbar contains icons for controlling the processor. Each icon's label has one of the letters underlined. Pressing the <Alt> key and the underlined letter will perform the same action as a button press.

In the same way, keyboard shortcuts are available for each menu item. Pressing the <Alt> key and the underlined letter in the menu and sub-menu item will allow quick access to the commands. For instance pressing <Alt> <V> followed by <Alt> <T> will toggle the Time Series view.

Additional shortcut keystrokes called "accelerator keys" are available for selected menu items:

Accelerator keys:

F1	Help
F4	Settings dialog box
F5	Scaling dialog box
F6	Trigger dialog box
F7	Save Configuration File
F8	Load Configuration File
F9	Calibration dialog box
F10	Total Harmonic Distortion Utility
F11	Signal Generator Utility
Ctrl+0	File Open
Ctrl+S	File Save
Ctrl+A	File Save As
Ctrl+P	File Print
Alt+F4	Exit SpectraPLUS
Ctrl+Z	Undo last edit
Ctrl+Del	Edit Cut
Ctrl+Ins	Edit Copy
Shift+Ins	Edit Paste Insert
Ctrl+L	Play Special - Loop
Shift+F5	Window Cascade
Shift+F4	Window Tile Vertical
Shift+F3	Window Tile Horizontal

The Return Key:

Pressing the return key in the real-time or post-processing modes will cause the analyzer to make a single measurement and then stop (single step). The averaging is not reset between steps.

The Focus:

If you click the mouse on one of the scrollbars, that control then gains the "focus" and gains keyboard control. What this means is that you can then use the keyboard arrows to adjust its setting up and down. For example, clicking once on the Average control on the toolbar will then allow you to use the up/down arrow keys to adjust the averaging block size. The same is true for each of the scroll bars.

Note:

Processing will automatically stop when any menu item other than View or Window is selected.

Toolbars

SpectraPLUS contains both a main application toolbar and individual view toolbars:

Main Application Toolbar

The contents of the main application toolbar depend upon the Mode. Each toolbar contains icon buttons for controlling the processor status (Run/Stop etc). In addition, the Real-Time and Post-Processing modes contain controls for adjusting the averaging block size and peak hold status on the fly. These controls duplicate the functionality found in the Settings dialog box.

The Recorder and Post-Processing modes have a horizontal scrollbar which tracks the current position within the .WAV file. The left-hand label shows the current position within the file and the right-hand label shows the total time in the file.

Real-Time

Run	Start the processor
Stop	Stop the processor
Avg	Adjust the averaging block size
Peak	Set the peak hold type

Recorder

Rec	Begin recording audio data
Stop	Stop recording or playing

Fwd   Move current position to end of the file
Rwd   Move current position to beginning of the file
Play   Playback audio data through speakers
Scroll   Adjust current position within the file.

Post-Process

Run   Start the processor
Stop   Stop the processor
Rwd   Move the current position to beginning of the file
Avg   Adjust the averaging block size
Peak   Enable/disable peak hold
Scroll   Adjust the current position within the file

Note:

Double clicking any where in the toolbar background will activate the settings dialog box.

Secondary Toolbar

The secondary toolbar is located just below the main toolbar menu along the top edge of the application.

File New - Close any open WAV file and prepare to record a new one
File Open - Open an existing WAV file
File Save - Save the current WAV file

Print - Print the active plot (time series, spectrum, phase, spectrogram, or 3-D surface)

Time Series View - open/close the time series plot
Spectrum View - open/close the spectrum plot
Phase View - open/close the phase plot
Spectrogram View - open/close the spectrogram plot
3-D Surface View - open/close the 3-D Surface plot

Settings - Brings up the processing settings dialog box
Scaling - Brings up the scaling options dialog box
Calibration - Brings up the calibration dialog box

Trigger - Brings up the triggering options dialog box
Markers - Brings up the marker options dialog box
Device I/O - select input and output devices

Signal Generator - open/close the signal generator

Hz - open/close the Peak Frequency utility
dB - open/close the Peak Amplitude utility
Pwr - open/close the Wideband Level (total power) utility

THD - open/close the Total Harmonic Distortion utility
THD+N - open/close the Total Harmonic Distortion + Noise utility
THD Freq – open/close the THD+N versus Frequency utility

IMD - open/close the InterModulation Distortion utility
SNR - open/close the Signal to Noise Ratio utility
Leq – open/close the Equivalent Noise utility

Mac - open/close the Macro Command Processor
Log - open/close the Data Logging options

Dly - open/close the Delay Finder
Rvb - open/close the Reverberation Time tool
Scp - open/close the Stereo Phase Scope

Note: Double clicking anywhere in the toolbar background will activate the settings dialog box. Use the <Options><Toolbars><Secondary Toolbar> to show or hide the toolbar.

Individual View Toolbars

Selection Arrow: Allows you to graphically select a time segment to edit. Click the selection arrow, then click and drag the mouse across a range of data. When the mouse button is released, the selected time segment will be shown with inverse video. Edit operations (playback, filtering, cut/copy/paste) can then be performed.

Playback Button: Playback the selected time segment. A time segment must first be selected. This performs the same action as the <Edit><Play> menu item.

Filter Button: Filter the selected time segment. A time segment must first be selected. This performs the same action as the <Edit><Filter> menu item

Zoom Selector: Allows you to graphically select the region to expand. Click the Zoom Selector button, then click and drag the mouse across a range of frequency/time data. When the mouse button is released, the display will zoom in to show only the selected data set.

Zoom In 2X: Adjusts the horizontal axis to display one half as many data points.

Zoom Out 2X: Adjusts the horizontal axis to display twice as many data points.

Zoom Out Full: Shows the entire range of frequency/time.

Display Options button: Activates the display options dialog box for the associated view. This performs the same action as the <Options><Spectrum> menu command.

Line Graph button (spectrum view only): Changes the spectrum plot type to a Line Graph.

Stepped Bar Graph button (spectrum view only): Changes the spectrum plot type to a Stepped Bar Graph.

Bar Graph button (spectrum view only): Changes the spectrum plot type to a Bar Graph.

Autoscale button: Evaluates the currently displayed data and sets the Plot Top/Range parameters to optimum settings.

Plot Top: Changes the value of the top edge of the amplitude axis scale. This corresponds to the same control in the display options dialog box for the view. Clicking on the up/down arrows allows you to quickly adjust the plot window as needed.

Plot Range: Changes the total displayed range of the amplitude axis scale. This corresponds to the same control in the display options dialog box for the view. Clicking on the up/down arrows allows you to quickly increase or decrease the displayed dynamic range.

Channel Selection (time series view only): This control allows you to change the display mode from single to dual channel configuration. In addition, it allows you to display the difference between the left and right channels. If "Mono" mode is selected (see Processing Settings), this control is disabled and only the left channel can be displayed.

Intensity Control (Spectrogram only): The Intensity control "slides" the color palette values up or down within the color/grayscale limits when a 256 color display is used.

Note:
Double clicking any where in the toolbar background will activate the settings dialog box.

Status bar

The status bar is located along the bottom edge of the application. It contains the status of key processing parameters:

- Run status (Stopped, Running etc).
- Operating Mode
- Sampling Format (rate, precision and channel count)
- FFT size
- Smoothing window

Notes:
- Clicking the mouse on either the sampling rate, FFT size, or smoothing window items will popup a menu allowing you to quickly change the current settings (the sampling rate cannot be changed when a WAV file has been opened).
- Double clicking anywhere in the toolbar background will activate the settings dialog box.
- The status bar always shows the peak for the left channel only.

Video Display modes

Although changing the video display mode is controlled through the Microsoft Windows setup utility, an overview of the various options is provided here.

There are two primary attributes that define the video display mode:

Screen Resolution:

Determines the number of pixels displayed on the screen (specified as W x H). Several popular screen resolutions are in use today:

- 640 x 480 standard VGA - default setup for Windows.
- 800 x 600 Super VGA - highest resolution useful on 14 and 15 inch displays.
- 1024 x 768 Super VGA - excellent for 17 inch displays or greater.

Increasing the screen resolution will give you more "desktop" space at the expense of smaller fonts. You should be able to comfortably run at 800x600 on a 14 inch monitor unless your monitor quality is poor or your eyesight is marginal.

Color Resolution:

Determines the number of colors that can be simultaneously displayed.

- 16 - (standard VGA) this is the default setup for Windows.
- 256 - required for optimum Spectrogram operation.
- 32 K, 64k, etc.- normally only required for high-end image processing.

Selecting a video mode offering more than 256 colors is counterproductive for SpectraPLUS as it requires more display memory for the same screen resolution and consequently slower refresh times due to the extra overhead. In fact, using more than 256 colors will prevent the use of the spectrogram's Intensity control. This is because the underlying feature for the intensity control relies on what is called "Palette Animation"; something that is only supported in 256 color mode.

Changing Video Modes:

The choice of screen and color resolution are typically limited by the video display expansion card in your computer. As you can see, the default Window setup is less than optimum for running SpectraPLUS. To change the video mode you must use a different display driver - these are provided with your video card. To do this, right click on the Windows desktop and choose the "Properties" menu and then clidk the "Settings" tab. Refer to the Microsoft Windows Users Guide for additional details.

Technical Support

Please feel free to call or write.

Before calling, make sure that your sound board is properly installed and that the correct sound driver is installed. A good test to determine if your sound card is working is to run the "Sound Recorder" utility provided with Windows 3.1 in the Accessories group. Also, take a moment to view the "README.TXT" file for additional instructions.

Pioneer Hill Software LLC
24460 Mason Rd NW
Poulsbo, WA 98370
360-697-3472 tel
360-697-7717 fax
pioneer@spectraplus.com
www.spectraplus.com

Copyright 1995-2010, Pioneer Hill Software LLC
All Rights Reserved.
Legal Notices and trademark attributions.
All trademarks are the property of their respective owners.

Chapter 15 - Application Notes

Voice analysis

By its very nature, the spectrum of a person's voice is quite complex. Because the spectrum changes rapidly with time, we will use what is called "time-frequency" analysis techniques to characterize the voice.

Key settings:

- Mode: Recorder/Post-Processing
- Views: Spectrogram, Spectrum
- Sampling rate: 22050 Hz
- FFT size: 512
- FFT overlap percentage: As detailed below.
- Decimation: 1
- Smoothing window: Hamming
- Averaging size: 2 or 3
- Amplitude Scaling: Logarithmic
- Frequency Scaling: Linear
- 16 bit mono.
- Mic Compensation: As required.
- Standard spectral weighting: flat

Procedure:

- Record a short time segment of the voice to be analyzed.
- Change to the Post-Processing mode.
- Initially set the Overlap Percentage to zero.
- Click the rewind button to position the current file position to the start of the file.
- Run the analyzer until the spectrogram display is filled or the end of file position is reached.
- Identify the voice segment to be analyzed in detail by highlighting a portion of the spectrogram and play it back through the speakers.
- Use the left mouse button to readout the time corresponding to the start of the voice segment under analysis.
- Use the scroll bar on the main application toolbar to adjust the current file position to the start of the desired time segment.
- Set the overlap percentage to 90%
- Run the analyzer and examine the spectrogram view.

- Readjust the overlap percentage to expand/contract the total time displayed on the spectrogram time axis to give you the desired detail.
- Double click at any point on the spectrogram to place the spectrum corresponding to the cursor position in the spectrum view. (if the time series view is displayed, the corresponding time series data will be displayed as well)
- You can open the spectrogram display options dialog box and adjust the plot top, plot range and frequency span to optimize the display. The spectrogram will be recomputed with the new settings when the dialog box is closed.

Notes:

The same procedure applies when using the 3-D surface plot in place of the spectrogram.

Transfer Functions

A transfer function is a total measurement of the amplitude and phase response of an audio device.

A key part of an accurate transfer function is the removal of the input test signal response from the final measurement. This is done by passing a sample of the input signal to the second channel of the analyzer and removing its response from the response of the device under test.

Transfer Function Measurement Setup
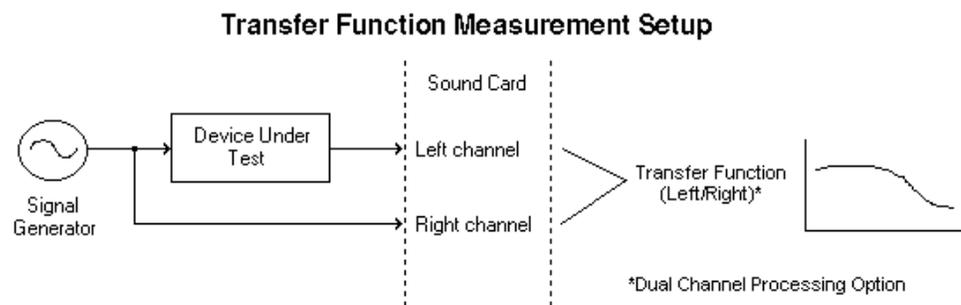
Key Settings:

- Mode: Real-Time
- Views: Spectrum, Phase

- Sampling Rate: At least twice the highest frequency produced by the device under test
- FFT size: 4096 typical
- Averaging: 4
- Peak Hold: Off
- Smoothing Window: Hanning
- Dual Channel Spectral Processing: Real Transfer Function Left/Right"
- 16 or 24 bit Stereo
- Frequency scaling: your choice
- Amplitude scaling: Logarithmic
- Spectral Weighting: flat

Procedure:

- Connect a white noise generator to the input of the device under test.
- Connect the output of the device under test to the Left input channel of the analyzer.
- Connect same signal generator to the Right input channel of the analyzer.
- In the <Options><Settings> dialog box, select the dual channel processing option called "Real Transfer Function Left/Right".
- Run the analyzer until the average spectrum response settles to a consistent level.

Interpretation:

The spectrum display contains the amplitude response of the device and the phase display contains the phase response.

Total Harmonic Distortion (THD) Analysis

Real world audio devices introduce distortion in addition to the desired response. Total Harmonic Distortion (THD) is a measure of this quantity.

THD is the ratio of the harmonic power to the fundamental power. It is expressed as a percentage. The lower the THD value, the "cleaner" the response. A typical THD measurement will use a 1kHz reference test tone. This reference signal is passed through the audio device under test, and its THD computed from the spectrum

Key Settings:
- Mode: Real-Time
- Views: Spectrum, Time Series
- Amplitude axis: Logarithmic
- Frequency axis: Linear
- Standard Weighting: Flat
- Mic compensation: None
- Sampling rate: 44100Hz
- FFT size: 8192
- Decimation: 1
- Averaging size: 10
- Smoothing Window: Hanning
- 16 or 24 bit Mono

Procedure:
- Connect a 1kHz test tone to the input of the device under test.
- Connect the output of the device under test to the input of your sound card.
- Run the analyzer for sufficient time to allow the spectrum response to settle.
- Stop the measurement and select <Total Harmonic Distortion> under the <Utilities> menu item.

Machine Vibration Analysis

Vibrations in machines can cause equipment failure, down-time in production, and poor manufacturing of materials. These failures cost industry billions of dollars every year. Discovering causes and sources of these vibrations can save money and lead to increased productivity.

Key Settings:
- Mode: Recorder/Post-Processing
- Views: 3-D surface, Spectrum
- Sampling rate: 11050 Hz
- FFT size: depends on test length (15 min. test use 32k)
- FFT overlap: As detailed below.
- Decimation: 15 for low frequency vibration < 60 Hz
- Smoothing window: Hanning
- Averaging size   20
- Amplitude Scaling: Logarithmic
- Frequency Scaling: Linear

- Sampling Precision: 16 or 24 bit mono
- Input: Accelerometer

Test Procedure:

- Start the Recording
- Slowly ramp-up machinery to full speed. Then let machines run for a few minutes at full speed and ramp-down until stopped.
- Stop the Recording
- Change to the Post-Processing mode.
- Initially set the Overlap Percentage to 50%.
- Click the rewind button to position the current file position to the start of the file.
- Run the analyzer until the 3-D display is filled or the end of file position is reached.
- You should easily see the ramp-up and ramp-down. Look for resonances and beat frequencies these are often the cuase of damage.
- Use the left mouse button to readout the time corresponding to the start of the resonances or beats.
- If the machine produces a product, inspect the product that corresponds to the times of the resonances and beats for defects.

Note:

The same procedure applies when using the spectrogram in place of the 3-D surface plot.