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What is a Spectrum Analyzer

Welcome to the world of signal processing! Just what is a Spectrum Analyzer and what can it do?

What is it?

A spectrum analyzer is a laboratory 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 shows this information. An **Audio** Spectrum Analyzer, by definition, is limited to processing signals in the audio band. The specific frequency limit is determined 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. **Spectra Plus** 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.

Applications:

Amateur Radio:

Use as an aid for tuning in RTTY and other digital modulation on an HF receiver. Examine the characteristics of passband and notch filters.

Speech:

Determine the dominant frequencies and harmonics in an individual's voice.

Music:

Examine the spectrum of an electronic keyboard or synthesizer. Determine the frequency response of various musical instruments and audio components.

Industrial

A multitude of applications from vibration analysis to audio equipment design, diagnostics and repair.

Basic Operation

The program has three distinct operating **Modes** and four 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 in memory and, optionally, store it on disk in 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 Spectrogram view displays the spectra over time. The amplitude is shown in color.
- The 3-D Surface view displays a perspective of the spectra over time.

Real-Time Mode

When in this 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 and display the results until stopped. The toolbar contains controls to allow you to modify the gain and averaging block size while running.

The digitized audio data is not saved in memory and cannot be replayed or saved to disk.

Note:

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.

See also: [Basic Operation](#), [Performance Issues](#), [Settings](#)

Post-Processing Mode

This mode allows you to process from a .WAV file or from data stored in memory and not yet saved to disk.

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 position the offset to any point in the file using the scroll bar in either the toolbar or [Time Series](#) view.

When a .WAV file is opened, the [Sampling Rate](#) and [Precision](#) 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.

See also: [Basic Operation](#), [Overlap Percentage](#), [Performance Issues](#), [Working with Audio Files](#)

Recorder Mode

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 Precision and Sampling Rate. The program will also display the spectrum of the signal while recording or playing.

Since the recorded audio data is stored in memory, the maximum recording time is limited by the amount of free memory on your computer. The program will attempt to allocate enough memory for 60 seconds of audio and will reduce the recording length if not enough memory is available. After 60 seconds of data have been recorded, click <Rec> again to record another 60 seconds.

The amount of memory required for each minute of audio varies widely depending upon the Sampling Rate and Sampling Precision. Some typical values are shown below:

<u>Sampling Rate (Hz)</u>	<u>Sampling Precision (bits)</u>	<u>Bytes/Minute</u>
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		

When a .WAV file is opened, the Sampling Rate and Precision 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.

See also: Basic Operation, Sampling Rate, Device Options

Time Series View

This view displays the digitized audio data and is similar to an oscilloscope display with the amplitude shown on the vertical axis and time on the horizontal axis. The amplitude scaling depends upon the Sampling Precision. Full scale 8 bit is +/-127; full scale 16 bit is +/-32,767.

When in Real-Time mode, the plot will display the most recently acquired block of data (FFT size points) and since the previous data is not stored, you can only scroll within the current FFT size block of data. The horizontal scaling is in milliseconds with respect to the start of the data block. For instance, if the sampling rate was set to 10240 Hz and the FFT size to 1024, the display will show 100 milliseconds (1/10th of a second) of audio.

In Recorder or Post-Processing modes you may scroll through the entire data file. The horizontal scaling is in seconds relative to the start of the file. In addition, the **Zoom** button is displayed. The **Zoom** button toggles between displaying the entire WAV file and an FFT size data block. Double clicking anywhere in the plot while Zoomed "Out" will move the current processing position. In this mode, the data may look distorted - this is because there are typically many more points in the data file than pixels on the display.

Note:

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

See also: Performance Issues, Device options

Spectrum View

This view is a 2 dimensional plot of the spectrum. The vertical axis shows the relative power of each frequency line and is scaled in Decibels (dB). The horizontal axis shows the frequency in Hertz (Hz).

The number of 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.

The plot may be resized to show only a portion of the total lines; however, the FFT algorithm requires that all frequency lines must be computed regardless of whether or not they are displayed.

See Also: Cursor Measurements, Scaling, Measurement Accuracy, Performance Issues

Spectrogram View

This view displays the spectral data over time. The frequency axis is the same scale as the [spectrum plot](#) but the Y-axis now shows time. Color is used to show the amplitude. This view is especially useful for tracking frequency changes over time. The vertical axis is scaled in seconds relative to the start of the plot.

Notes:

Due to memory considerations, the display cannot be scrolled vertically.

The display will automatically be cleared if one of the following parameters changes:

- [Sampling Rate](#)
- [FFT size](#)
- [Plot Range](#)

The display can also be cleared by using the Clear All command under the View menu.

See Also: [Cursor Measurements](#), [Scaling](#), [Measurement Accuracy](#)

3-D Surface View

This display is a perspective view of the spectral data over time. The frequency axis is the same scale as the [2-D Spectrum plot](#), and [Spectrogram](#). The Y-axis shows time. This view is useful for better visualizing the audio spectra. The vertical axis scaling is in seconds relative to the start of the plot.

Notes:

Due to memory considerations, the display cannot be scrolled vertically. Maximize the window to show a full screen's worth of data.

The display will automatically be cleared if one of the following parameters changes:

- [Sampling Rate](#)
- [FFT size](#)
- [Plot Range](#)

The display can also be cleared by using the Clear All command under the View menu.

See Also: [Cursor Measurements](#), [Scaling](#), [Measurement Accuracy](#), [Performance Issues](#)

Sampling Rate

The program 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.

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. A note of caution: if there are unwanted signals which are greater than 3,000Hz, Aliasing will occur.

When a .WAV file is opened, the Sampling Rate and Precision is changed to match the rate at which the file was recorded and the 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 the upper and lower sampling rate supported by your card, enter a value of -1 in the edit box and press OK. A message box will display the hardware limits.

See also: Aliasing, FFT size

Settings

The Settings dialog box contains the following options:

- FFT Size
- Sampling Rate
- Averaging Block Size
- Smoothing Windows

Averaging Block Size

This 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 an average of the previous 4 traces.

Averaging will effectively reduce the noise level. 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.

Note:

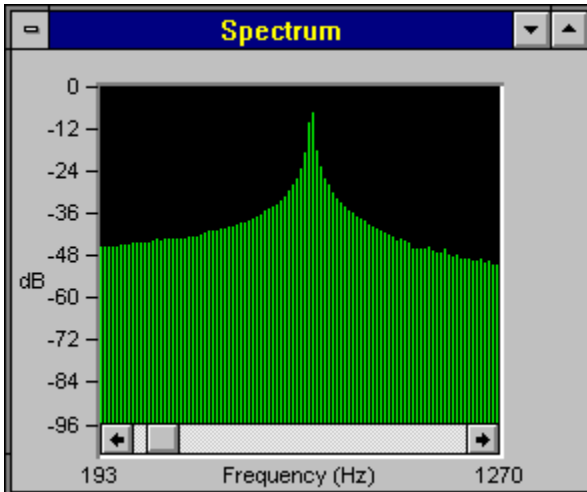
In both the Real-Time and Post-Processing modes, the averaging block size can be modified while processing by using the "Avg" control on the toolbar.

Smoothing Windows

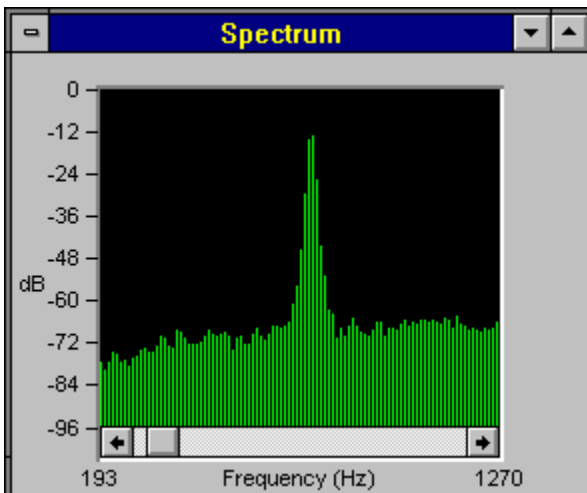
Use this dialog box to select the weighting function to apply to the time series before performing an FFT. These functions are used to control "spectral leakage" due to the edge effects of the time series sequence. They accomplish this by forcing the beginning and ending values of the sequence to approach zero. To make up for this reduction in power, windowing algorithms give extra weight to the values near the middle of the sequence. 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.

The functions are listed in increasing order of leakage suppression with **Uniform** providing the least suppression but sharpest tone resolution. The effects of these functions are most noticeable when using Logarithmic scaling.

A Uniform window has a weighting function of 1.0 and is really no function at all. The plot below shows the spectrum of a single tone using a **Uniform** smoothing window:



The following plot shows the spectrum of the same signal using the **Hanning** smoothing window function. Note the reduction in the spectral leakage (sidelobe level) and the broadening of the peak.



Comparison of Smoothing Window performance:

Function	Sidelobe level (dB)
Uniform	-13
Hanning	-23
Hamming	-43
Blackman	-58

The Hamming window is the most popular and widely used general purpose window function.

Note: The Time Series View displays the spectra **prior** to the application of any smoothing window.

FFT Size

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:

$$8192 / 1024 = 4 \text{ Hz.}$$

Larger FFT sizes result in a more accurate spectra but take longer to compute.

Notes:

- The current frequency resolution is displayed along the top of the settings dialog box and will update whenever the sampling rate or FFT size changes.
- The current FFT size is always displayed in the status bar along the bottom of the application.

See also: Sampling Rate

Scaling

Linear Scaling

This causes the amplitude of the spectra to be scaled linearly, but since the units are in Decibels (dB), the tic marks are not equally spaced. Linear scaling is recommended when 8 bit sampling precision is used.

Logarithmic Scaling

This causes the amplitude to be scaled logarithmically and the tic marks are equally spaced. Logarithmic scaling is useful in that it allows both strong and weak signals to be viewed together easily. Logarithmic scaling is recommended when 16 bit sampling precision is used.

Plot Range

This sets the total plot range that will be displayed (Logarithmic scaling only). It does not change the sampling precision or the actual dynamic range of the measurement.

Auto Gain

When selected, the spectral display is automatically scaled so that the strongest spectral line in the current measurement will extend to the top of the plot region. Note: when little or no signal is present on the input, large random spectral lines will result.

Manual Gain

When selected, the display is scaled using the selected gain value. A gain of 0 dB represents the strongest possible signal from the sound board. Because the strongest possible signal is an absolutely pure sinusoidal tone - rare in the real world - some gain is necessary.

Notes:

- Scaling does not affect the Time-Series view.
- Scaling changes will be in effect for the current Spectrum plot and for all subsequent traces in both the 3-D Surface plot and Spectrogram plot. Previous Spectrogram and Surface Plot data is not rescaled.

See Also: Measurement Accuracy

Keyboard Shortcuts

The 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.

There are several standard shortcut keys for all MDI (Multiple Document Interface) applications which are worth noting:

- <Ctrl> <F6> Cycles through each view.
- <Ctrl> <F4> Close the active view.

The horizontal scroll bar can be controlled by the keyboard as well as with the mouse. Click once on the scroll bar to give it the "focus". The following keys can then be used.

Right and Left arrow keys

The arrow keys cause the spectra display to scroll right or left one line at a time.

PageUp and PageDown

The PageUp and PageDown keys cause the spectra display to scroll right or left by 1/8 of the full spectra width.

Home and End

Home causes the spectra display to scroll fully to the left; End causes the spectra to scroll to the highest possible frequency.

Note:

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

Markers

Markers are used to highlight specific frequencies of interest.

The selected frequencies are marked differently depending upon the view:

- Spectrum - marked line is displayed in red.
- Spectrogram - red marks are shown along the bottom edge of the display.
- 3-D Surface view - a red band is displayed at the marked frequency.

Note:

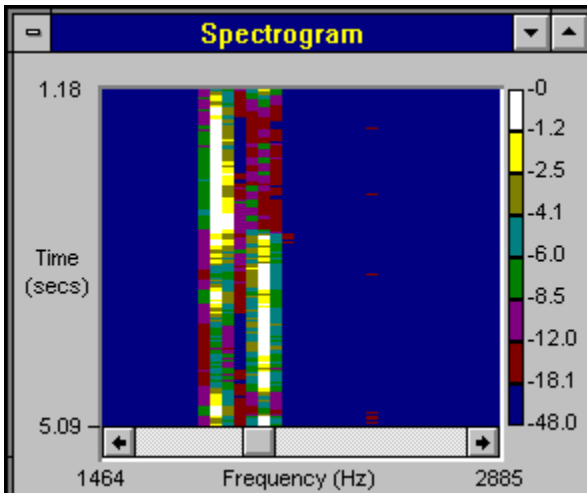
The line marked may not be the exact marker frequency depending upon the Sampling Rate and FFT size, but it will be the closest possible frequency.

Overlap Percentage

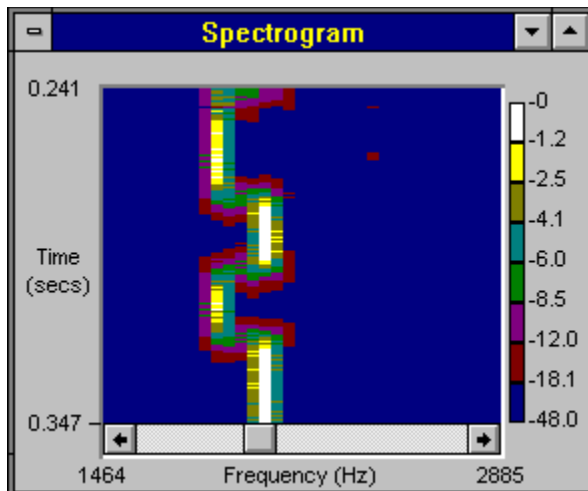
This feature allows the time axis on the Spectrogram and 3-D Surface plot to be "stretched" by re-processing portions of the time series. The setting determines what percentage of the FFT size block of data points is re-used for the next trace. High overlap percentages (90+%) and low FFT block sizes 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 (Radio Teletype (RTTY) uses such a scheme), it is difficult to discern the pulse width of each tone because both frequencies are often contained within a single FFT size time series block. If we performed an FFT and, instead of moving forward a whole FFT size block in the series, moved forward only a small amount then we can isolate the pulse timing for each tone.

The Spectrogram below shows a typical RTTY signal processed with 0% overlap percentage. Note how it is impossible to determine the pulse widths.



The same signal was re-processed using an overlap percentage of 98%. In this case it is possible to measure the pulse width along the time axis. Note how the time axis is "stretched" compared to the plot above.



Note: High overlap percentages will require greater processing times. If you know the approximate time resolution required, you can use the text line along the top of the dialog box to help set the percentage.

Triggering

When triggering is enabled, the processor will test for a time series data value which exceeds the specified threshold before performing an FFT. This provides a method for analyzing transient pulses. Negative or positive thresholds can be entered.

Caution:

With triggering 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.

Note:

Triggering is available in both Real-Time and Post-Processing modes but is disabled in the Recorder mode due to performance limitations.

Device Options

This allows you to select a specific sound card if more than one is installed in your system. In addition, the Sampling Precision (8 or 16 bit) can be selected.

Once a WAV file is open, the device options cannot be changed. Close the file to change device options.

When a .WAV file is opened, the Sampling Rate and Precision 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.

Notes:

- 16 bit cards can be used in 8 bit mode.
- If you have an 8 bit card you will be unable to select 16 bit sampling precision but you can Post-Process from a pre-recorded 16 bit WAV file.

FFT

FFT is an abbreviation for a Fast Fourier Transform.

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.

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.

To prevent this, the sampling rate should always be set **greater than twice** the highest possible frequency that will be present on the input. Applying a low pass filter to the audio line prior to the input jack on the sound card will also solve the problem.

See also: [Sampling Rate](#), [Measurement Accuracy](#)

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:

Because of the limited dynamic range of 8 bit sound cards, most of the 8 bit boards perform internal Automatic Gain Control (AGC) and it is therefore impossible to determine the absolute amplitude of the input signal. However, many 16 bit boards provide control over the input level via a software utility and it is possible to determine the actual signal amplitude by comparison with a known source. (See notes below for tips on determining if your card uses AGC).

Dynamic Range:

The theoretical dynamic range of the system is as follows:

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

The actual dynamic range achieved is somewhat less due to factors such as computer generated electrical noise.

Notes:

To determine if your card uses AGC, set the analyzer to Real-Time mode and display only the Time Series View. Suddenly switch on a strong signal and examine the response. If AGC is present the signal will peak initially but then decay down to a constant value. If no AGC is used the signal level will not decay and will easily overdrive the input.

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> from the <File> menu. Change any printer options and click OK.

Notes:

- For best results when printing the color Spectrogram on a monochrome printer, reduce the **intensity** on the options dialog under the Print Setup dialog box (not all printers support this feature).
- For printing the 3-D Surface plot on a monochrome printer, you may wish to turn off **dithering** on the options dialog under the Print Setup dialog box (not all printers support this feature).
- Color printers are supported.
- Pen Plotters are not supported for the Spectrogram and 3-D Surface plots

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. 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 device options cannot be changed. You must close the file. This is because a WAV file format requires a constant Sampling Rate and Precision.

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

See also: Recorder Mode

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.

Right Mouse Button:

Pressing and holding the right 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 tones.

Note:

- It is recommended that you "Stop" the Analyzer while using these mouse actions. There are no corresponding keyboard commands.
- The 3-D Surface plot can be deceptive; you must measure along the base of the plot to determine the frequency. Aligning with the top of a peak will skew the measurement to the left because of the perspective orientation of the display.

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. For stereo cards, use the left channel.

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 (displayed directly under the title of the dialog box) meets your objectives. Click OK to close the dialog box.

Amplitude Range:

Go to the Scaling dialog box by selecting the <Scaling> command under the <Options> menu item. Select Linear scaling if you have an 8 bit sound card and Logarithmic scaling for 16 bit cards. Set the Gain control to **Manual** and the gain value to 20 dB. These are the default settings.

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 memory 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 keys <alt> <R> to start the processor. The displays will then begin updating.

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)

See also: Basic Operation, What is a Spectrum Analyzer?, Working With Audio files, Aliasing, Printing

Audio Band

Frequencies in the audible range of hearing (20 to 20,000 Hz)

System Requirements

Hardware requirements:

- IBM PC or compatible with 80386 CPU or higher.
- 2 MB RAM minimum, 4MB or more recommended for recording.
- VGA monitor capable of a minimum of 16 colors.
- Hard Disk with 1MB free space. Additional space is required to save audio files.
- Math co-processor recommended but not required (recommended for Logarithmic scaling).
- Windows compatible sound card: 16 bit card recommended.
- Mouse or other pointing device.

Software requirements:

- Microsoft Windows Version 3.1 or higher.
- Sound Card drivers (supplied with Sound Card)

Status Bar

The status bar is located along the bottom edge of the application. It contains the status of several key parameters. The left-most label contains the current processing status (i.e. "Running...").

The selected FFT size and Sampling Rate shown just to the right of the processor status.

The "Peak" label contains the frequency of the strongest tone. This label is constantly updated with the current value unless only the Time Series view is open. In this case no FFT is performed and the peak frequency is not available (N/A).

Toolbar

The toolbar controls change depending upon the Operational Mode. Each toolbar contains icon buttons for Run, Stop, Rewind etc. In addition, the Real-Time and Post-Processing modes contain controls for adjusting the gain and averaging block size on the fly. These controls duplicate the functionality found in the Scaling and Settings dialog boxes respectively.

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 in seconds, and the right-hand label shows the total number of seconds of sound data in the file.

List of Tools:

Real-Time

Run	Start the processor
Stop	Stop the processor
Avg	Adjust the <u>averaging block size</u>
Gain	Adjust the <u>gain</u> value

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 <u>averaging block size</u>
Gain	Adjust the <u>gain</u> value
Scroll	Adjust the current position within the file

See also: Basic Operation, Keyboard Shortcuts

Processing Speed

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

Video speed is another 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 monitor scope

