# Spect&Gen

The Spectrum Analyzer

The Spectrum Sheet shows the Audio spectrum, with a linear frequency scale in abscissa and a logarithmic ordinate in "dB" (decibels) for the signal Amplitude. The frequency scale depends on the Sampling Frequency of the Input or Output signal, and on the Upper Frequency chosen by the user.

Remark: the 0dB level corresponds to the maximum numerical level, on 16 or 24 bits, not to the conventional analog level of 775 mVRMS.

Most often, an analog level of 775 mV will have an magnitude between -1dB and –12dB on the spectrum sheet, depending on the hardware of your Audio Input peripheral.

To get the precise values of the frequency and the magnitude of the spectral peaks, it is advisable to add "Peak Labels" that will be drawn on the spectrum sheet.
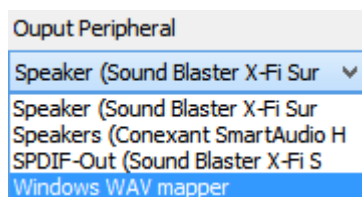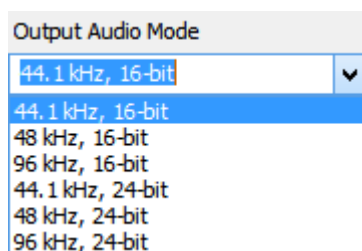
User Interface:
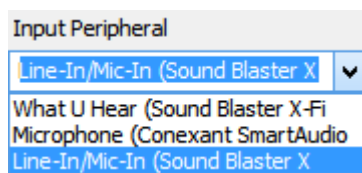
Selects the Audio input peripheral. The combo box lists every Audio input peripheral discovered on the system.
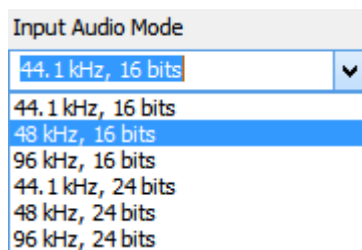
Selects the Audio input sampling rate (44.1 Ksps, 48 Ksps, 96 Ksps) and the sample resolution (16 bits or 24 bits). Not every format is supported by every peripheral. Notice that a peripheral may accepts the 24 bit depth while its genuine resolution is only 16 bit (fake 24 bit samples). The software verifies the real resolution.

Selects the Audio output peripheral. The combo box lists every Audio output peripheral present on the system.

Selects the Audio output sampling rate (44.1 Ksps, 48 Ksps, 96 Ksps) and the sample resolution (16 bits or 24 bits). Not every format is supported by every peripheral. When a peripheral accepts the 24 bit depth, this does not mean that it is capable of delivering it! You can check the overall quality by looping the Audio Output on the Audio Input and using the Audio Generator together with the Spectrum analyzer.

Open Audio Input

Opens (or closes) the currently selected Audio Input peripheral. The Audio Input peripheral can be opened even if the Audio generator is running.

Open Input WAV File

Opens an Audio file in "WAV" format. Once opened the file is played and its spectrum is displayed.

Play

Plays the Audio Input signal or the "Wav" file. The Audio generator must be stopped to free the Audio output peripheral.

Stop Spectrum

Starts (or stops) the Spectrum display.

Spectrum Channel
Left + Right

Selects the Audio channel used for the spectrum: left channel, right channel, both channels (L+R) or differential (L-R) to use with the equipments delivering a differential audio signal.

Frequency range
20000

Selects the maximum frequency for the spectrum analysis and display. This choice has no impact on the Audio output and on the recorded signal.

High Pass filter
No Filtering

Cuts the low frequencies to get a better display of the significant frequencies. This choice has no impact on the Audio output and on the recorded signal. The filter is a third order smooth filter with a Bessel or Butterworth frequency response.

◉ Bessel ○ Butterworth

The Butterworth filter has a maximally flat magnitude in the passband.
The Bessel filter has a maximally linear phase response.

FFT length
8192

Selects the length of the Fast Fourier Transform, i.e. the number of audio samples accumulated to perform the Spectrum analysis. The longest length gives the best frequency resolution, but the worst time reactivity. Long lengths can only be used with a stable input signal, as provided by an Audio Generator.

Windowing
Gauss

This is the type of "Window" used to weight the audio samples before performing the Fast Fourier Transform.
Only the Gaussian and the "Blackman" windows are proposed because they permit to use a very efficient interpolation algorithm to compute the frequency and the level of the spectrum peaks.

Nb of Peak Labels
16

You can display up to sixteen labels on the spectrum sheet, indicating the frequency and the level of the highest spectrum peaks.

☐ Lin. Amplitude Scale

Check this box to display a Linear Amplitude scale.

The maximum level is '1', but the input sensitivity can be multiplied by 1, 2, 5, 10, 20 or 50.
Otherwise the Amplitude is Logarithmic in dBs with 0dB being the max level for a sine wave.
Depending on your Audio Board, the max level does not necessarily corresponds to the conventional analog level of 775 mV RMS.

☐ Log. Frequency Scale

Check this box to display a Logarithmic Frequency scale, otherwise the Frequency scale is Linear.
The maximum and the minimum frequencies depend on the sampling frequency and the FFT length.

Record

When Recording is requested, the Audio signal it is stored to the opened "wav" file in "wav" format. When the Audio Input is opened, the Audio Input signal is recorded. When the Audio Generator is activated, its signal is recorded (Output signal). The filtering that may eventually be active on the Spectrum display does not impact the recording.

Generator

When activated, the Generator takes hand on the Audio output peripheral.
Its output signal can also be recorded to a file. When the generator is running its spectrum is displayed.

Scope

Click on this button to open a simple oscilloscope.
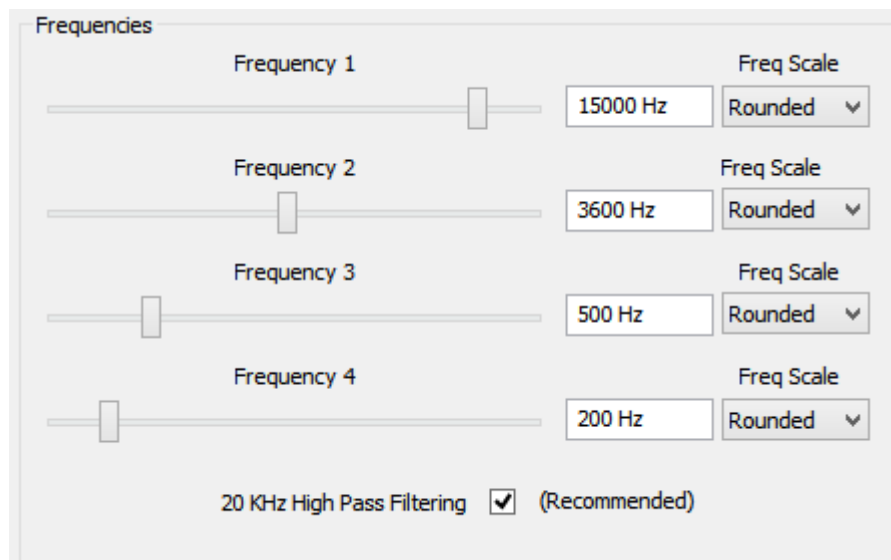
The Wave Generator

Four wave generators are available. Each can generate Sine, Square, Triangular and Sawtooth waves.
The non sinusoidal waves are "spectrally optimized", this means that only the harmonic contained within the Nyquist band are generated ($<$ half the sampling frequency). Otherwise, the Shannon criterion would be violated and aliazed frequencies (= mirror frequencies) would be present.
As a consequence, in the time domain, the oscillogram is not perfect. Some "overshot" and "ringing" can be observed on the scope. This is normal. Tis caused by the missing harmonics. A signal cannot be optimized both in the frequency and in the time domain. (Heisenberg uncertainty principle).
The harmonics between 20 Khz the the Nyquist frequency can be filtered (but not cancelled) to smooth the square and saw waveforms.
The purpose of the generator is not to measure the rise and fall time of an equipment, but its behavior when fed with complex signals. (e.g.: intermodulation distortion).

User Interface:

The frequency can be adjusted between 20 Hz and 44000 Hz, depending on the Audio Output Sampling rate and on the chosen Frequency scale.
The slider can be adjusted with the mouse and also with the "<-" and "->" arrow keys on the keyboard. The mouse wheel can also be used when available.
The Frequency scale can be "Linear", i.e. adjustable by 1 Hz increments, or "Rounded" with only round frequency values or "Tempered" with the 88 frequencies of a grand piano.
A filter is available to smooth the amplitude of the harmonics of the Square, Triangular and Saw waveforms above 20 KHz. It can also be used to limit the White Noise bandwidth.
This is recommended for sampling rates above 48 Ksps.
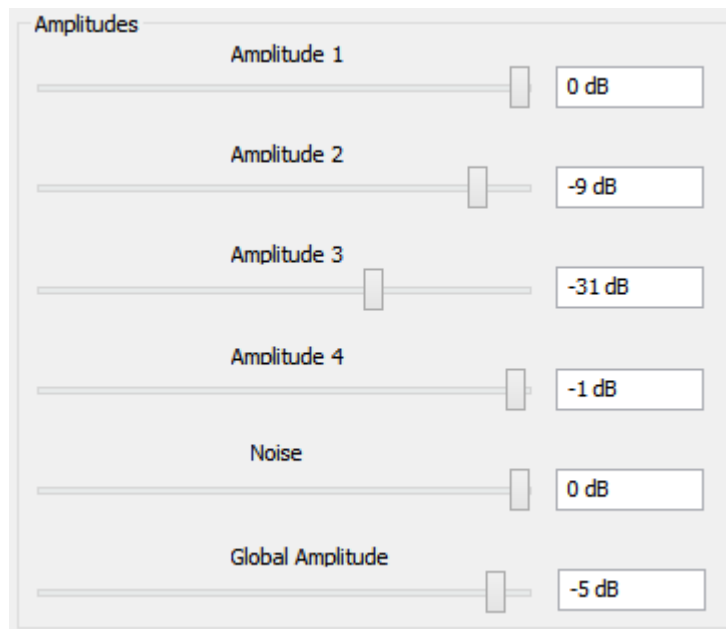
The Amplitude can be adjusted between 0dB and -100dB.
A "Noise" generator is also available, that delivers "white" or "pink" noise.

- Remark1: The 0dB level corresponds to the maximum numerical level, on 16 or 24 bits, not to the analog level delivered by your audio peripheral. Most often, this maximum level is much higher than the conventional analog level of 775 mVrms at 0dB. The slider can be adjusted with the mouse and also with the "<-" and "->" arrow keys on the keyboard. The mouse wheel can also be used when available.

- Remark2: The amplitude of the "noise" cannot be reliably measured with the spectrum analyzer because the power of this random signal is noise is uniformly distributed over the whole bandwidth, and the level displayed depends on the FFT analysis bandwidth. Therefore the spectral level is meaningless. Use the oscilloscope to estimate the peak level.
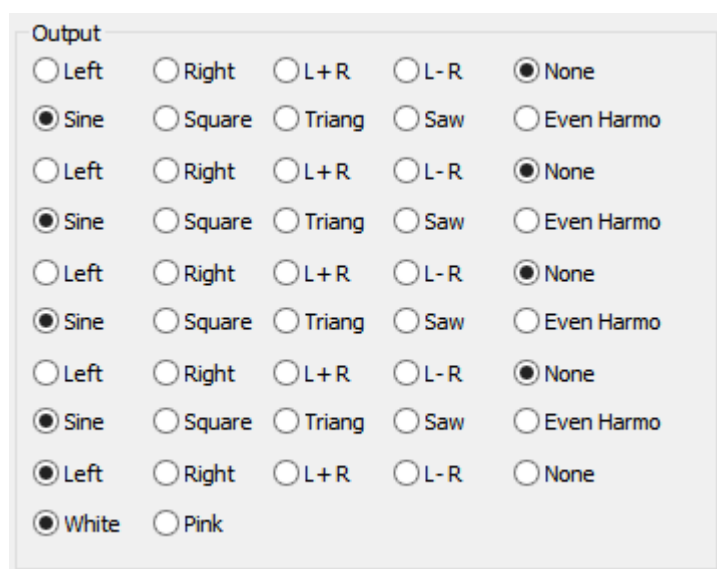
The output signal can be directed to the left channel or the right channel or both, in phase or out of phase. Be aware that when the output is directed to both channels, and the spectrum analyzer is programmed to observe the "Left+Right" channels, the observed amplitude will be TWICE the amplitude programmed on the generator. (+6dB).

Several waveforms are available: Sine wave, Square wave, Triangular wave, Saw wave and a awkward waveform with only even harmonics.

For the noise generator, two possibilities are available: a 'White' noise with a flat spectrum or a 'Pink' noise with a power density which falls off at 10 dB/decade of frequency.
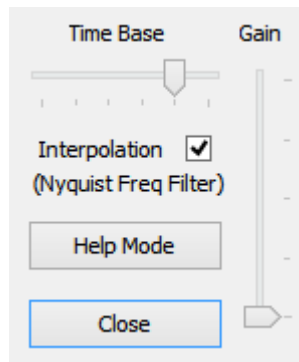
The Oscilloscope

A very simple oscilloscope is available to observe the signal over time.

The Y ordinate gives the signal amplitude. This is a linear scale (1 = 0dB = maximum level).

The X ordinate is the Time scale, in units of time per major graticule division. The time unit depends on the selected speed and also on the signal sampling rate.

User Interface:

The "Time Base" slider is used to adjust the "Time Base" or "Horizontal Sweep" from the slowest to the fastest sweep rate.

The "Gain" slider is used to control the sensitivity, i.e. the signal amplitude. The gain can be adjusted from x1 to x20."

A check box is available to enable the signal interpolation. When activated, the display smoothness will be much improved, especially at fast sweeping rates. This is advisable, unless you need to observe the sampling effect itself.

Here is an example of the read-out of a 10 KHz sine wave without and with the interpolation filter activated: