





### Benefits

- PC-based loudspeaker or sound reinforcement system measurements
- Convenient, powerful measurement and optimization tool for the sound reinforcement system installer
- Multiple measurements can be taken at several measurement locations and averaged
- Automatically determines parametric PEQ values (Frequency, Q and Gain)
- X-Over alignment tab where you can find the delay needed to align subs with full-range speakers



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### Presets

saves your settings for the stimulus and analyzer. A range of useful settings is provided for different applications.

### GUI

Much effort is put in the new user interface to reduce the need for dialog boxes and spin-boxes. These are replaced with modern mouse wheel and grab 'n' drag mouse based controls. For example if you need to change the range of a graph, this is now done with the mouse wheel. Dragging the scale makes zooming.

### Waterfall 3D and 2D display

Wavelets are considered as more hearing-like frequency-time analysis than traditional Spectrogram. Wavelets offer the same frequency resolution, i.e. constant relative bandwidth, but increased time resolution at higher frequencies.

Wavelets are useful for room acoustic analysis such as diffuser tuning, loudspeaker driver time domain analysis and system analysis of time smearing.

### X-over aligner

This is the quick way to fine-tune the levels and delay between bands in a multi-way active system. The X-over Aligner offers an intuitive graphical aid for delay alignment with synchronized time and frequency graphs. A range of useful curves can be displayed such as Impulse response, ETC or Cepstrum in the time domain. In the frequency domain both group-delay and wrapped phase can be chosen, depending on the preferred alignment strategy. The X-over Aligner also offers two automatic delay finder algorithms.

### Quantities

Added background noise measurement and considering background noise in the STI calculation. Considering Masking in the STI calculation is a new feature. Improved Noise Correction and regression line algorithms for reverberation time calculations.

### Features

- Acquires complex frequency response with up to 8 channels simultaneously
- MLS (up to 4096k points), Log Sine Sweep (Chirp) and Dual FFT (using external Wavefiles)
- Import text-files from Audio-Capture, Clío, MLSSA (frequency and impulse response)
- Variety of output formats for direct transfer to most common digital loudspeaker processors
- Built-in frequency response and microphone compensator
- Support for any full duplex stereo soundcard or multi-channel soundcard
- Calculation of ISO 3382 acoustic quantities such as RT60, clarity (C50) and lateral energy
- Calculation of STI/RASTI/STITEL/STIPA according to IEC 60268-16 and other measurement system weighting tables (MLSSA, TEF)
- %ALcons computed from the weighted STI value
- Filtered Inverse Schroeder Integral, ETC and CEPSTRUM available in the acoustic quantities tab
- Waterfall which can be viewed in both 3D and 2D
- RTA with up to 48 points per octave
- Spectrum Analyzer with distortion analyzer
- Function Generator

### Requirements

- PC with XP SP2 or Vista SP2 or Win7 or Win8; 32 or 64 bits
- CPU: Intel i3 or better. Two or more cores.
- RAM: 2 GB min, 4 GB or more is recommended for 64 bits.
- Display: minimum 1280 x 800 pixels.
- Soundcard: Windows compatible (Wave/WDM or ASIO) with 2, 4, 6, or 8 inputs, 16-bit/44.1k to 24bit/96k sampling, with full duplex (simultaneous play and record) capability