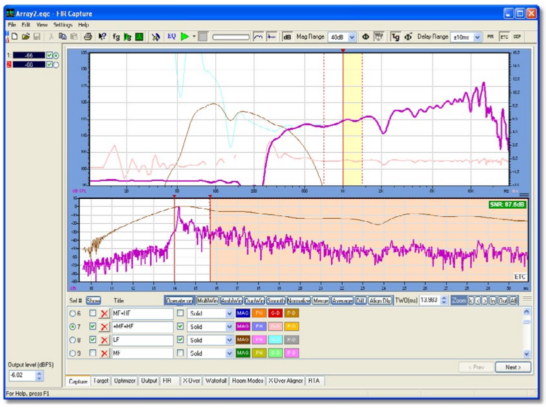
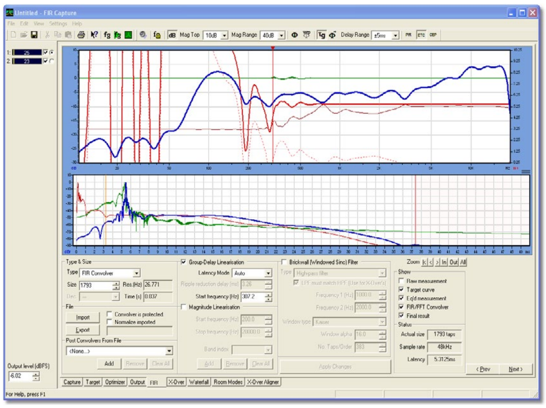
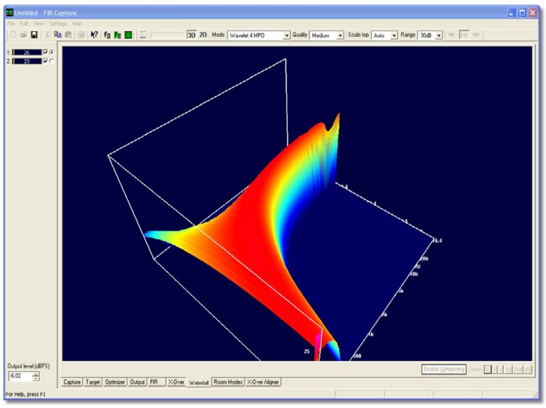
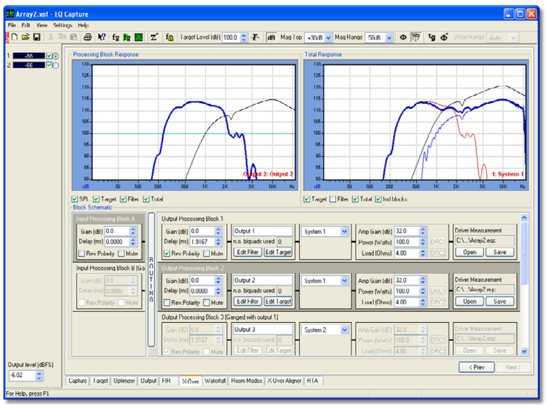
WaveCapture.com

a Bävholm/ Grenander Mission









Take control of the time domain!

FIR-Capture

FIR-Capture is a powerful PC-based measurement software tool for generating filters parameters, FIR coefficients, FFT convolver filters, linear & mixed phase filters and brick wall filters.

FIR simulator

Any captured transfer function can be run through an optimization algorithm that produces magnitude and/or group-delay / phase equalization. The magnitude equalization offers multiple bands linearization and using predefined target curve. The Group-Delay linearization provides adjustable start-frequency and ripple reduction delay.

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 drag the start and stop frequencies for the different filter types with the mouse and the convolver is then updated immediately when the mouse button is released.

The FIR designer shows both frequency graph and impulse graph simultaneously with individual curves like before and after convolution and the actual FIR/FFT convolver filter.

Linear-phase brick-wall filters with preset 24 – 96dB slopes or freely optimized slopes can be enabled to replace their IIR counterparts.

Up to 16 arbitrary curves or FIR/FFT convolver filters can be mixed to one common filter. Most one or two column .TXT and .CSV export and import format are supported. A number of well-known processors with FIR capability will have direct network interface.

Crossover simulator

FIR filters and parametric filters can be combined into the crossover simulation to be able to get an overview of a complete loudspeaker processor’s preset. The different bands will be combined and the total complex response can be analyzed for up to 16 outputs.

Parametric Equalizer simulator

Any captured transfer function can be run through an optimization algorithm that produces a list of parametric EQ parameters (Frequency, Q and Gain) that are required to fit a user-defined target curve. For further optimization EQ parameters can be manually fine-tuned using a convenient graphical user interface. A list of created filters can be transferred to a DSP device or the resulting filters can be listened to or used as stimulus EQ.

Listen to filters

You can use your PC’s multimedia core to run up to 256 parametric filters or any bi-quads (shelving, x-over filters, allpass etc.) and FIR/ FFT convolver filter through your soundcard. You can listen to .WAV files from your hard-drive or assign an input of the soundcard to feed the filters with any program material. You can measure with the filters applied on the stimulus, so you can check your filter setup on and off axis for example.

Measurement

FIR-Capture acquires complex frequency response by applying either a maximum length sequence (MLS) stimulus, Log-sine sweep, Farina sweep (THD sweep) or Dual FFT (using external Wavefiles) to the loudspeaker or sound reinforcement system under test.

MultiWin

Sophisticated windowing functions allow the user to window out room reflections and focus on either equalizing the direct sound while retaining low frequency resolution or spatial averaging of the room transfer function.

Pseudo Averaging

Room-Capture also supports Pseudo Power Averaging, which recreates an average of the time response. PPA allows the user to perform a series of measurements throughout the coverage pattern of the sound reinforcement system and base system EQ on the weighted, spatially averaged response.

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 loudspeaker driver time domain analysis and system analysis of time smearing.

Room Mode finder

Automatic Room Mode finder with automatic calculation of parametric EQ that can be used to remedy dominant room modes or for optimization of installed Helmholz resonators.

Real Time Analyzer

Room-Capture also offers a multi-channel Real Time Analyser with up to 1/48 octave resolution. Besides industry standard FFT based RTA, a high-resolution low frequency band-pass mode is introduced.

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

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.

Features

- Supports FIR, FFT and Multi rate convolver filters
- Magnitude equalization with multiple bands using predefined target curves
- Independent magnitude and group-delay / phase equalization
- Linear-phase brickwall high-pass and low-pass filters with preset slopes and freely adjustable slopes
- Acquires complex frequency response with up to 8 channels simultaneously up to 192k
- MLS (up to 4096k points), Log Sine Sweep (Chirp), THD sweep and Dual FFT (using external Wavefiles)
- 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
- 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/192k sampling, with full duplex (simultaneous play and record) capability

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
- Filter parameters can be edited graphically
- Automatic room mode finder helps to identify and compensate for room resonances

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