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The WaveCapture team proudly introduces benchmark sound reinforcement system optimization software:

# Live-Capture Light

The Ultimate FOH Audio Analysis Tool

Derived from the benchmark Live-Capture Pro, Live-Capture Light is an easy-to-use, PC-based software tool for real-time live sound measurements, optimized for FOH use. Live-Capture Light measures time domain and frequency domain simultaneously. Measurements can be made using program material (music and speech), before or during a performance, with the audience present.

Live-Capture Light is dedicated to sound reinforcement system optimization with a range of unique analysis tools that provide fast and accurate tuning information. The full suite of features includes delay finder with group delay and Cepstrum analysis, Auto EQ finder, sound level logging, and display of reverberation time graphs.

Live-Capture Light uses threaded computing to acquire impulse responses between 0.35 to 11 seconds, and then applies suitably large FFTs to display time domain and frequency domain data at a maximum 23.4 frames-per-second refresh rate. Advanced complex averaging is used in both domains, taking the coherence and phase stability into account. The default resolution is 96 points per octave with a resolution of up to 192 points per octave available.

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 on spatial averaging of the room transfer function.

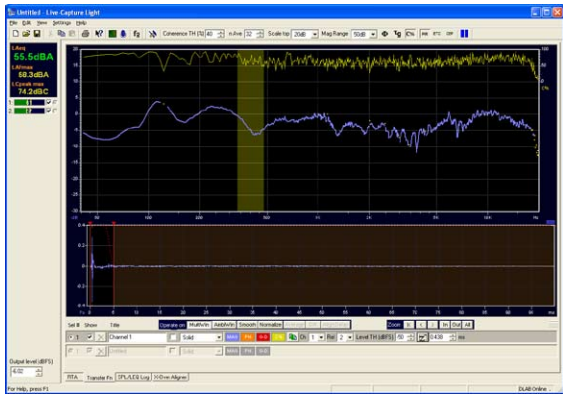
For room tuning and sound reinforcement system optimization, Live-Capture Light supports spatial averaging. This allows the user to perform multiple measurements throughout the coverage pattern of the sound reinforcement system and base system equalisation on the weighted, spatially-averaged response.

The Auto Peak finder can automatically find peaks and dips in the magnitude response, in a predefined frequency range. These peaks or dips can be identified as parametric filters with the Auto EQ tool.

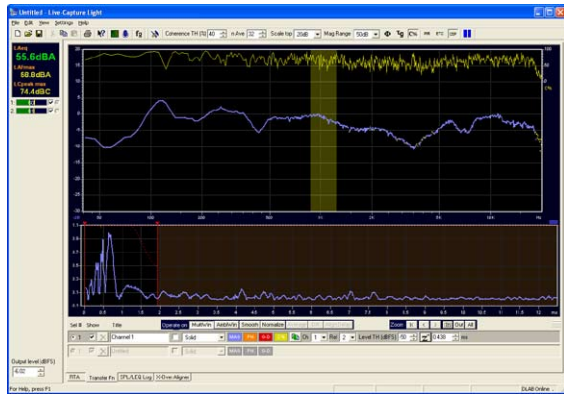
The Auto Delay updater will facilitate the selection of microphone position and compensate for the time variance between positions.

### Features

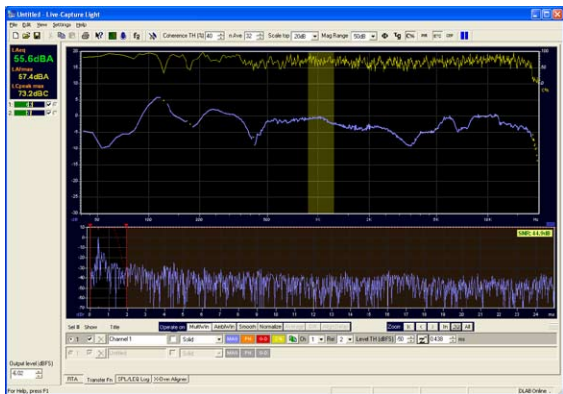
- Real time Magnitude, Phase, Group Delay, PIR, ETC and Cepstrum displays
- Displays both frequency and time domain at high refresh rates in real time
- Complex coherence display including phase stability
- Multiple time windowing for room reflection suppression
- Analysing tools: Delay finder, Auto EQ, SPL and LEQ logging, and Reverberation Time
- The Curve Manager can hold up to 16 captured measurements
- Noise, Sine and Multi-tone generator



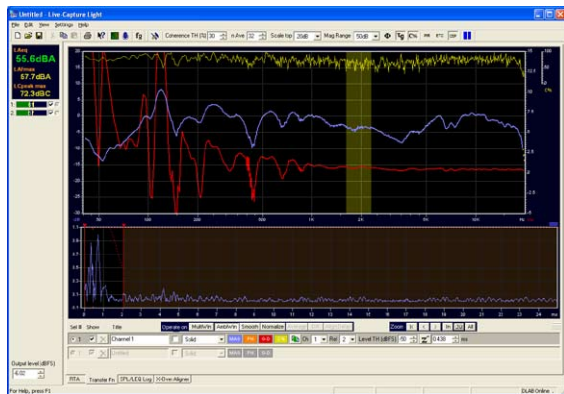
The time graph shows the impulse response with the propriety Multiwin (multiple windowing) applied.



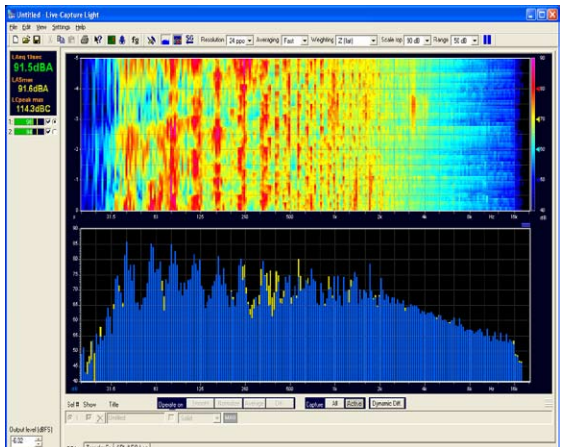
The time graph shows the Cepstrum. Time windowing is shown in real time.



Display shows three captured transfer functions and a spatially-averaged sum. Time graph shows the ETC and the Signal to Noise of the measurement.



Group Delay and wrapped or unwrapped phase can be shown in real time.



RTA with 1 to 24 points per octave. A 2.5 second to 10 second Sonogram is shown simultaneously. Integration is Fast (125 ms), Slow (1 s) or Infinite. 1 to 256 averages.



Sound level logging according to IEC 61672 standard. Graph shows LAFmax, LCpeak and LAeq.

# Live-Capture Light

## The Ultimate FOH Audio Analysis Tool

Measurements use program material (music and speech), before or during a performance, with the audience present.

### Requirements

- PC with Win 2000 SP4 or XP or Vista
- CPU: 1 GHz or faster Intel Pentium 4 or better
- RAM: 512 MB min, 1 GB recommended
- Display: minimum 1024 x 768 pixels, 16 bit color
- Soundcard: Windows compatible (Wave/WDM or ASIO) with stereo inputs, 16-bit/44.1k to 24bit/96k sampling, with full duplex (simultaneous play and record) capability. 1024 samples is the minimum required buffer size

**WaveCapture** a Bävholm/  
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