Real time measurements - Theory and praxis

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Abstract

To measure impulse responses in rooms and free fields is a daily job for acousticians but mainly this is done in unoccupied cases. Results for the occupied case are derived by use of simulation software or by estimations based on experience. Using a newly developed multithread algorithm, speech, music or any other signals from a microphone input and from a mixing console can be utilized to obtain impulse response data for further evaluation. New noise suppression methods and a new developed TFC window are presented that allow these impulse responses to be acquired in full-length even in occupied venues. Practical demonstrations with usual but also unusual sound sources are done. Required measuring conditions and limitations are explained.

1. What is the relevant state of the technology?

From system theory is known:
If a signal e (t) is fed into a test device DUT (electronic device, but also into test room over loudspeaker or other sources), the impulse response h (t) of the DUT with this input signal "convolves" and one receives an output signal a (t) (it is picked up in rooms with the microphone). If the input and output signals and the still unknown impulse response in request of the DUT are Fourier-transformed, a simple multiplication happens in the frequency domain:

$A(\omega) = E(\omega) \cdot H(\omega)$ and after rearrangement $H(\omega) = A(\omega) \cdot E(\omega)^{-1}$

Then by an IFFT the impulse response h(t) can be easily determined.
As input signals besides noise signals, also MLS sequences, Sweeps, but also speech and music signals are used, cf. Fig. 1.

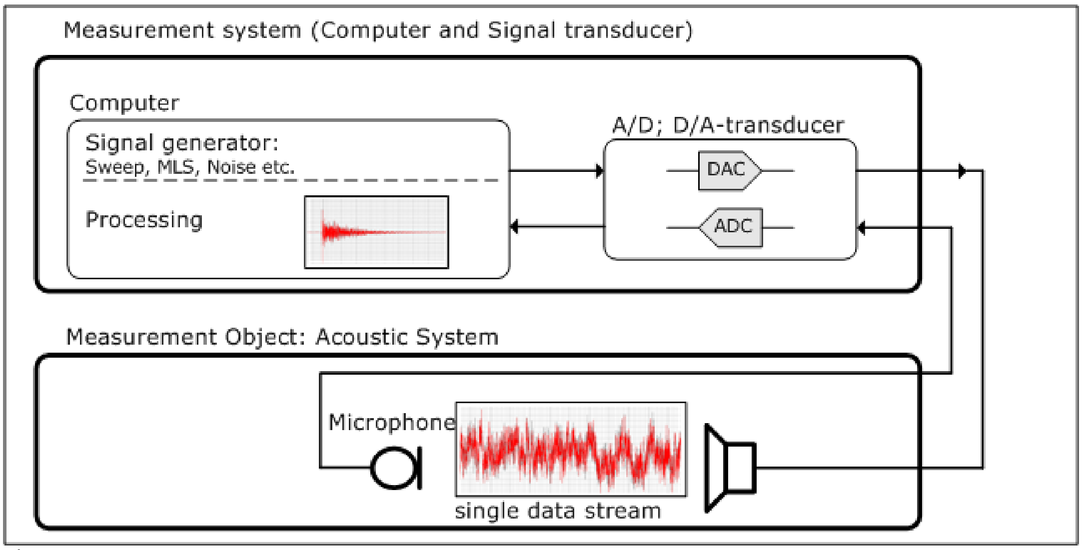


Fig. 1

However, to all these measuring procedures is common that the DUT is stimulated first audibly and then the output signal is measured. Then in the post processing and mostly in the lab later, but also immediately after conclusion of the measurement, the impulse response is calculated. Also the

complex transfer function can be determined, and then the above-mentioned IFFT back into the time domain may be dropped /1/,/2/.
Beyond that, for nearly 20 years a source independent measuring procedure SIM is also known where by use of costly hardware and the application of Fourier transformations in real time complex transfer functions are derived containing magnitude and phase, however, no complex impulse responses are obtained with lengths according to longer reverberation times in the rooms. SIM serves first of all for the setting of magnitudes and frequency responses of loudspeaker and array arrangements and not for room acoustic investigations in rooms or in the open air /3/.

2. Which disadvantages or problems exist with the relevant state of the technology?

With electric measurements the signal excitation with Noise or Sweep does often not disturb, probably, however, with acoustic measurements in rooms or outside. Here rehearsals are interrupted or shortened, only for tuning the systems. It is even more critical if measurements are to be carried out in the presence of audience. Then such play-ins of Noise or even Sweeps interfere of course, so that important intelligibility measurements, e.g., in stadiums, happen mostly only in the empty state. However, the values received this way are right only partly, because we know how strongly audience changes the acoustic properties of a room or even an open stadium. Then often extrapolations are done on different occupation degrees which can be, however, naturally strongly defective.
It would be better therefore in this case if the visitors would not note that any acoustic measurements are carried out. The already described measuring technology SIM permits the loudspeaker-related commissioning in this respect, so that frequency and phase responses are adjusted. Time responses are thereby derived immediately to setting up delays (delay finder option). Acoustic measures, which are normally derived from the impulse response directly, however cannot be shown in real time. Also no dynamic displays are possible which make transparent the dependence of these ascertained parameters from the local measurement position. Therefore the so-called SIM procedure produces in real time local information to magnitude and phase responses and that in the time and the frequency domain, in each case, however, at a previously defined measuring position.
The measuring system SMAART of the company SIASoft delivers results better usable for room acoustic post processing. Here the impulse response is derived by means of a dual-channel measurement, while from the transfer function with static excitation of both channels by an IFFT the time function (max. length approx. 11s) is calculated. Every time the function is calculated individually and needs an actual excitation. In a dynamic way, or on-line, only the transfer function in the frequency domain is derived. A conversion into a time function is not possible /4/.

3. Background of the new solution

The method described above starts to determine first an impulse response by recording a so-called raw signal and later follows his independent analysis.
Now the essential invention thought consists in the fact that based on the most modern state of technology it is possible to construct a PC-based measuring system which can determine a room impulse response (RIR) in real time and in full length by deconvolution (Real Time Deconvolution RTD), cf. Fig. 2.

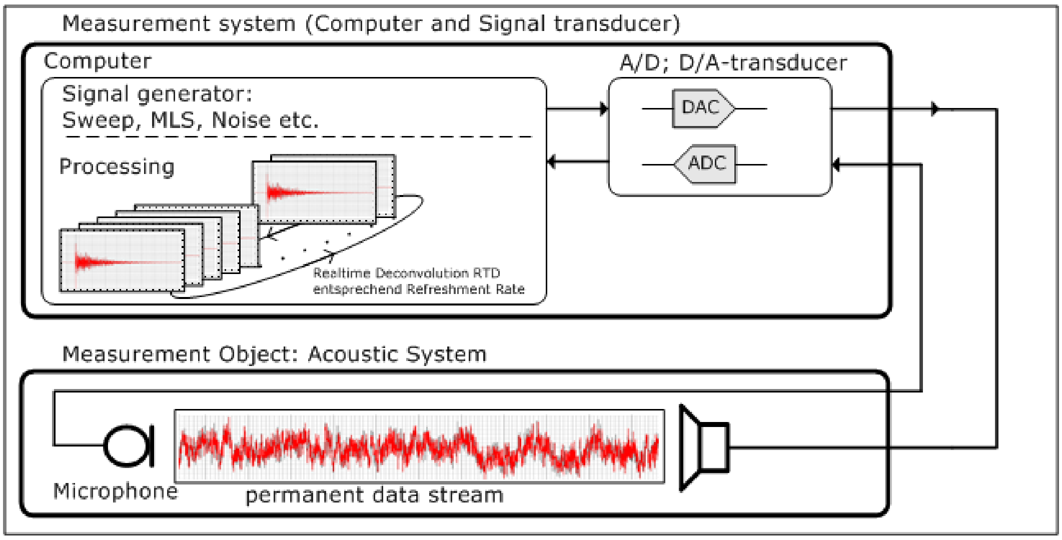
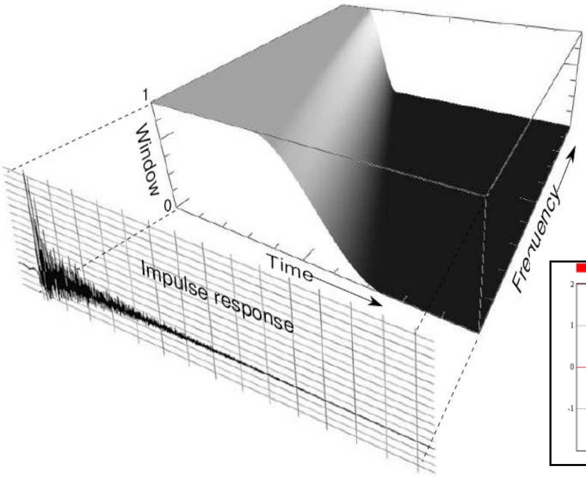


Fig. 2

The dynamically ascertained RIR is - because of a number of optimized evaluation procedures - qualitatively absolutely equivalent to a statically ascertained RIR. The RIR can have typical lengths, e.g., 4 s - 8 s. The transformation between the frequency and time range occurs linear and in full length, analogously to the static procedure, however, there it is known only as a one-time process. The RIR is calculated without data loss by e.g. time windowing or compression. All electric-acoustic and room-acoustic measures can be derived from the dynamically ascertained RIR like in the static case. Moreover, an averaging can be likewise used to suppress the noise. A newly developed Time-Frequency-Constant (TFC) Window (s. Fig. 3) provides additional filtering functions for the analysis.



TFC window™:
Excludes late energy at high frequencies but includes energy at low frequencies and this continuously over the frequency range

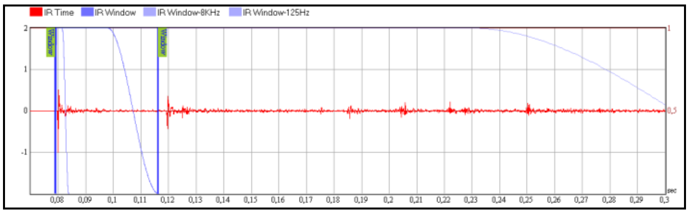


Fig. 3

The real time ability of the measuring system presents itself in such a way that the system can keep to very high refresh rates for the calculation of results and their display and analysis. In addition, the system can switch in real time between various result presentations without data loss. Further an interaction with the system is possible at any time during the measurement. In the end, changes of the measured object can be pursued in real time, i.e. within the running measurement. Expressed in a simple way one can understand the measuring system also as an "oscilloscope for room impulse responses ", which was not possible this way until now.
For the procedure an efficient memory and resource management is essential to allow running on a normal PC generally. That means that only results are calculated which should be also presented. Only memory is requested for the data that should be also presented. The whole remaining math power and the available memory is used dynamically to achieve very high refreshment rates (approx. 10 /sec). These algorithms have been implemented in the new measurement platform EASERA SysTune.

To obtain a complete IR a reference signal defined over the full audio spectrum is needed. For measurements with live music or even speech it is improbable to have broadband-signal all the time, so as a solution, the FFT-size or the number of FFT-averages could be raised. In this example it is not necessary to do this, because for measurements of arrival times of direct-sound even the spectrum of speech gives enough data for an IR, so it is possible to use quite small FFT-sizes and get fast refresh-rates, which means fast results and less waiting time.

To get better results, there is a tool inside EASERA SysTune to interrupt the processing during pauses in the reference signal (“skip noise in reference signal”), see in Figure 4 the working principle of the noise skipping feature in EASERA SysTune. Additionally for suppressing unwanted noise (e.g. humming of air condition systems) in broadband displays EASERA SysTune provides an adjustable band pass-filter.

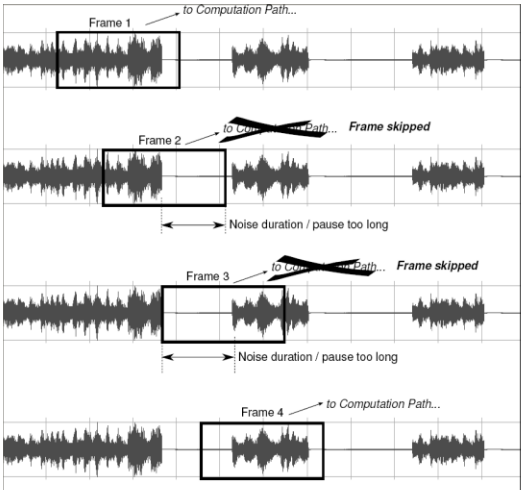


Fig. 4

4. **Practical Measurements in a stadium**

In simulation programs predictions are done for an empty facility, for a full occupied one or even for 50% of occupancy. But how this will be done with measurements until now? To measure the reverberation time or the speech intelligibility in an empty stadium is simple and standard now. But how to do it in a stadium in the presence of 50.000 to 70.000 visitors? In context with the soccer world championships 2006 in Germany a couple of stadiums have been built and after construction of the sound systems tuning measurements have been done. Until now the measurements have been done in the empty stadiums, see fig. 5 for the Allianz arena in Munich.

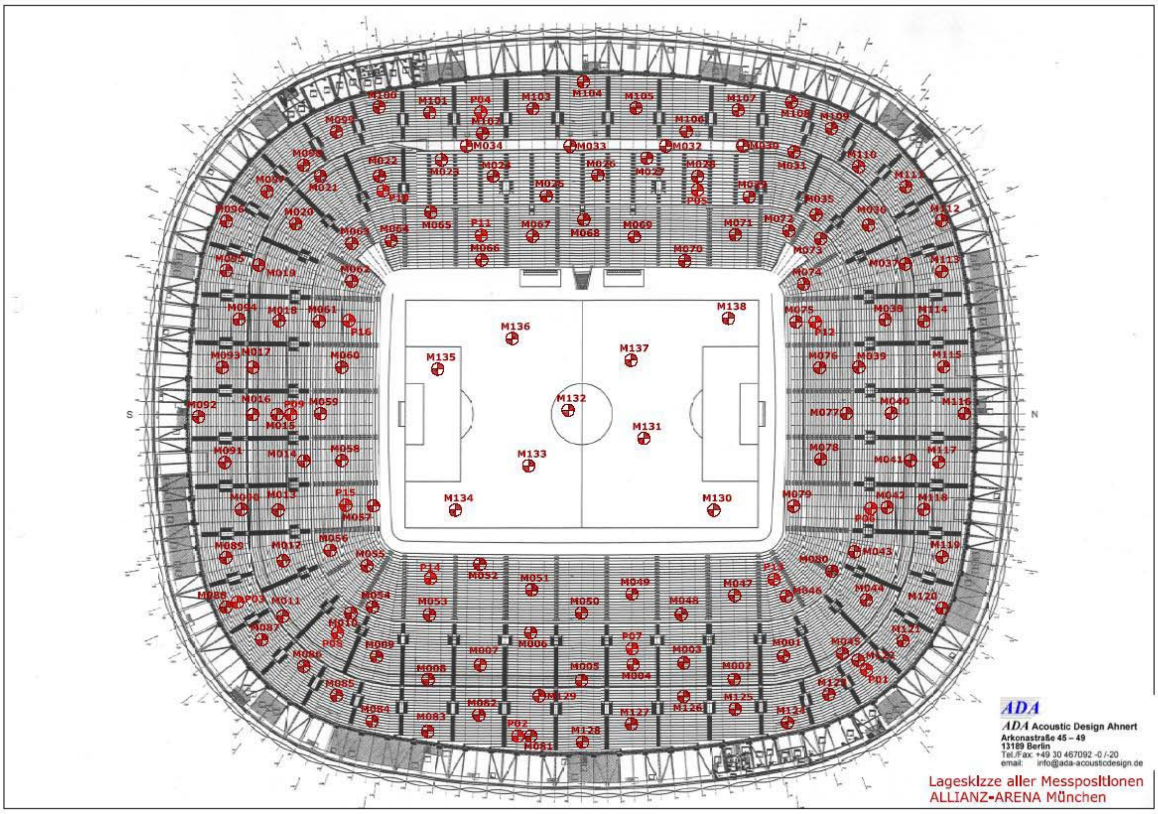


Fig. 5

The measured values from the empty arena have been entered in a prediction program (in our case EASE 4.2) and then the occupied values have been derived. So from the empty case has been concluded to the occupied one, even for different levels of occupancy.
Fig. 6 shows the measured results for two different cases in the empty stadium (with and without a sun sail covering upper stadium opening) and the extrapolation for 3 different cases of occupancy. Of course such extrapolations must always be considered with caution.

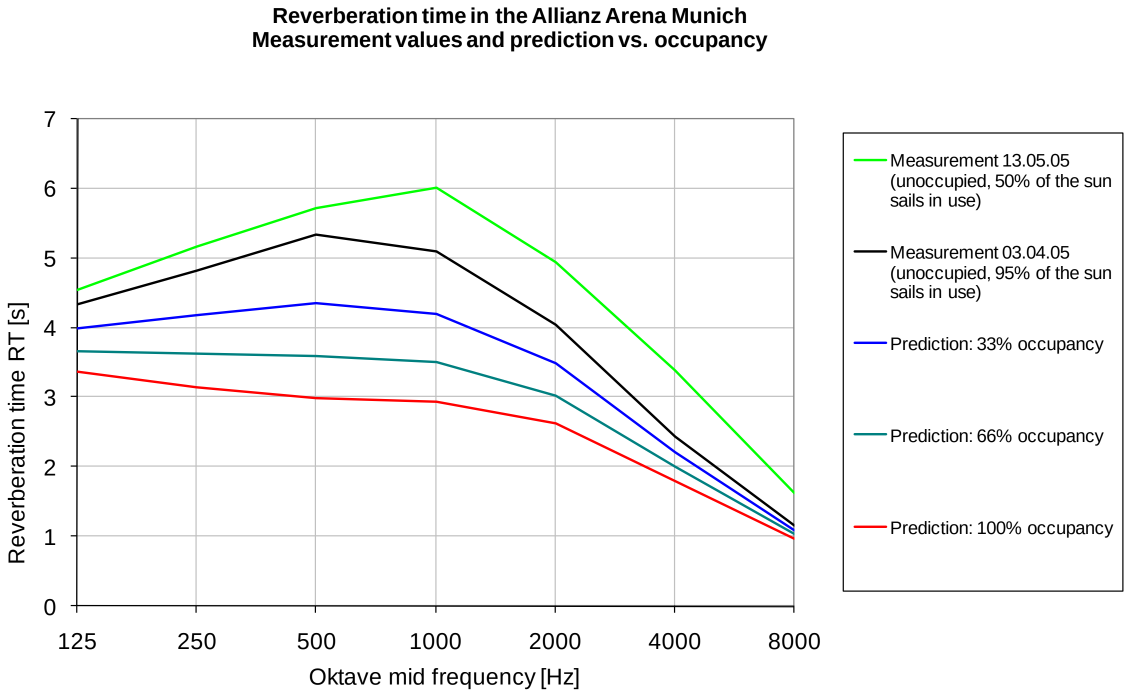


Fig. 6

By using EASERA SysTune in 2007 first measurements have been done in a stadium (Mönchen-gladbach) in the empty case as well as in the occupied one (55.000 visitors during a soccer game). Here the following block diagram has been used, fig. 7.

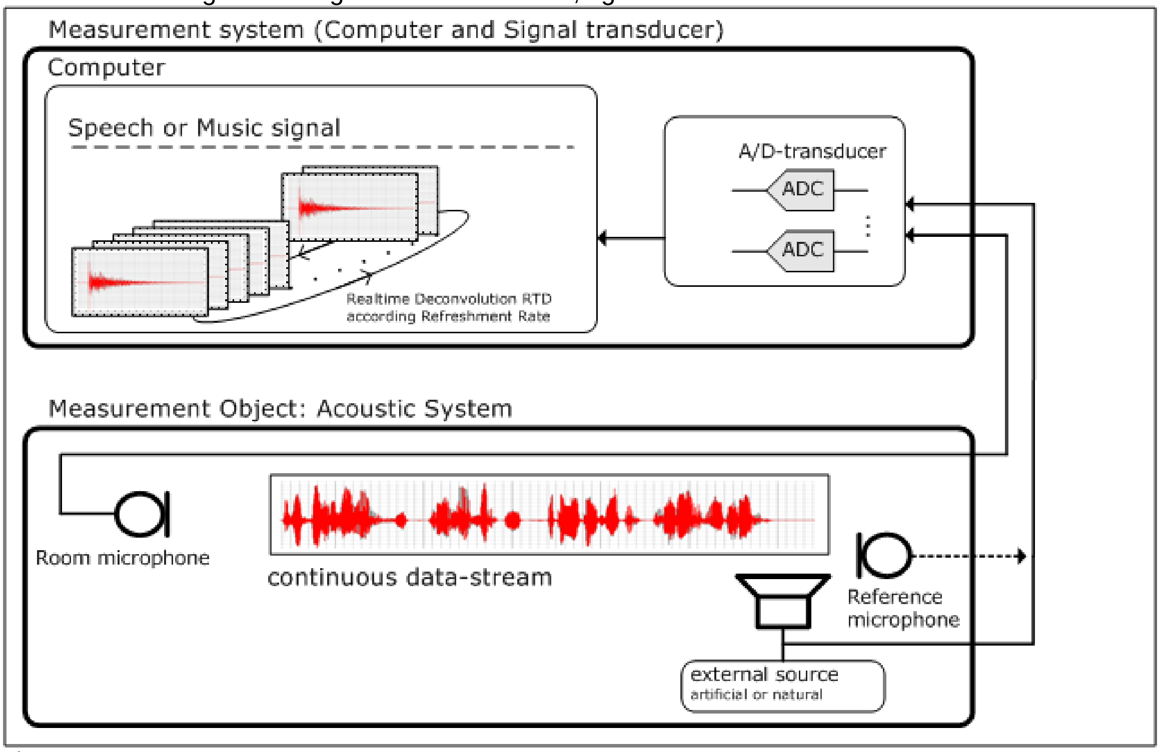


Fig. 7

The measurement setup has been installed in the control booth of the stadium, see fig. 8. As an external excitation signal the usual advertisement and information signals of the stadium (fig. 9) have been used. Therefore these signals controlled by the mixing console have not only radiated over the stadium sound system but also as fig. 7 shows have been used as a reference signal for SysTune (channel 2). In the stadium a mobile wireless microphone had been used and its signal was fed to channel 1 of the measurement tool. For these first measurements in a stadium only one wireless microphone has been used, but 6 other can be installed parallel because SysTune has altogether 8 inputs.

Fig. 8 (EASERA Systune computer on the right)

Fig. 9 (Stadium in Mönchengladbach)

The results of the measurements in the empty one (measured with Sweep excitation) and the occupied one (measured with speech and music signals) are shown in fig. 10.

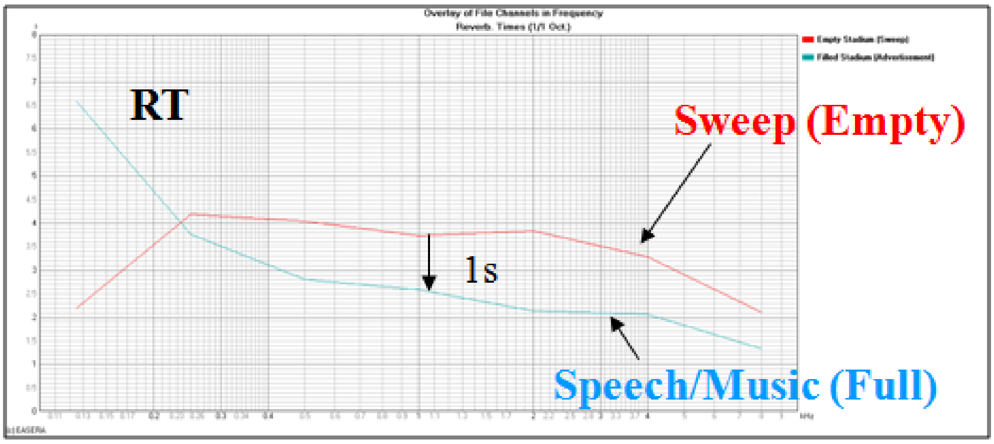


Fig. 10

Caused by the audience the Reverberation time derived from the measured impulse responses is dropped by around 1 sec. Accordingly the speech intelligibility is increased significantly. Note that for the 125 Hz octave band there was not enough S/N in the measurement to generate useful information.

During the SysTune measurements in occupied spaces a couple of problems have been found:

- Open Space and moving people cause high noise floor
- The Changing Source Material and the pauses produce distortions and wrong reference signals
- A Long measuring time must be combined with more averages

5. Practical demonstrations

6. References

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/3/ SIM Audio Analyzer, http://www.meyersound.com/products/#sim, Meyer Sound, Berkeley, CA, USA; www.meyersound.com

/4/ Smaart Software, http://www.eaw.com/products/software/index.html, Eastern Acoustic Works, Whitinsville, MA, USA, www.eaw.com