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SIA-Smaart® Pro Analysis Module Case Study #1: Domed Stadium

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Hi all,

I ran into an interesting acoustical measurement problem recently and thought it might make a good example for our web site. I was asked to measure and evaluate the acoustics and sound system performance of a large major league baseball/football domed stadium (in the past few years I have been involved in the system setup and acoustic design of a large number of sports facilities). I used Smaart Pro to measure both transfer functions and impulse responses.

To measure the complete impulse response in such a reverberant room, I used a 131072 point FFT and *lowered* the sampling rate from my normal 44100 Hz to 11025 Hz in the Delay Locator. The reason for this is something ALL Smaart users should know: **the delay locator requires a time constant that is about the same length as (or longer than) the decay time of the system under test.** The FFT Time Constant is expressed in seconds and is calculated as the FFT size divided by the sampling rate. The domed stadium is extremely reverberant so I selected a large FFT and lowered the sampling rate, to achieve an FFT time constant of 11.888 seconds! The price that one pays for this is that all information above 5512 Hz is lost (due to the Nyquist rule).

This measurement was made using pink noise as a reference signal with the microphone at the front portion of the upper seating deck. The stadium was empty and all of the speaker clusters on.

To analyze this wave file I used the Smaart analysis module and displayed the data on a logarithmic amplitude scale. I then zoomed in (reduced the time scale) and saved the shortened wave file using the *Save Wave Zoom As* feature (in the Analysis module's *File* menu). Figure #1 shows the measured result with annotations added using Microsoft Word's "Picture Edit" commands.

Figure #1 (on the next page), displays a signal that represents the "sound" or "signature" of the Stadium's acoustics. Called an *impulse response*, this signal contains information about both reverberation and individual reflections. The x-axis is time (in seconds) and the y-axis is magnitude (in decibels). Figure #1 represents a broadband measurement, which means (in this case) that information about energy at all frequencies up to about 5.5k is contained within the impulse response.

The highest peak in plot in Figure #1, represents the “direct sound.” This is the sound that travels directly from the loudspeaker to the listener (or in this case, a microphone). The arrival of the *direct sound* is followed by a few *reflections* then a build-up of *reverberant energy*. Finally, the reverberant decay can be seen dropping away into the noise floor.

Several quantities can be calculated from the impulse response to help evaluate sound system performance. One such parameter is the ratio of the level of the direct sound to the level of the reverberant energy. As a general guideline, to achieve intelligible speech in a public assembly space, a broadband direct-to-reverberant ratio of more than 15 dB is desirable.

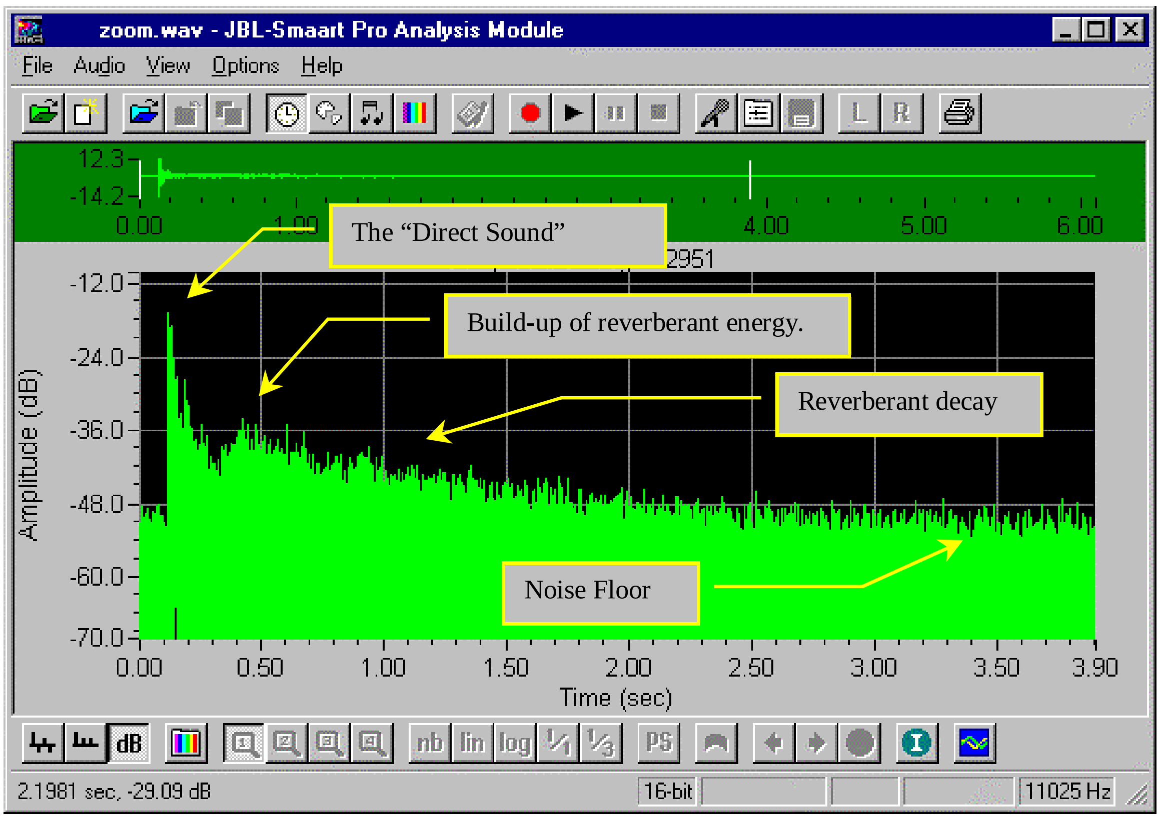

Figure #1: An impulse response measured in the 3rd row of the Upper level seating. Full system was on and the stadium was empty.

To continue my analysis of the room’s response, I used the Analysis module’s *Transform* command to calculate a set of FFT frames. I selected a 1024 point FFT, (which provides 10.8 Hz resolution for data sampled at 11025 Hz) with an 80% *Overlap* between frames. Lastly I used a Hanning data window (the window type I almost always use).

To look at the frequency domain data, I generally take a quick look at the spectrograph to see if anything looks usual, and then use the *Time Slice* view mode, to see how the energy in the impulse changes over time for various frequencies.

Figure #2 (below) is a plot of just the energy in the octave centered on 1000 Hz within the same impulse response. 1000 Hz (or 1 kHz) is in the upper portion of the frequency range for speech. The level of the direct to reverberant ratio can be seen to be approx. 12 dB. This is an acceptable (if marginally low) value.

The rate of reverberant decay can be also measured in Time Slice mode (by drawing a line along the slope using your mouse). For this room reverberant decay rate is approximately 6 dB per second. This decay rate equates to a broadband reverberation time (or RT60) of approximately 10 seconds!!!!

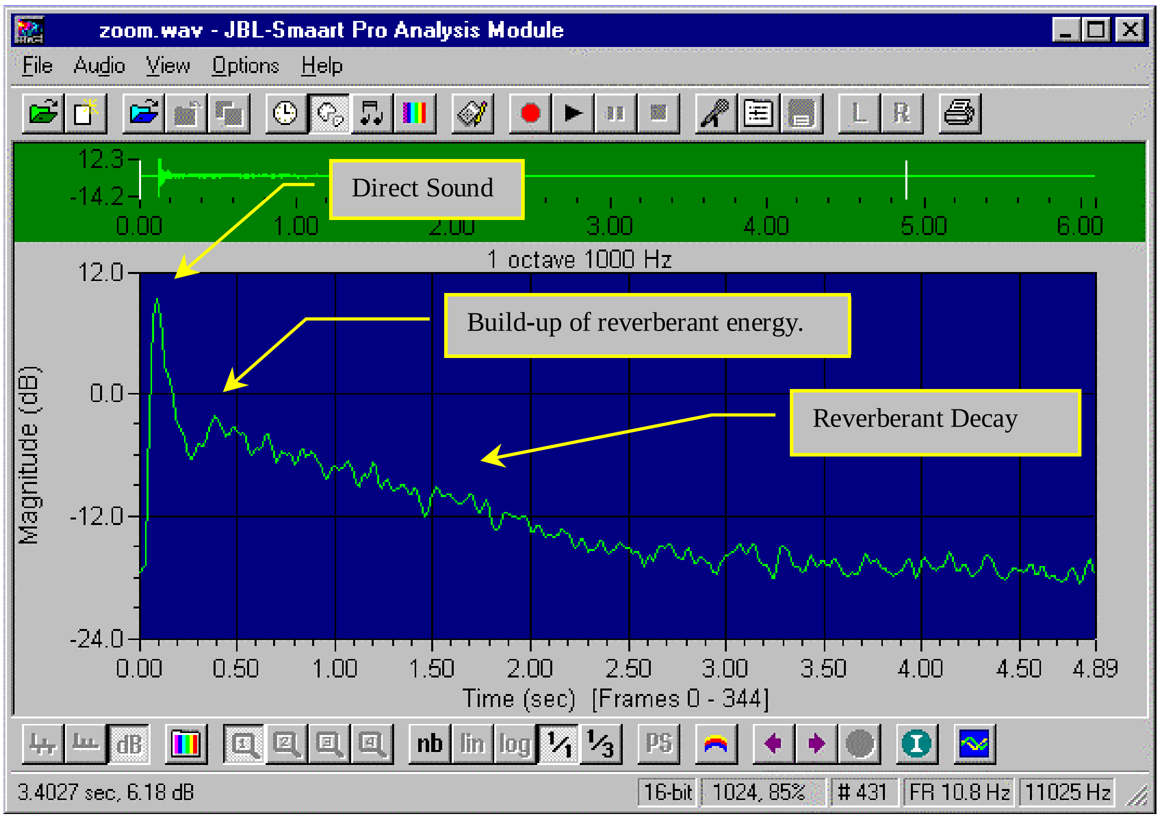

Figure #2: The 1 kHz energy contained within the impulse response displayed in Figure #1.

Figure #3 (below) displays the 250 Hz energy of the same impulse response shown in Figure #1. 250 Hz is a fundamental frequency in male speech. Notice that the direct to reverberant ratio has decreased to the point that the level direct sound is not much higher than the reverberant level).

Also notice that the decay at 250 Hz appears to be much more erratic (less smooth) than the decay at higher frequencies. The decrease in the direct-to-reverberant ratio can be understood as a resulting from the decrease in directionality of the Dome's loudspeakers at 250 Hz when compared to higher frequencies. This serves to illustrate how the ability to control the directionality of sound can help to reduce the amount of energy that can excite (or stimulate) the reverberant sound field.

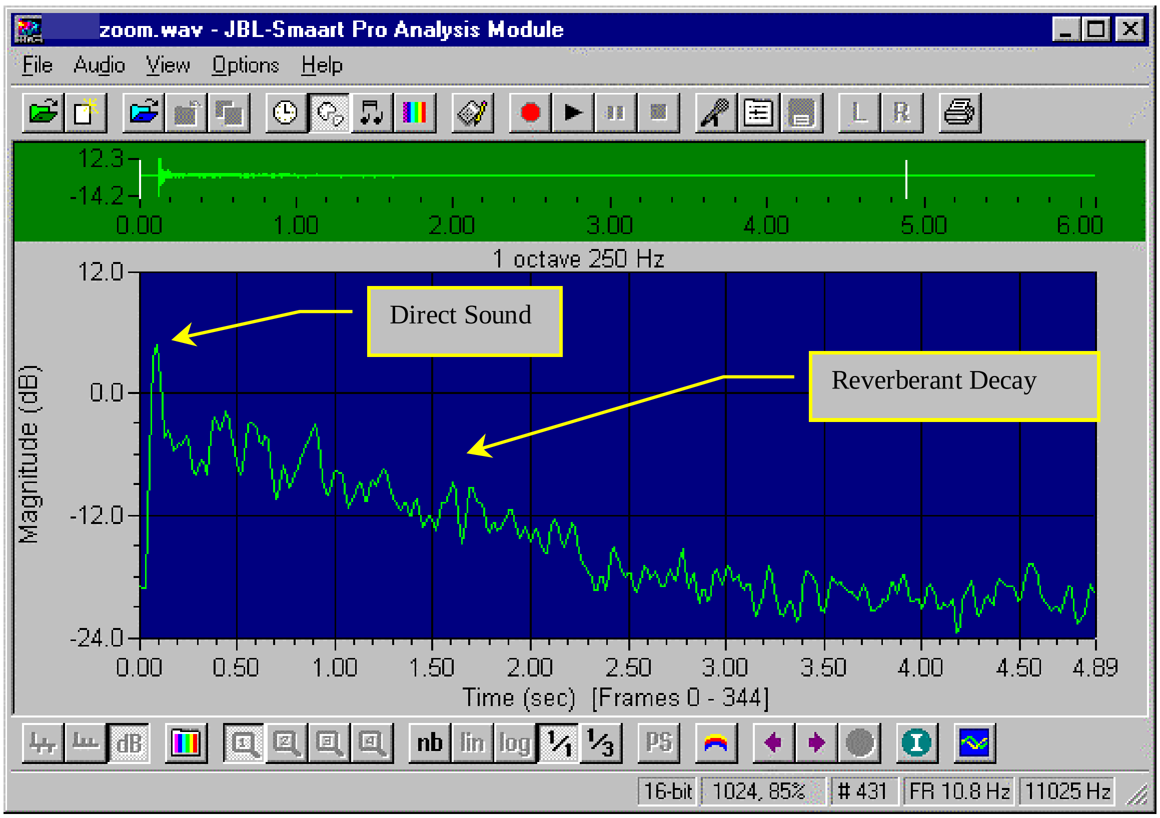

Figure #3: The 250 Hz energy contained within the impulse response displayed in Figure #1.

I hope this example demonstrates both how to use the some of the features in Smaart's Analysis module and that a significant amount of information can be gained by studying the impulse response of a system.

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