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SIA-Smaart® Pro Real Time and Analysis Module Case Study #2: Critical Listening Room – Home Theater

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Hi all,

For the second in a hopefully ongoing series of case studies, I have selected a small room designed for both critical listening and home theater. This room is equipped with high-end listening equipment and state of the art home theater projection… however acoustical concerns have raised a number of issues. I was asked to evaluate the room. I was only told that the owners were not happy with the sound of the room. Here's what I did, some results and some interpretation of these results.

To begin my evaluation I studied a drawing of the room, which is essentially rectangular with the following dimensions: Length – 26'-0", Width - 15'- 4", and Height – 9'-0". The room had a carpeted floor, a hard ceiling treatment with diffusive elements and some modest (i.e. thin) acoustical treatment on the walls.

Before making any measurements I spent a few minutes listening to music! I believe that this is a critical step, never underestimate your ears! This year I am carrying several discs for critical listening. My favorite of the moment is Shawn Colvin's "Steady On". Track 1, starts with a kick drum and bell. The kick has a couple of "double-taps" that will often disappear in rooms with excess LF reverberance. Also it is easy to hear if the drum or bell are out of balance, which I find to be a great reference.

The first measurement I made was an impulse response. This allows me to find both the delay value required for making transfer function measurements and to see any potentially disturbing reflections or uneven reverberant decay. To make this measurement I set up a small measurement mixer that feeds its left and right output into my computer and I sent a line level output on an aux bus to <u>*one*</u> of the stereo loudspeakers in the room. I used pink noise for my first measurement.

To make the impulse response I used Smaart's Delay Locator feature. I selected a 44100 sampling rate, a 65536-point FFT and 5 averages. This sampling rate and FFT size resulted in a FFT time constant of 1.48seconds. After making the measurement, I clicked the Analysis button in the Delay Locator to load the file into Smaart Pro's analysis module. The result is show on the next page in Figure #1.

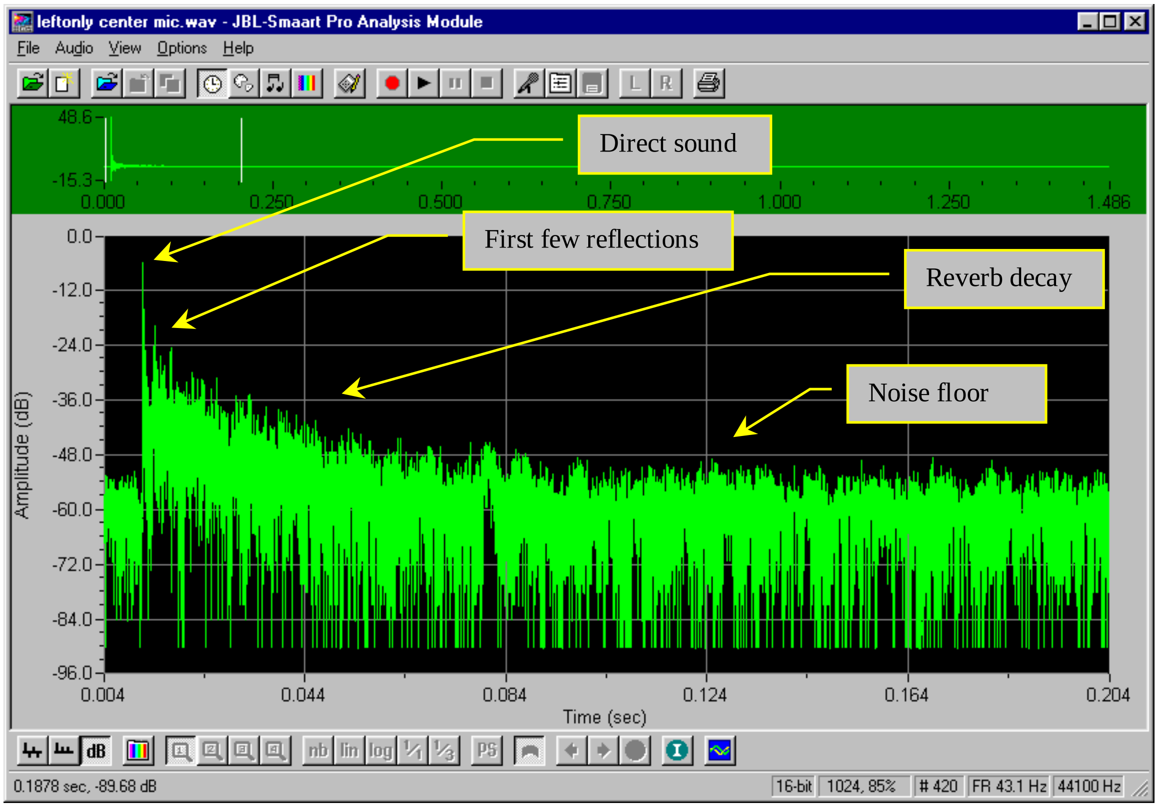

Figure #1: An impulse response made with a pink noise reference signal. The microphone was placed at a central listening position in the room at ear height.

A quick glance at the impulse response shows a fairly typical result. The direct sound can be clearly seen, followed by a few discrete reflections, and a reverberant decay. However with a microphone only 11msec from the loudspeaker one might expect a slightly greater broadband direct-to-reverberant energy ratio. None of the reflections seem obviously problematic, however the first two are slightly higher than might be desirable.

Figure #2 (shown on the next page) shows the transfer function measured from the same point as the impulse response measurement above. The transfer function was made in Fixed-Point Per Octave resolution, and saved in a reference file. Thirty-five (35) averages were used, with pink noise as the source signal.

Several elements of the transfer function plot stand out:

a) The large resonances (positive peaks) at low frequencies
b) The dip at 600 Hz
c) The roll-off above 5Khz.

In general, the first item; the Low Frequency resonances, are the most likely to be acoustic in origin. At 600 Hz the dip could reasonably be due to a "floor bounce" reflection in the measurement or perhaps a crossover problem in the loudspeaker. The last item (the 5k roll-off) is the least likely to be acoustic and, as a first guess, probably attributable to a relative gain or driver efficiency problem. I took the following steps to investigate the first problem, the low frequency resonances:

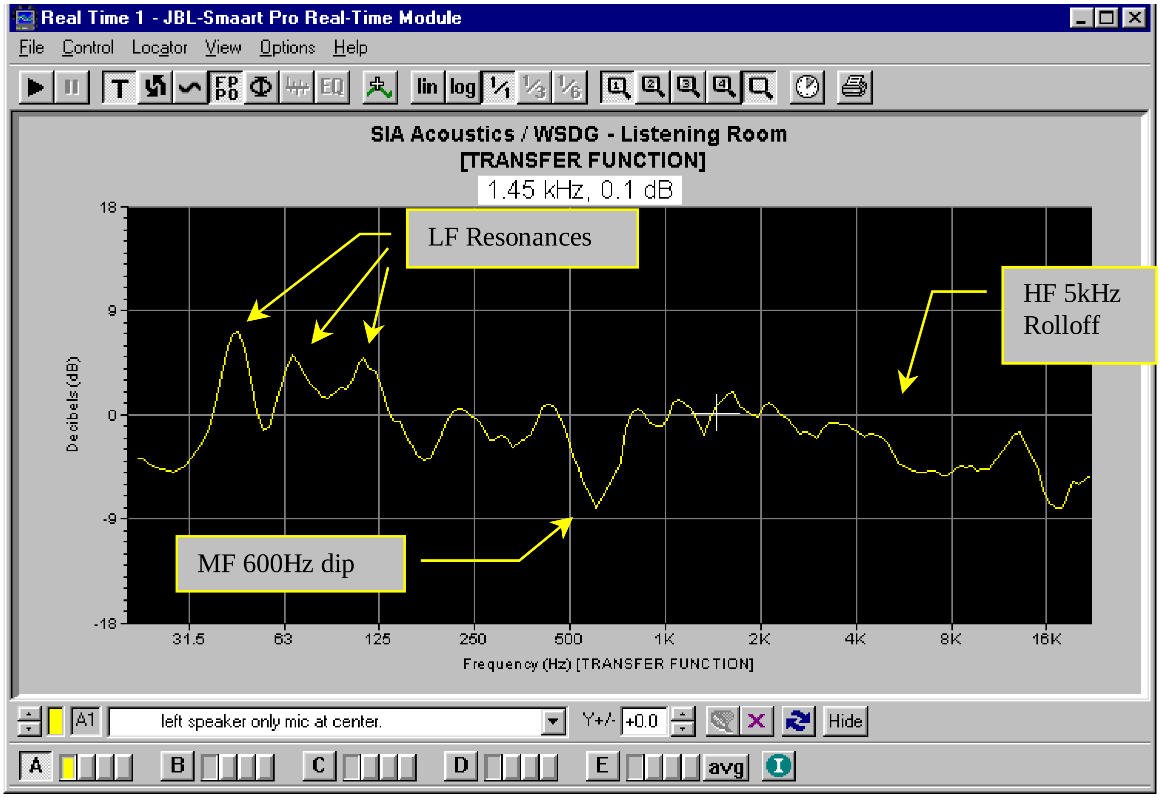

Figure #2: A transfer function measured at the same location as the impulse in Figure #1. The measurement was made in Fixed-Point Per Octave Resolution using 35 averages. Pink Noise was used as the source signal, however music would yield the same result. On this type of plot, values above the 0 dB line represent amplification or resonance. Values below 0 dB represent either some type of attenuation or cancellation.

To study the low frequency resonances I first calculated the expected modal frequencies of the room. This is a simple calculation, which is relevant only in well-shaped rectangular small rooms (no balconies please)! Given the length, width and height of a rectangular room, expected mode frequencies can be calculated using the following equations:

P_i = (565/Length) * i, Q_i = (565/Width) * i, R_i = (565/Height) * i

"i"	P modes (from length, Hz)	Q modes (from Width, Hz)	R modes (from Height, Hz)
1	21.7	36.9	**62.8**
2	**43.5**	73.9	125.6
3	**65.2**	**110.8**	188.3
4	86.9	184.6	251.1
5	**108.7**	221.6	313.9

In our case, the rooms dimensions are: Length = 26.0 ft., Width = 15.3 ft., Height = 9.0 ft. The value 565 in the equations comes from one half the speed of sound ~1130 ft/sec.

A careful look at Figure #2 (or the ref file) reveals that the low frequency resonances occur at 43 Hz, 66 Hz and 109 Hz. It is extremely interesting to note that these frequencies also appear in the predicted P modes (associated with the length of the room). This correlation would suggest that the loudspeaker is exciting modes in the room. This result also seems to be in agreement with our earlier observation that only thin absorptive materials were used on the walls. Thin materials often have poor low-frequency performance — particularly below 200 Hz. To verify this I transformed the impulse response (from Figure #1) into the frequency domain and looked at the both the Time Slice and Spectrograph displays in the Analysis Module. I selected a 1024-point FFT with 85% overlap for my transform parameters.

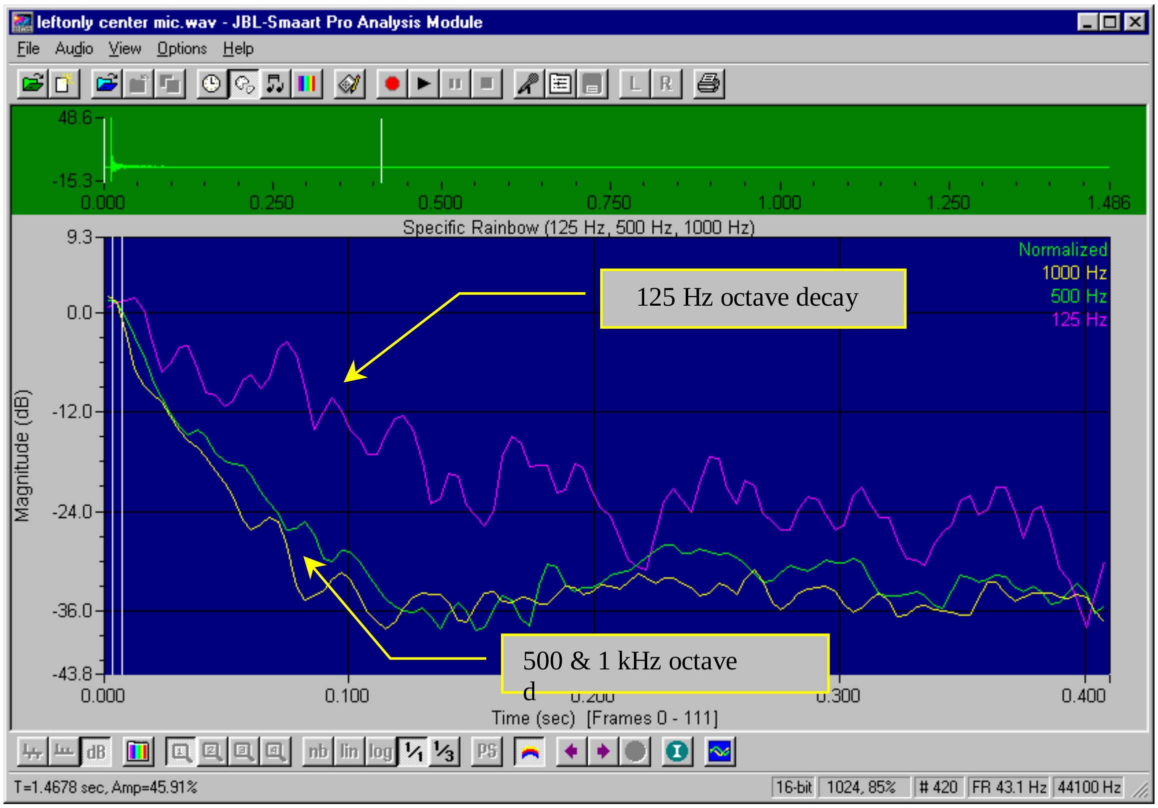

Figure #3: The Rainbow Time Slice display. This display shows the decay of sound in the room for three octave bands centered on 1000 Hz, 500 Hz and 125 Hz. The x-axis is in decibels and the y-axis is in seconds. Notice that the 125 Hz band decays much more slowly and less smoothly than either the 1kHz or 500 Hz bands.

A disparity between the decay rates of mid-high frequencies and low frequencies is expected as most absorbing materials are more effective at higher frequencies. However in this case the decay rate at 125 Hz is less than half the decay rate at 500 Hz (the *reverb time** at 500 Hz was measured to be ~0.2 sec, while the decay at 125 Hz was determined to be ~0.55 seconds). I used my cursor to find the decay rate for each frequency range.

To further investigate the low frequency problems (did I mention that upon my first listening I was struck by excessively “tubby” low frequency sound?), I looked that the Spectrograph. The Spectrograph is a 3D plot, with time on x-axis, frequency on the y-axis and amplitude displayed by color.

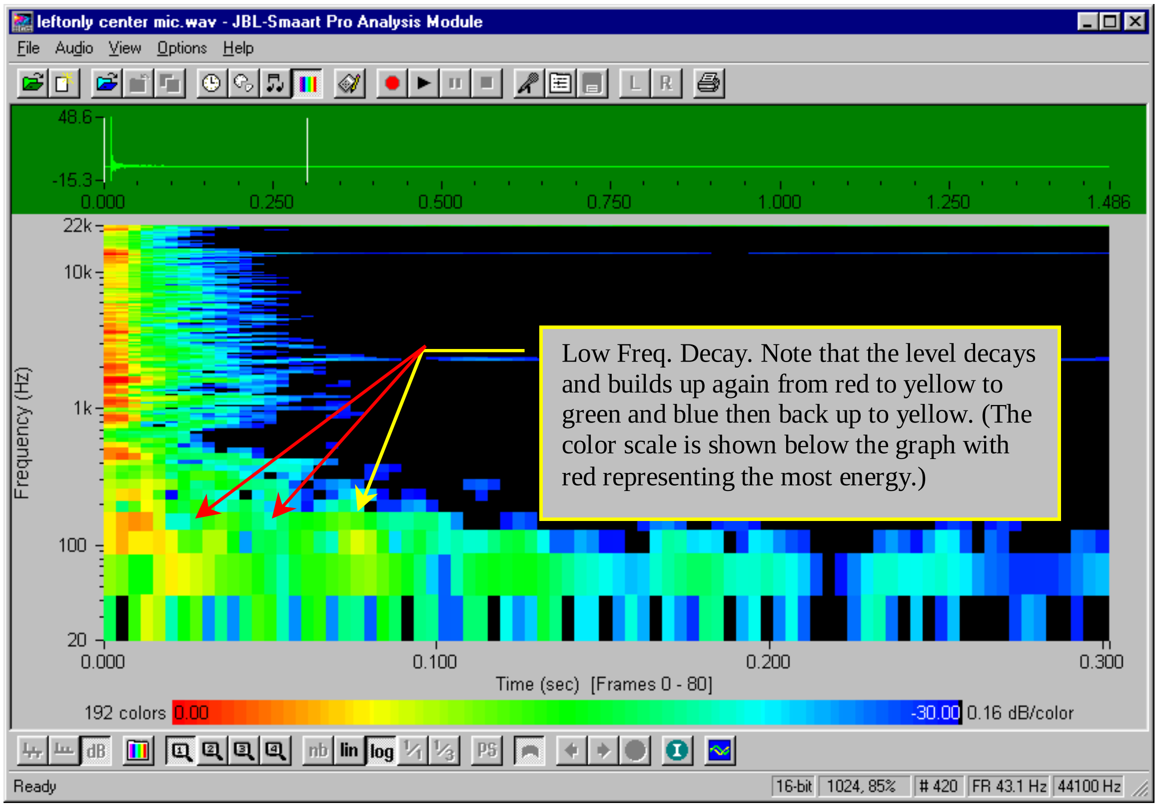

Figure #4: The Spectrograph Display. This display shows the decay of sound in the room for frequencies from 20 Hz to 22kHz. The x-axis is time (in seconds) and the y-axis is a logarithmic frequency scale. The red color represents the highest level (magnitude). Notice that the low frequency energy decays much more slowly and unevenly than the higher frequencies.

Once we had looked at the transfer function, the Time Slice and the Spectrograph, in conjunction with my earlier listening, we (the infamous John Storyk and I made these measurements together) concluded that this room suffered from a lack of effective low-frequency absorption. In fact, this is a problem in many small listening rooms, home theaters and control rooms. The fact that the three big resonances in the low frequency range all appeared in the predicted room modes associated with the length parameter suggests that adding low frequency absorption or diffusion to the walls perpendicular to the long axis of the room might be an effective treatment.

One important note is due here, I used Smaart in this case to help identify a problem in a room. I think that the results strongly suggest that the room was behaving modally. Further, the measured results, when compared to a simple prediction of modal frequencies, gave extremely similar results! While I think that this is a very valuable process, I would like to point out that the a complete solution includes selecting and detailing the proper type, amount, and installation details for any proposed treatment. Aesthetics, cost and durability considerations will also play a role in finally deciding how to treat this room.

Once again, I hope you have found this case study helpful. I have included one small footnote below addressing the use of the term reverberation time in small rooms. This is suggested reading for acoustics geeks (like me) only!!!

* The term reverberation time is extremely common and in fact not very useful! The parameter is best applied in rooms where a diffuse sound field exists. Unfortunately this is NOT the case in most control rooms and home theaters, and even many concert halls! Reverberation Time or RT60 is a very strange parameter as it is a relationship between volume and absorption — however it has little correlation with subjective preference. I like to point out that rooms with identical reverberation time spectras (reverb time at various frequency bands) can sound radically different due to the distribution of reflections. So in closing this short tirade, let me say that Reverberation Time should be treated as one of a large number of pieces to the acoustics puzzle (and if fact not a very important one).

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