# SIA Software Company, Inc.

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## SIA-Smaart® Pro Case Study #5: Live Music Show in a Midsize Auditorium

*By Don Pearson, Ultra Sound/ProMedia*

I recently worked a live rock music show in a 1050-seat auditorium. As system engineer for the show, this venue presented me with a number of challenges. The main room is approximately 100 feet in length and 60 feet wide with a 30-foot ceiling. A balcony extends along the back and (house) left walls. Along the (house) left wall there are openings to bar areas on both the main floor and the balcony. The stage runs nearly the entire width of one end of the room, extending approximately 25 feet into the room with an 8-foot fire aisle on the (house) right side.

On the main floor, carpeting extends about 20 feet into the room along the back and side walls leaving a large uncarpeted (i.e., highly reflective) area, approximately 50 feet by 20 feet, directly in front of the stage as a dance floor. The stage itself is poorly braced and lacks any sort of acoustical damping materials inside (due to requirements of the local fire department and the need to place alarm equipment and sensors there). The walls on the house right side and behind the stage are covered with floor-to-ceiling curtains with a 12-inch air space between the wall and curtains.

The auditorium has a permanently installed sound system composed of high-quality components although the system configuration, in my opinion, is somewhat less than ideal. The house system consists of 4 two-way cabinets flown on either side of the stage in two tightly packed horizontal arrays. A smaller, self-powered cabinet is under-hung from each of the main clusters for center down-fill. Three double 18-inch (driver) subwoofer cabinets on each side sit on the floor of the room, under the mains in cutaways in the left and right front corners of the stage.

For system equalization, I used four parametric equalizers, two Meyer Sound CP-10s and two BSS Varicurves. One CP-10 and one Varicurve were used (in series) for each side of the system (left and right). When configured for mono operation, the Varicurve provides 12 parametric filters and the CP-10 provides an additional 10 — a total of 22 filters for each main channel.

The measurement system I used consisted of Smaart Pro version 2.1 running on a Hewlett Packard Omnibook 5700ctx notebook computer, two B&K 4011 microphones, and a custom MIDI-controlled switcher/mixer built specifically for use with Smaart Pro. The Varicurves and the MIDI switcher/mixer were connected to the computer using a MIDI adapter cable on the game port. This allowed control of both the switcher/mixer and the Varicurves directly from the Smaart Pro Real-Time module. Audio signals from the switcher/mixer were sent directly to the computer's built-in line-level inputs

Initial measurements were made (with the room empty) in the direct field of the main loudspeakers. I began by placing a microphone directly in front of each speaker array about 2/3 of the length of the room away -- being careful to stay within the critical distance. The critical distance is the point at which you start to hear the reflected sound from the back wall rather than the direct sound from the front.

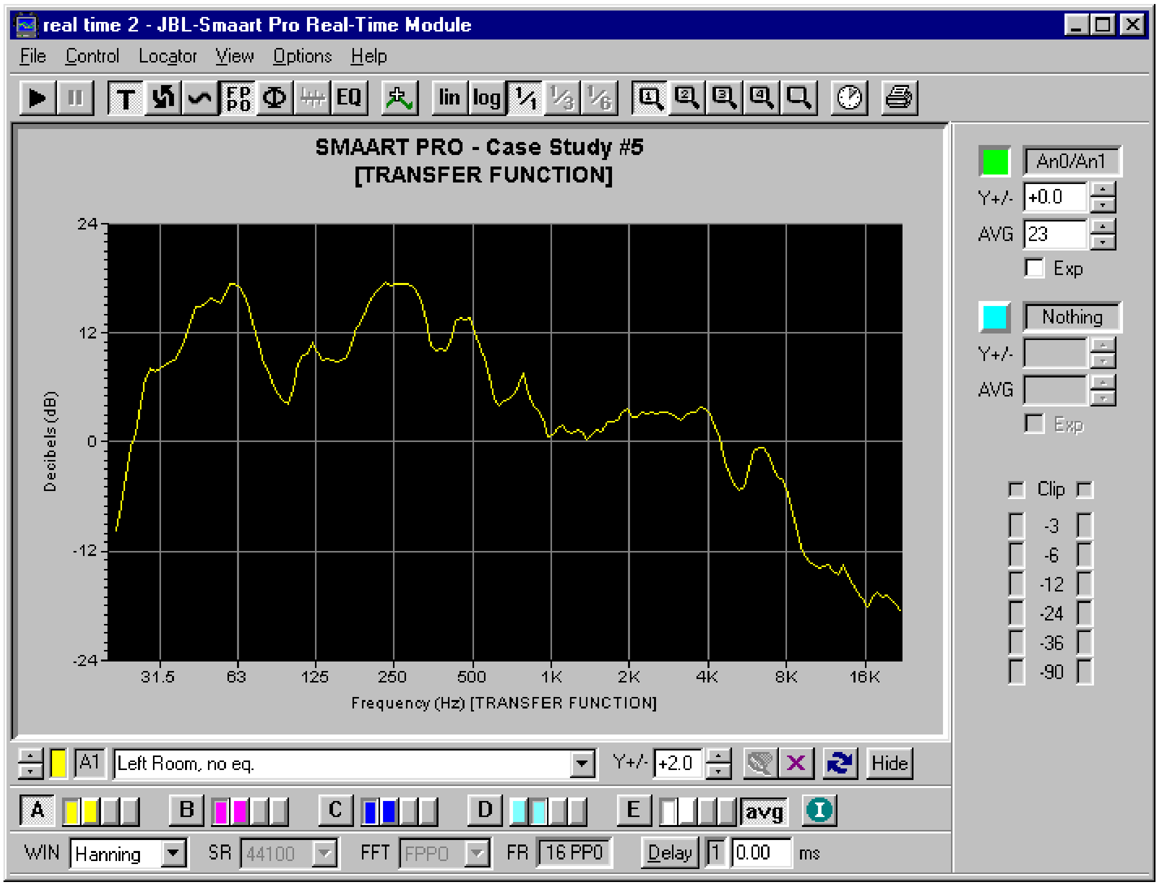


Figure 1: A Fixed Point Per Octave Resolution Transfer function measurement of the Raw system frequency response. Notice the huge low-frequency bump at 60Hz. This is a result of the stage cavities trying to "take off" (resonate acoustically). To help with this problem, I reduced the overall level of the feed to the subwoofers by 6db.

The first task was to verify that each of the individual drivers in the system were operational, wired in correct polarity, and had approximately the same efficiency. This was accomplished (using pink noise as the source signal) by turning amp channels up and down or disconnecting speaker cabinets as required. I used the delay locator in Smaart Pro with linear amplitude scaling, and the time scale expanded out to a high magnification to determine whether the initial arrival was positive- or negative-going, as well as observing the overall amplitude.

Having satisfied myself that all components were operating properly, I proceeded to adjust the overall system frequency response. After finding the delay time between loudspeaker and microphone (using the Smaart Pro delay locator), I looked at the transfer function. I used the FPPO option with some gentle smoothing.

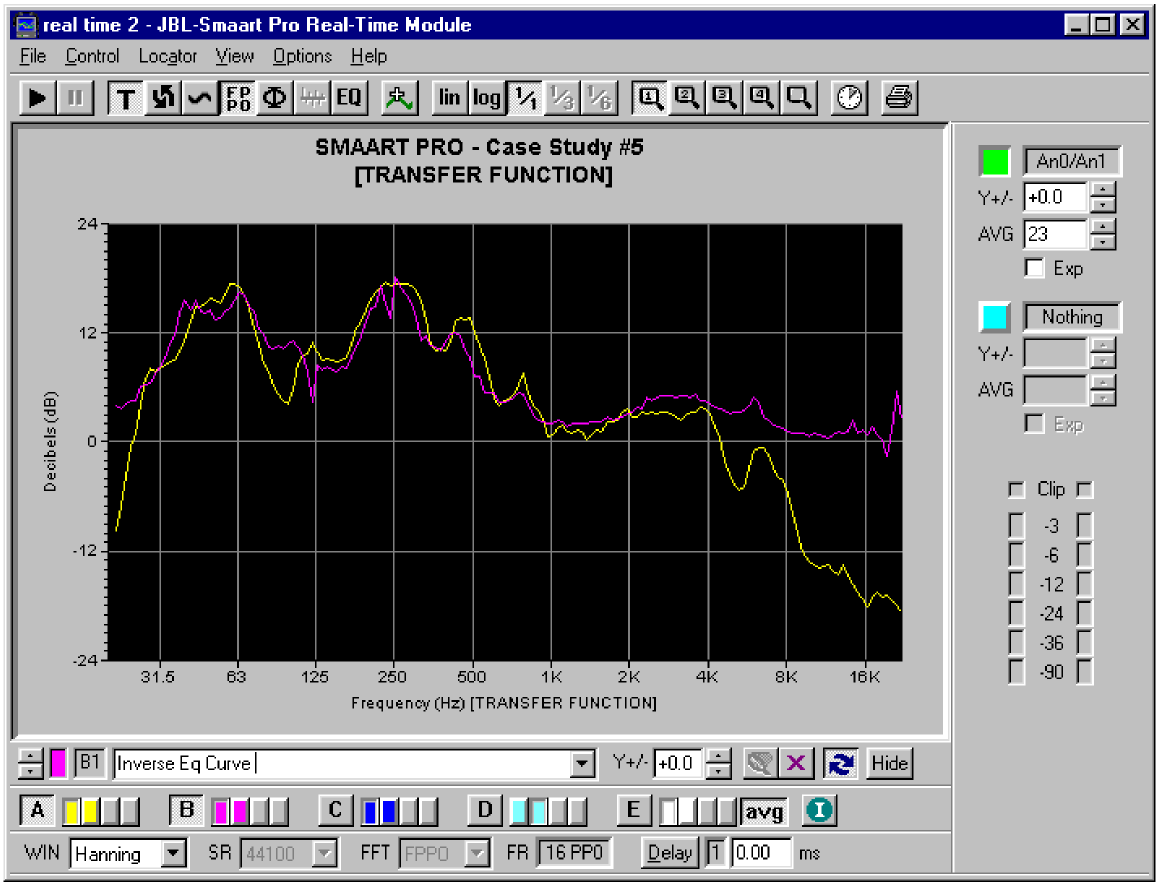


Figure 2: Inverse EQ response used to adjust system. This trace was captured sometime during the early part of the performance. Music was the noise source here which explains the relative lack of smoothness in the trace (there was not a lot of low-frequency energy present)..

Initial EQ setup was done using the original microphone location with pink noise. Afterwards, the reference microphones were moved to the FOH mix position. One microphone was placed in each of the front corners of the booth. The microphones were on boom stands and extended out toward their perspective stacks as much as possible. They ended being set approximately fifteen feet apart.

One more complication presented itself here. The mixing booth was centered in the back of the room under the balcony, whereas the stage — and consequently the PA — was slightly off-center due to the fire aisle on one side. As a result, I had to contend with a difference in the arrival times of direct sound from the left and right sides of the system.

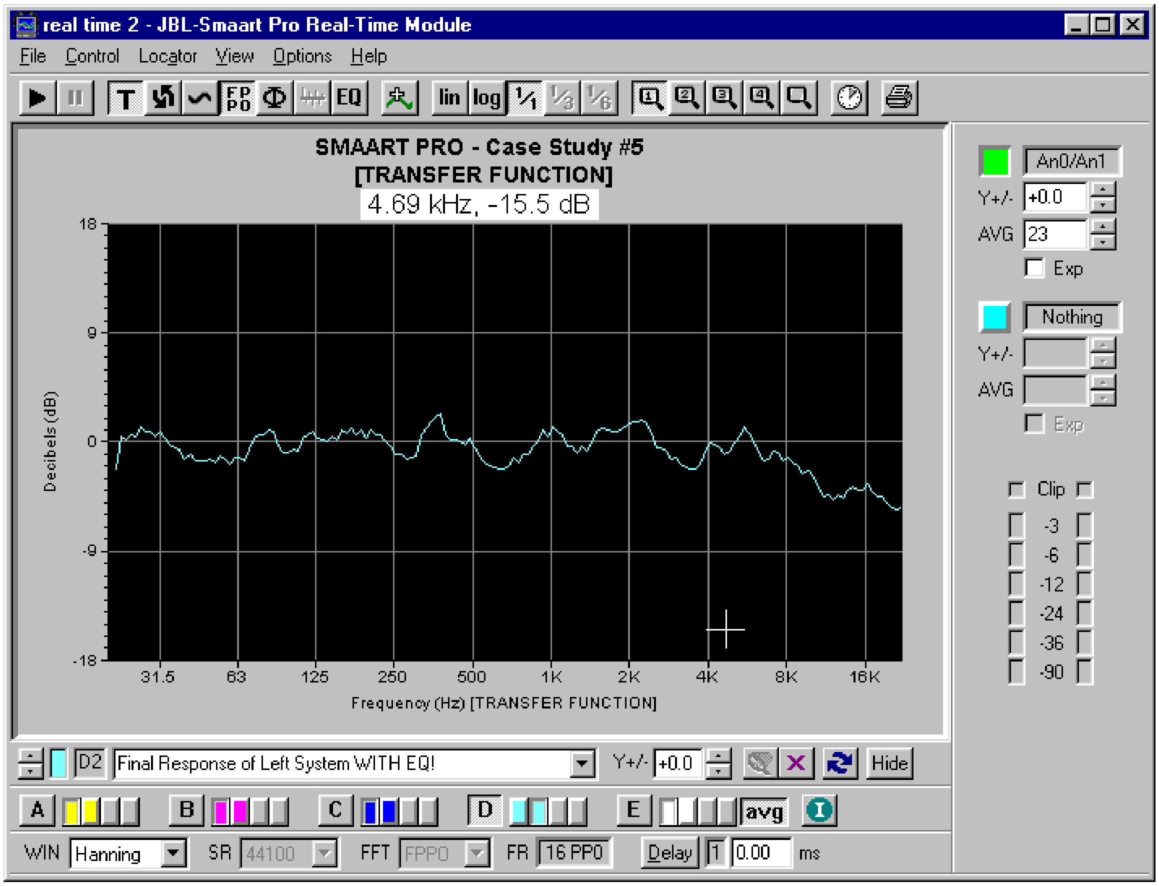


Figure 3: Frequency response after EQ — full room, microphones at the FOH mix position.

At this point the band showed up for rehearsal and sound check. All subsequent measurements were made with music as the source signal. I continued to refine the system EQ and adjust for changing conditions throughout the night using both my own subjective judgement and Smaart Pro -- with the board mix as the reference signal and my measurement microphones positioned in the mix booth, as described above.

The ability of Smaart to accurately measure the system response, and overlay the inverse equalizer response is an extremely effective way to address the problem of system equalization. I have found that being able to adjust the system during the show, by using the show as the measurement source makes for better sounding shows.

*A note from SIA:* Don Pearson is something of an Audio GURU. We realize the term "*guru*" may sound a little corny, but ask Don an audio question and chances are you will get both in-depth theory and practical, nuts-and-bolts ideas on how to address specific problems. As a founder of Ultra Sound, and now as a consultant for Pro Media, Don has been a fixture on both the San Francisco and worldwide the live sound scene for more than 25 years. As system engineer with the Grateful Dead for over 17 years, Don has faced just about every possible live sound situation imaginable. In recent years Don has been active with several companies dealing with control systems, audio on the net, and even a small company developing audio/acoustic measurement software (that would be us). Don is by far our most active "alpha" test site, suggesting changes and enhancements to SIA software products on an almost daily basis. We at SIA have learned TONS from Dr. Don. Please feel to address any questions you might have about this Case Study, Smaart Pro, Audio, Live Sound or life in Marin County to Don on the SIA Support forum.