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SIA-Smaart® Pro Case Study #4: Multi-Cluster Reinforcement System in a Large Church

by Paul Henderson

Recently I was asked to evaluate and optimize the sound reinforcement system at a large community church. The problems were very familiar: insufficient acoustic gain when using the lectern and choral microphones, combined with tonal and intelligibility issues. At the time, the multimillion-dollar facility was only a few months old, and the brand new sound system was already falling short of the staff's needs. This case study describes the basic acoustical evaluation and sound system optimization that I performed with Smaart Pro.

The main auditorium has seating capacity for over 1,200 people. It is a rectangular shaped room with a 35-foot ceiling. The sound reinforcement system is utilized for five services each week, each of which includes musical performances in styles ranging from classical to contemporary. A 100-member choir and full orchestra perform here frequently. The sound system must therefore be capable of quality musical reinforcement. The reinforcement loudspeaker system consists of a 3-way front cluster system with supplementary downfill HF horns (for the extreme front rows) and a 3-way delay cluster. Each device has a dedicated power amplifier channel and DSP output (manufacturer's parametric EQ, precision alignment delay, limiting, etc.).

Initial Evaluation and Acoustic Analysis

I began my evaluation by resetting all previously calibrated equalization devices to flat, and did a few minutes of critical listening with a familiar CD. I not only listened to the system in the audience area, but also on the stage, which immediately revealed an unusually large HF reflection arriving at the pulpit, located in the center downstage area directly under the front cluster. This was a likely cause of the gain-before-feedback problems with the lectern microphone. I decided to make some quick measurements.

I placed my reference microphone in the pulpit's location and connected it to a small mixer in the usual manner. I played pink noise through the front central cluster and adjusted it to present an acoustic level of approximately 85dBA in the audience area. For general measurements, I normally don't run the excitation signal much louder than 80-85dBA (of course, ambient noise level is the primary factor in determining this). I set up Smaart in delay locator mode, selected a 44.1kHz sample rate, a 32k point. Since I was simply looking for a strong reflection, I started with only 2 averages to speed things up. I selected the 44.1kHz SR because I was looking for a HF reflection; this ensured that the Nyquist frequency fell outside of the horn's range. The result of the impulse response measurement is shown in Figure 1, exported to the Analysis Module.

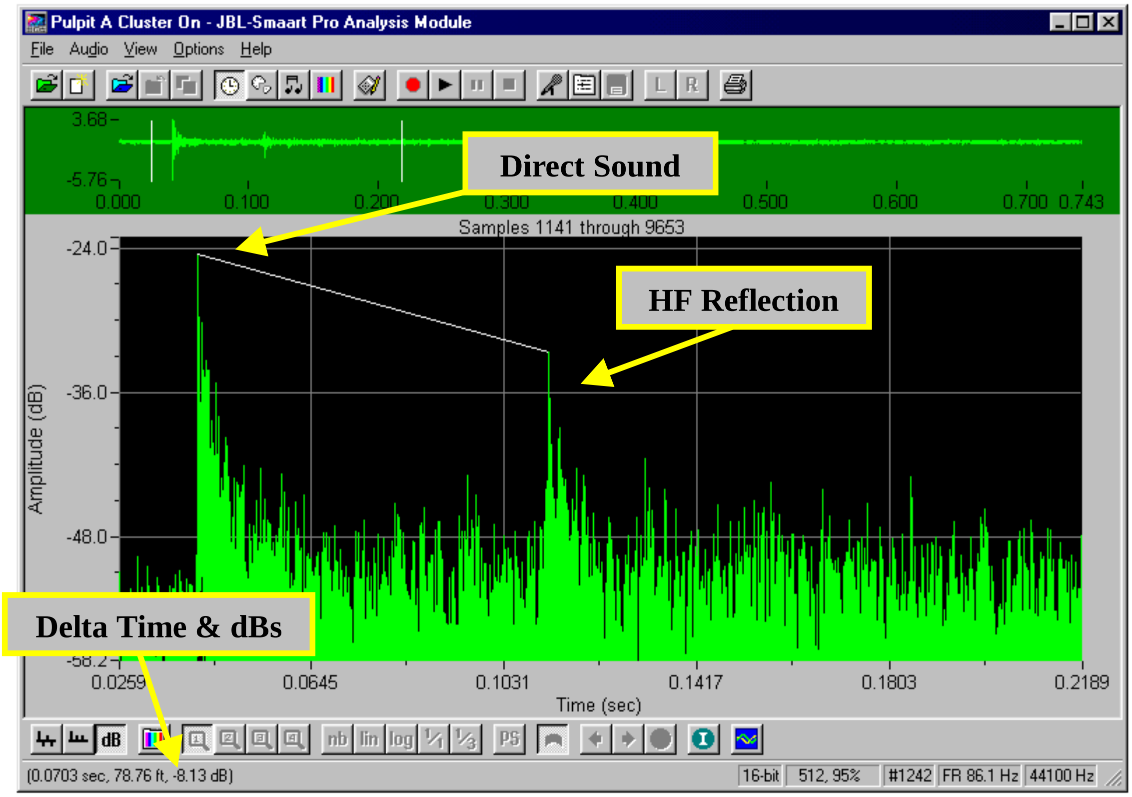

Figure 1.
Impulse response measurement in the pulpit location.

The reflection is only approx. 8dB under the direct sound, and arrives around 70ms later (equivalent to an additional 78 ft. of path length). After studying the architectural drawings of the facility and the building itself, I concluded that there was only one surface with appropriate distance and incidence angle to potentially create this reflection. The ceiling was designed with several large recesses to hide lighting fixtures. The stage-facing lip of one of these structures was in perfect alignment to cause a strong reflection from the cluster's HF horns.

This conclusion was then verified by a simple but effective method: I placed the delay locator into continuous mode and instructed a technician to move a sheet of acoustic foam around the microphone while I watched for the reflection to disappear. Note that a similar analysis can be achieved by using a directional microphone, such as a shotgun transducer, to isolate the reflection. After verifying the source of the difficulty, I recommended that the church install a small amount of acoustical absorption material, effective primarily in the 1kHz-up range, on the offending face of that structure.

System Equalization

In order to maximize the performance of the reinforcement system, I used Smaart to equalize the frequency-domain acoustic transfer functions of the various loudspeaker subsystems. The array component crossovers and relative levels were preset by the manufacturer and the original installation contractor had calibrated the precision delay between the devices.

I began the equalization process with only the full-range section of the front central cluster active (not including the downfill horns). I placed the reference microphone about halfway back in the seating area and slightly off-center. I again played pink noise into the front cluster system and used the delay locator to calculate the propagation delay time, to which it assigns the "locked cursor". I then used the Ctrl-F6 shortcut to load the delay into the Real-Time Module. To acquire the frequency-domain acoustic response of the loudspeaker system, I selected the Smoothed FPPO transfer function with 32 arithmetic averages as my measurement type. This provides maximum acoustic validity and frequency resolution across the entire audio spectrum. The unequalized transfer function of the front cluster is shown in Figure 2.

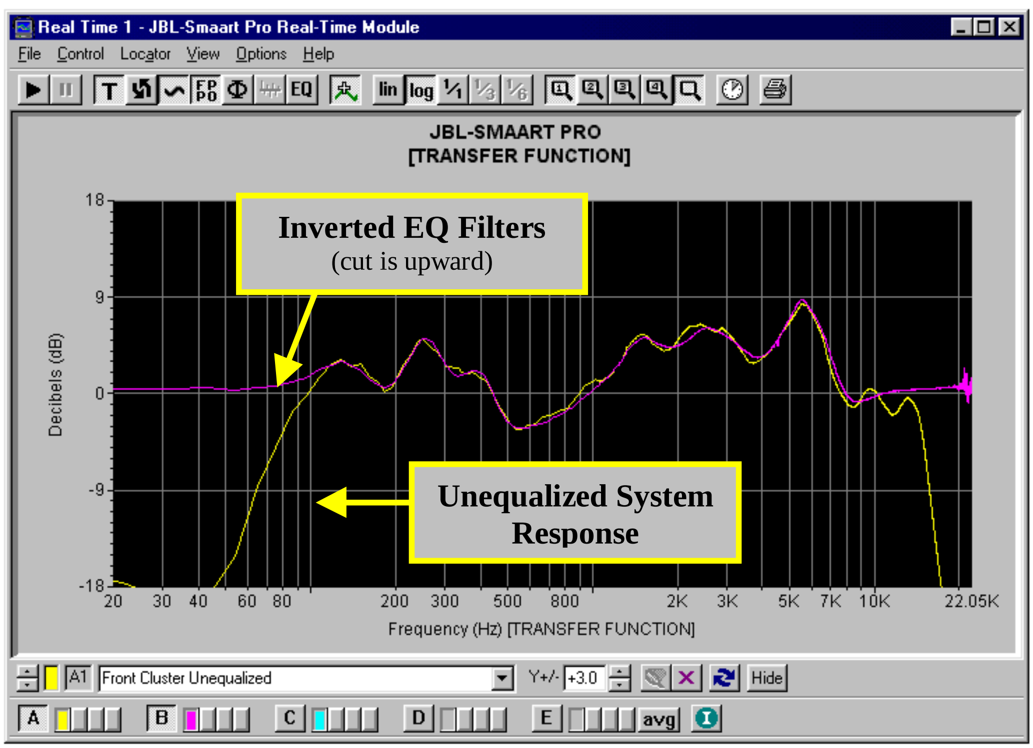

Figure 2.
Unequalized front cluster transfer function andoverlaid equalization filter response.

The equalizer associated with the main front cluster components was then calibrated to match the inverse of the unequalized response. To measure the equalizer, I patched into its secondary output and brought that signal back into the computer. This allows me to use music as my excitation source, letting me hear what changes are being made to the system in real-time. In my opinion, however, it's useful to fully calibrate the equalizer's response to best match the inverted unequalized trace before making changes based on listening. This prevents you from throwing the whole process out the door by thinking "that sounds terrible" until the final curve is in place; you'll most likely make less large changes and be much happier with the results. To tune the system's equalizers, I chose a 44.1kHz sample rate, a 4096 point FFT, and 10 arithmetic averages. The equalizers in this system were fully DSP-based, which introduces an inherent throughput delay. To compensate, I ran a quick delay locator measurement on the device. I then started the real-time module and calibrated the equalizer.

The equalization procedure described above was repeated for the downfill HF horns in the front cluster. In this system, the rear delay cluster was designed with the exact same components and alignments as the main cluster, so the equalization filter settings were simply copied into the delay cluster's equalizer. Figure 3 shows the resulting equalized transfer functions of both the full-range main cluster system and the downfill horns.

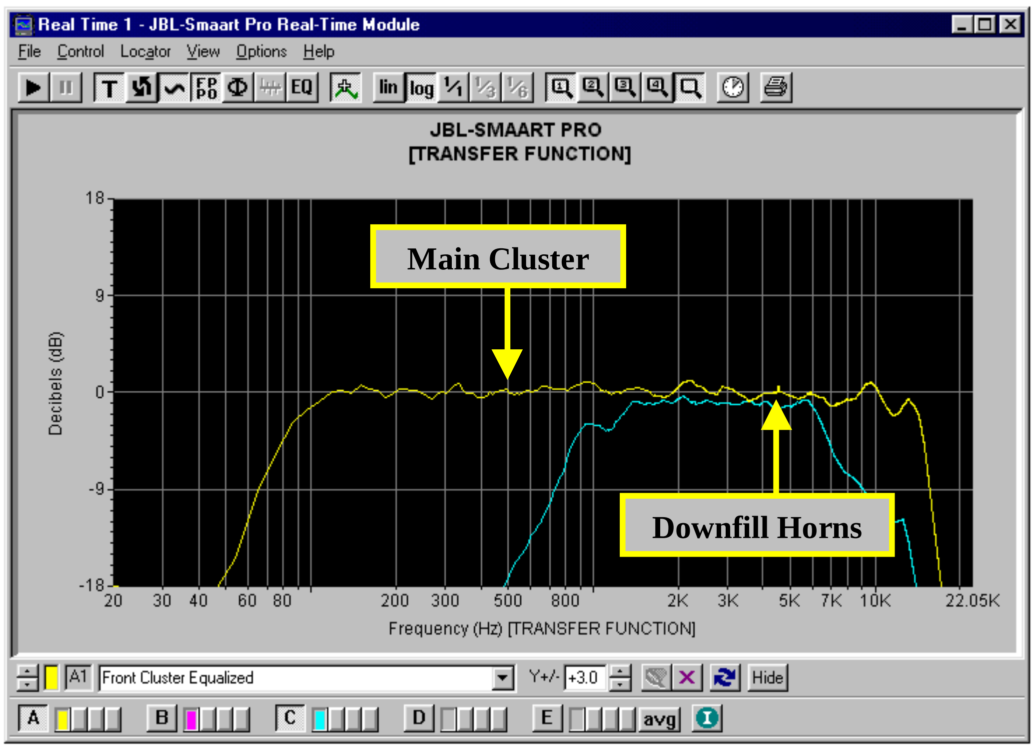

Figure 3.
Equalized transfer functions of the front main cluster and the downfill HF horn system.

As you can see, the equalization procedure greatly linearized the transfer function of the central cluster system. The potential acoustic gain-before-feedback of the choir microphones was increased over 4.8dB, and the pulpit attained an extra 1.5dB (before the acoustical modifications to the lighting recess). This highly effective system optimization provided the church with greatly increased vocal intelligibility, gain-before-feedback, and musical quality. As one church member remarked on the following Sunday, "We can actually hear the choir now!"