Case Study: Pine Grove High School Auditorium

A Tale of 'Muddiness'

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This article explores various technical audio topics, including interior acoustic design and sound system optimization. Rather than focusing on a single technical subject, the discussion is structured as a case study from one of our previous projects. While the article is intended for educational and informational purposes, it does not aim to provide an in-depth analysis of any specific audio topic. We hope readers find this article interesting, as it explores how our minds work from start to finish, from the perspective of an acoustic and electro-acoustic consultant. Please note that the article will mention several product brand names. As consultants, we are not affiliated with any organization. The product choices and software we use are merely tools to help us achieve our goals. On another separate note, several companies are working on the auditorium upgrade. This article only includes personnel or companies that work directly with HX Audio Lab. As a side note, HX Audio Lab has obtained permission to use the name and pictures of the Pine Grove Auditorium for this article.

The Initial Site Visit

In late Fall 2023, Paul Moyer of Moyer Electronics in Pottsville, PA, referred HX Audio Lab to the Pine Grove School District in Pine Grove, PA. The school district requested a major sound system upgrade for their current high school's auditorium. Understanding that the sound system interacts with the room, they gave us a green light to perform a thorough sound study of the room. The photos in Figure 1 below were taken during our first site visit.

Figure 1 - Photos from the first site visit

As audio practitioners, we recognize that the world of audio is a blend of 50% subjectivity and 50% objectivity. Therefore, the goal is to transform what currently sounds unsatisfactory (subjective) into a better-sounding system (subjective), through a process of technical work (objective). This line of thinking is illustrated in Figure 2.

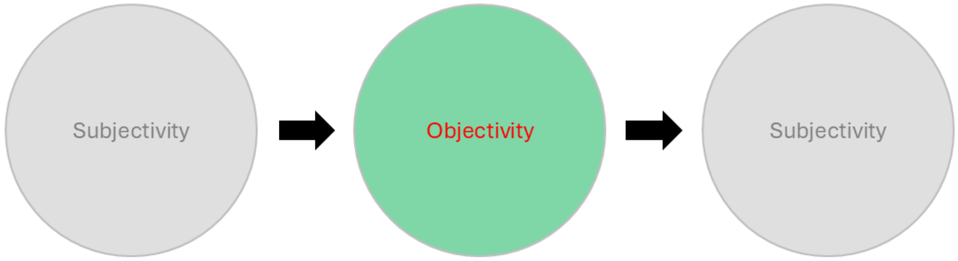

Figure 2 – The subjective-to-subjective process

Since we're discussing subjectivity, the first thing we noticed upon visiting the hall was the dark or muddy sound quality. In an objective sense, the treble/high frequencies seem to be lacking. We can see that the ceiling is shaped to redirect reflections to the audience, with reflective side walls. Carpeting isn't too much either. Where did the treble/brightness go? It turned out the ceiling was coated many years ago with what looked like a thin, porous cellulose material. Hence, the ceiling becomes absorptive above 2000Hz. Sadly, nothing can be done to the ceiling and most of the walls. We were restricted to adding panels only in limited areas. While very restrictive, this is typical of our work. We appreciate having restrictions presented at the beginning of any design work, as it clarifies our work focus.

Objective Measurement and Goal

To start our journey to the objective world, we set up our dodecahedron loudspeaker to measure the room's reverberation time. The dodecahedron loudspeaker is designed to distribute sound evenly across all frequencies. For the audio enthusiast, you may notice that our dodecahedron loudspeaker, as shown in Figure 1, is compact. How can we measure the low frequency? We also used the QSC K8.2 loudspeaker for low-frequency measurement, and the data is spliced around 200 Hz. Twelve microphone locations on one side of the auditorium are selected (Figure 3), and Figure 4 shows the reverberation time (T60) measurement for each microphone location and the average (red curve).

As a side technical note, we use AFMG EASE 4.4 software, a tool for calculating and designing room improvements and loudspeaker layout. For sound system optimization, we utilize AFMG Systune and EASERA for data collection, along with HX Audio Lab's Filter Hose to generate FIR filters. Several software screenshots are included below, showcasing the programs mentioned in this paragraph.

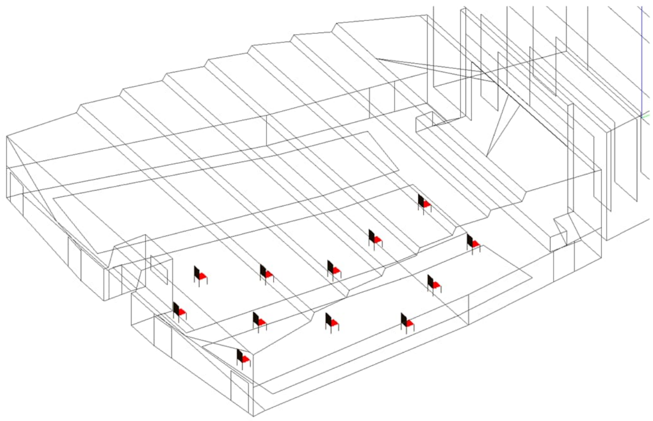

Figure 3 - Twelve microphone locations are shown as virtual chairs

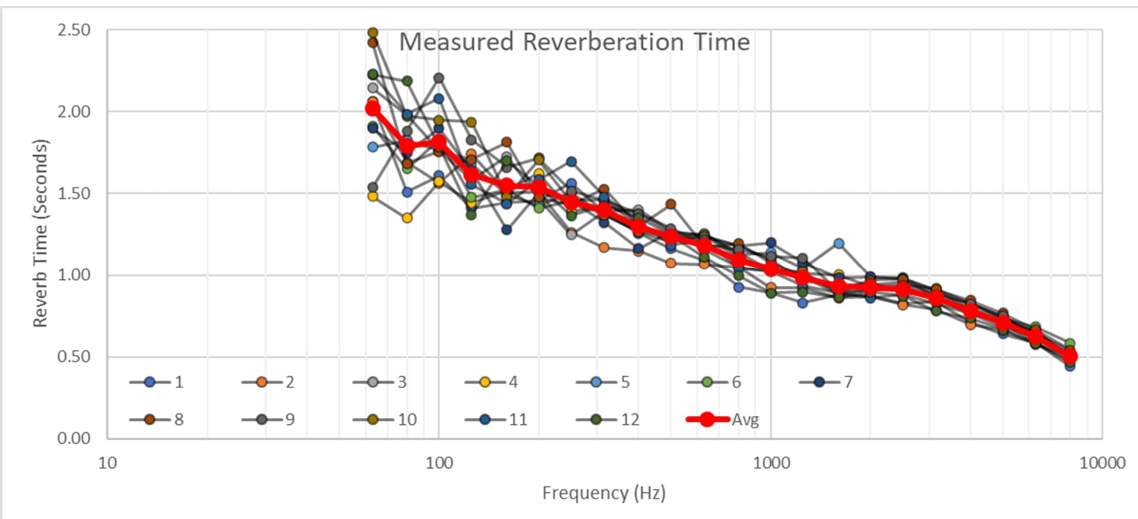

Figure 4 - Measured reverberation time

Notice how the red curve (averaged measured reverberation time) rises sharply toward lower frequencies. A ratio of 1.5, relative to 1000 Hz, is reached at around 200 Hz, and it worsens at lower frequencies. Preferably, the reverberation time below 1000 Hz should not increase by more than 1.4 times. As the auditorium is used for modern music, a less sloped reverberation curve is preferred. The acoustic improvements aim to reduce the reverberation time at frequencies below 1000 Hz.

As a side technical note, several studies have been conducted regarding the perceived warmth in a venue, such as observing the reverberation time perspective (bass ratio or T60 slope) or the G strength at low frequency. In this article, we focus solely on the rising reverberation time below 500 Hz, which contributes to the venue's dark and muddy sound quality.

Room Acoustic Modification

We proposed several changes, including upgrading the lower side walls, adding panels to the corners of the top side walls, installing diffusers, and placing several bass traps. Although material costs were not a concern, the renovation of several walls is prohibited. At the end, they only implemented corner side-wall panels and bass traps under the stage extension. Not ideal, but it can yield noticeable results. Figure 5 compares the average reverberation time values of the same twelve microphone locations after the corner side-wall panels are installed. We specified RPG Modsorbor panels and positioned them in a way that benefits the room while reducing low-frequency accumulation from the loudspeakers.

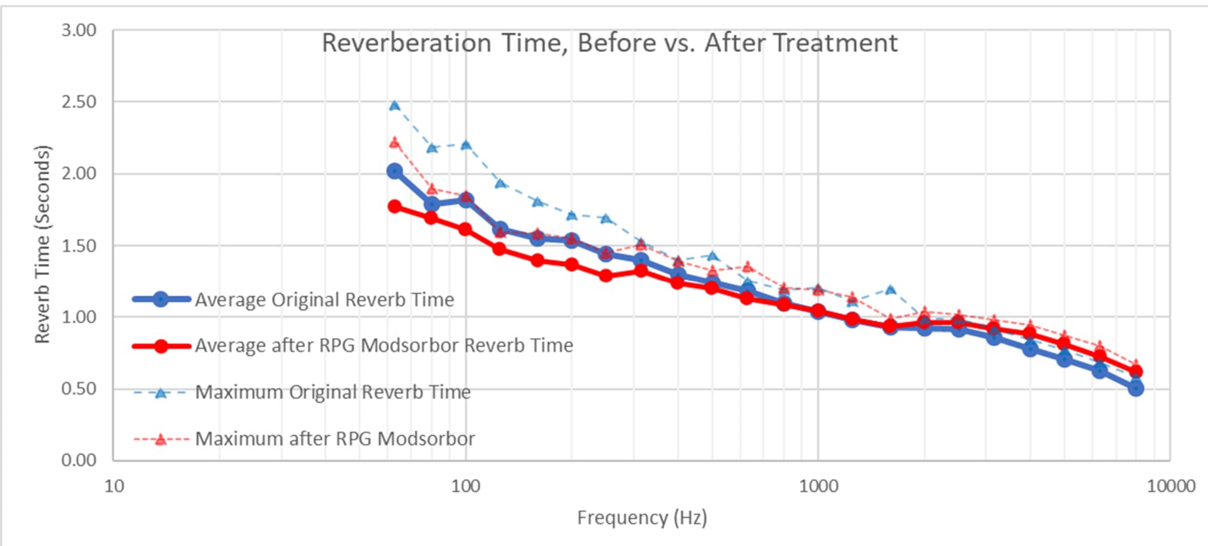

Figure 5 - Reverberation time measurements before and after the corner side-wall panel installation

The solid red curve shows the new average curve of the auditorium's reverberation time. Several notes are as follows.

- The high frequency average (solid red curve above 2 kHz) seems higher as there is less absorptive material (drapes, etc.) in the stage area during our second visit.
- The blue maximum original reverberation time (dashed blue curve) has a small spike at 1600 Hz. This error in the measurement is due to a strong reflection at a specific location.
- This measurement does not include the bass traps under the stage extension. We did not have the chance to measure the final room as the bass traps were added after the sound system was commissioned. With the under-the-stage bass traps, we expect the reverberation time at 125 Hz and below to decrease slightly, thereby tightening the low-frequency reproduction of the sound system.

The treatment reduces the reverberation time at frequencies below 1000 Hz as expected. Figure 6 shows the locations of the RPG Modsorbor panels in dotted rectangles.

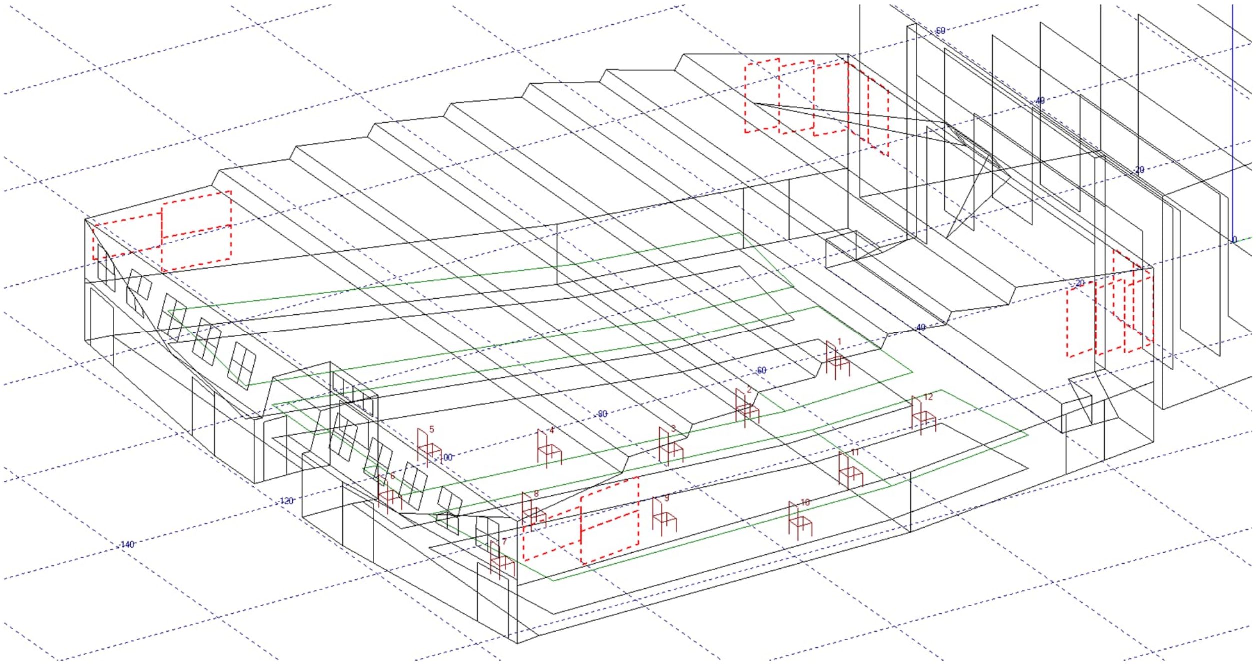

Figure 6 - Locations of the RPG Modsorbor panels, marked with dotted red rectangles

Although Figure 5 does not indicate a significant reduction in low-frequency reverberation time (i.e., the curve slope is still considered quite steep), everyone agreed that the difference was noticeable, even during simple conversations in the auditorium. The room sounds cleaner and less muddled while still retaining some warmth in the overall sound quality. The front panels also did a great job in reducing the low-mid accumulation of the loudspeakers installed near that corner. We were impressed by the effectiveness of these panels. The overall sound quality in the venue became warm and natural (instead of warm and muddy, initially), requiring minimal low-frequency electrical correction in the loudspeaker system. More about loudspeakers will be discussed in the subsequent chapters.

We have made every effort to enhance the room within the limits of renovation restrictions. Since the primary goal is to upgrade the sound system, this step has been essential in creating a supportive acoustical environment for it.

The Loudspeaker Layout

Two independent systems were requested: one designed for speech and the other for music playback. Biamp (previously Community Pro Loudspeakers) loudspeakers were selected, and the chosen products are as follows.

- Speech reinforcement: One LVH-909 beam-forming venue horn.
- Music reinforcement system: IV6 Line arrays in a stereo configuration.

Figure 7 - The look of the auditorium during our second visit

Figure 7 shows the LVH-909 located in the center, and the (audience) left IV6 array. Readers can notice the RPG Modsorbor panels surrounding the IV6 arrays, as well as several decorative red panels throughout the side wall, for aesthetic purposes.

Internal Loudspeaker Set Up

The IV6 array consists of passive arrayable loudspeakers, allowing users to adjust the overall output and high-frequency (HF) output by adjusting specific jumpers behind each loudspeaker.

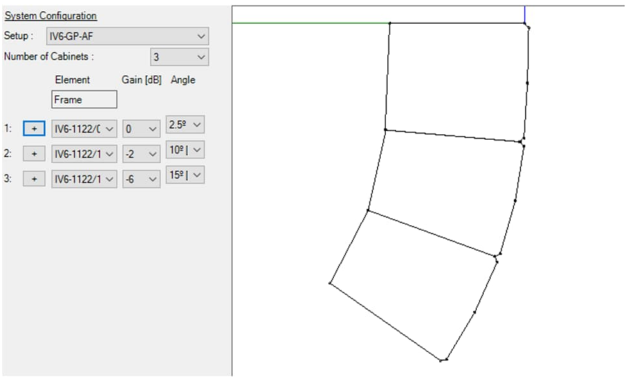

Figure 8 - IV6 set up

Each IV6 array consists of one 5-degree (where 'degree' refers to the vertical coverage) box for the top cabinet and two 15-degree cabinets for the middle (-2 dB gain adjusted) and bottom (-6 dB gain adjusted). Figure 8 shows the side profile of each array.

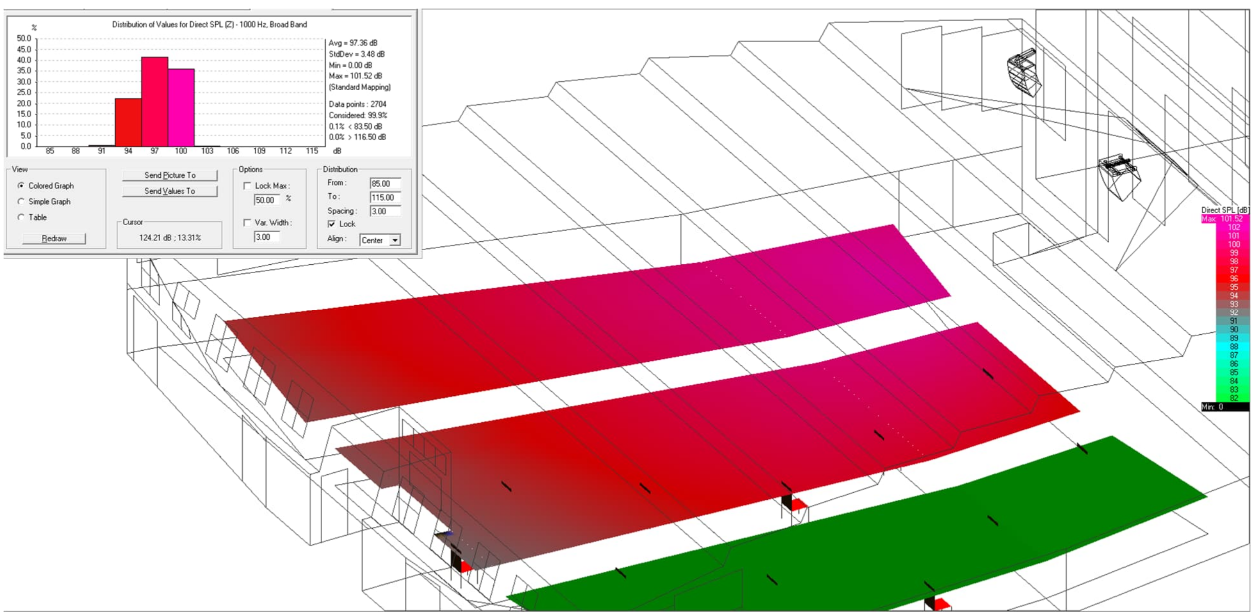

Figure 9 - Total direct sound coverage (100 - 10,000 Hz)

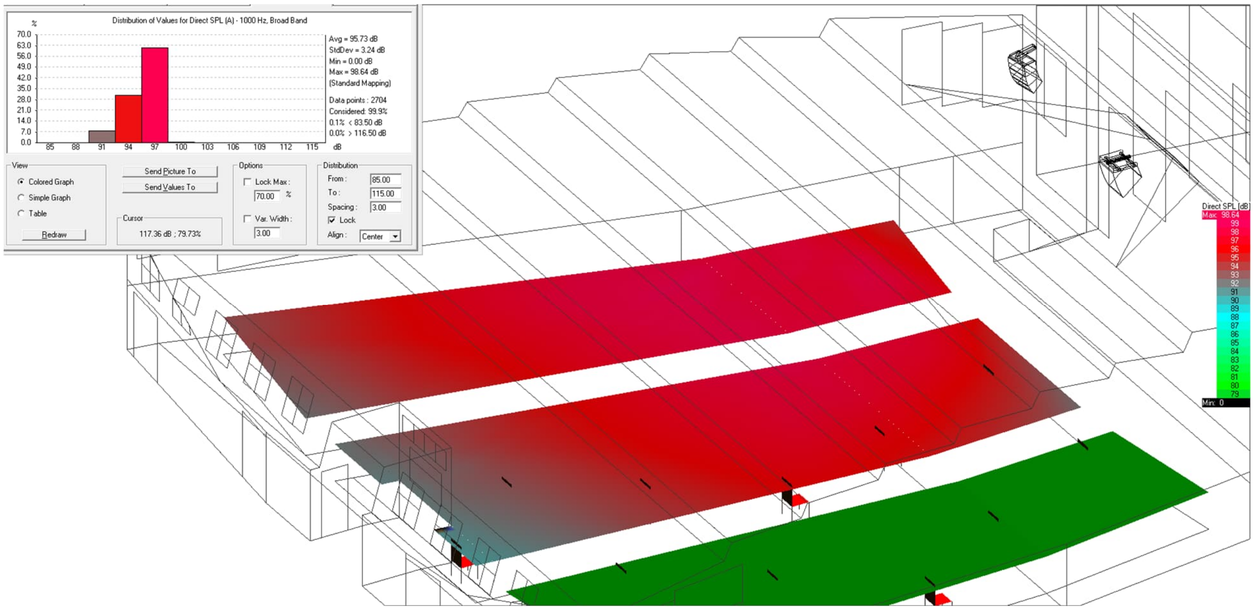

Figure 10 - Total direct sound coverage (100 - 10,000 Hz), A-weighted

Figure 10 shows the (audience) left array total direct sound coverage of 100 – 10,000 Hz, A-weighted. The left-top distribution graph shows excellent coverage for speech, with approximately ±3 dB of variation on 90% of the left and middle audience areas.

Figure 9 shows the total direct sound coverage of the (audience) left IV6 array for frequencies ranging from 100 to 10,000 Hz. The left-top distribution graph shows good coverage for music. This direct coverage can be improved by adding more loudspeakers to the array to focus the sound more tightly and distribute it more evenly, but we don’t think the additional cost is worth pursuing an ideal solution. As is, the loudspeaker sound coverage from the IV6 is considered uniform.

As electro-acoustic consultants, we also studied uniformity of the direct sound coverage on a frequency band basis. These graphs are not displayed to keep the length of the article short. It is important to note that what’s seen in the prediction software is not what you hear. Direct sound coverage shows how the loudspeaker spreads its sound. We, however, hear the direct sound along with all the room reflections.

For the speech system, LVH-909 with a custom configuration is selected. The loudspeaker was customized to provide inputs to each component. With the help of the Biamp Engineering team, they developed specialized FIR filters to shape the sound beam tailored specifically to this auditorium’s audience area, minimizing spills to the back wall and ceiling. On another note, since this configuration was not a standard loudspeaker at that time, we did not receive complete support from the other department. Therefore, for other electro-acoustic consultants interested in working on custom projects, obtaining support from the manufacturer may be challenging, and worse, receiving a snide comment. A slight hiccup is usual, and luckily, it didn’t cost us much time. HX Audio Lab is very grateful for all the support given by Biamp’s engineering team. However, to avoid any future unpleasantness, we won’t show the final prediction mapping or more details about this custom project.

Since the LVH-909 is mounted in the middle, close to the audience’s first row, the front-side corner audiences are not well covered. In this regard, we must further adjust the signal distribution so that the LVH-909 receives assistance from the IV6 arrays. The speech system is configured so that when the LVH-909 is activated, the bottom cabinet of each IV6 array is also activated, providing coverage for the corner front-side audience. The tuning process becomes tricky because precise timing and gain adjustments must be made to achieve seamless coverage with minimal comb filtering and an imperceptible image shift between different loudspeaker systems. The details of the alignment process are not covered in this article.

Due to the direct connection to each LVH-909 driver (a total of nine amplifier channels to run this loudspeaker), we also set up double limiters to protect the drivers from over-excursion and excessive heat. This is one essential aspect in sound system optimization that is often overlooked. With modern amplifier technology, setting up limiters is not hard, and most manufacturers provide a preset for their DSP hardware with the correct limiter setup. For a custom project, we must carefully study the specifications of each component. This article does not cover the details of the limiter setup, but we want to emphasize its importance.

A Peek at Equalization Development Process

Indeed, this article would not be complete without examining the fractions of the optimization process, especially the development of the equalization curve. We occasionally receive questions regarding this process, and this article shares a portion of the relevant information. For those interested, HX Audio Lab also offers an online course, "FIR Filter Creation using Filter Hose." The second part of the course covered a range of practical and technical aspects of EQ development, not confined solely to the FIR world or the Filter Hose software.

Figure 11 - Amplifiers and DSPs

Figure 11 shows the Biamp ALC's amplifiers used to power the IV6 arrays and the LVH-909. The amplifier has built-in advanced digital signal processing (DSP) for most system optimization needs. All filters, delay, gain adjustment, and limiters are implemented in the amplifier. First, let's look at the IV6 arrays. The measurement below is done using the (audience) right IV6 array and at the twelve microphone positions (Figure 3). Figure 12 shows the individual microphone response at each position without any post-processing, and Figure 13 shows the responses with windowing applied. The impulse response windowing is set up to include the first 50 ms of energy and is referred to as the direct field response. We will reserve the details of the impulse response windowing for future articles, as they involve another lengthy discussion. The steady-state response, on the other hand, is the response obtained from the full measurement without windowing or editing. The steady-state response represents what our ear hears, as it includes the loudspeaker's direct sound and all room reflections.

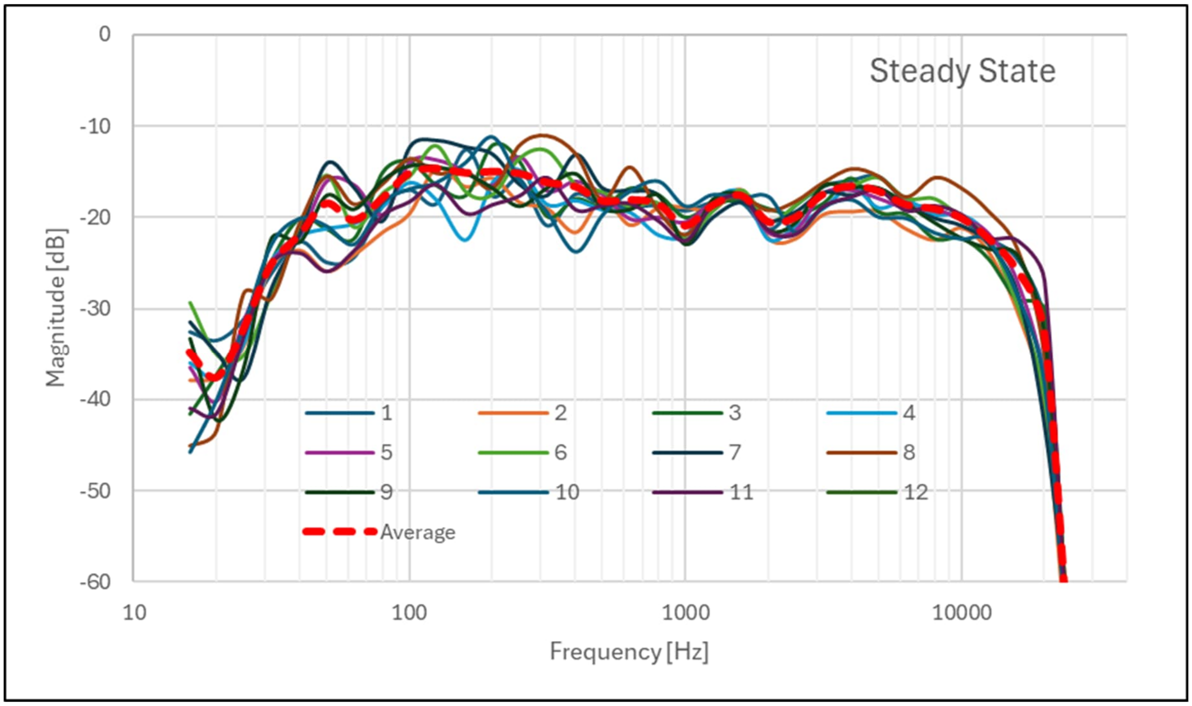

Figure 12 - IV6 right array at 12 mic positions, steady state response

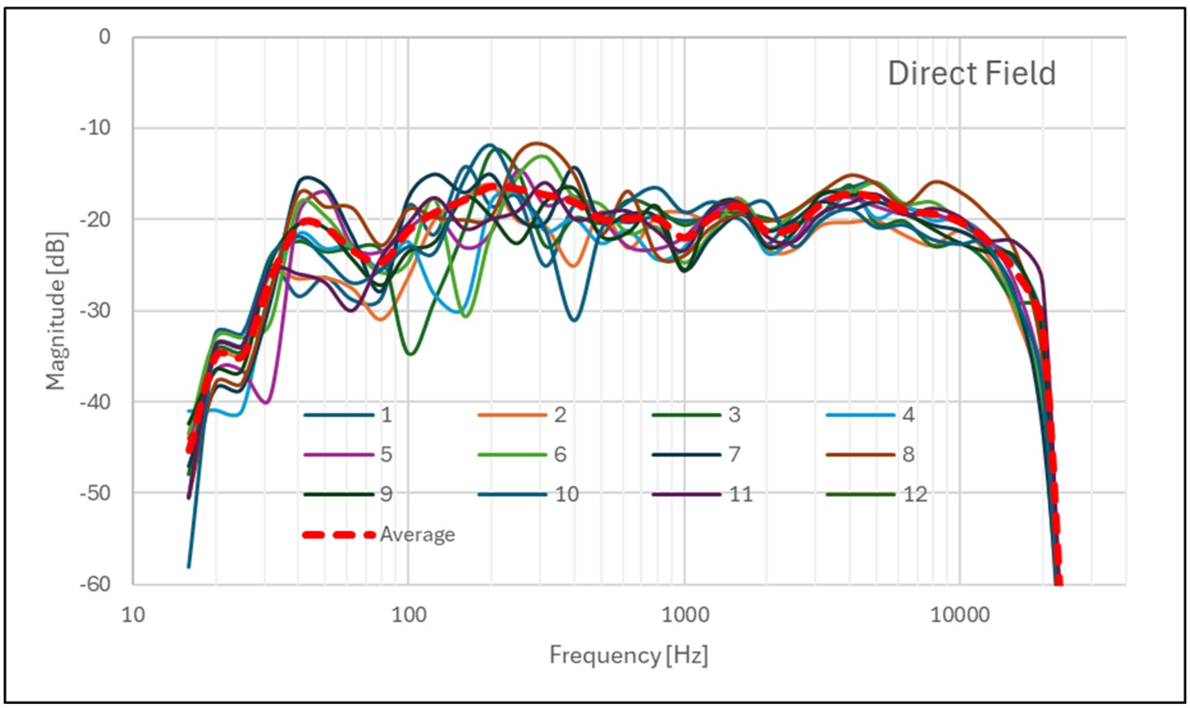

Figure 13 - IV6 right array at 12 mic position, includes the first 50ms of energy after the direct sound arrival

Figure 13 shows slightly more erratic behavior at frequencies below 400 Hz. The modal response and strong reflections (the IV6 arrays are installed near the corner of three boundaries) are the reason for this behavior. Zooming in on the average curves (red dashed lines), Figures 12 and 13 do not show a drastic difference below 1000 Hz. This is consistent with our subjective perception that the room sounds less muddy after the installation of the RPG panels. In other words, the room does not significantly alter the frequency response of the loudspeakers at low frequencies.

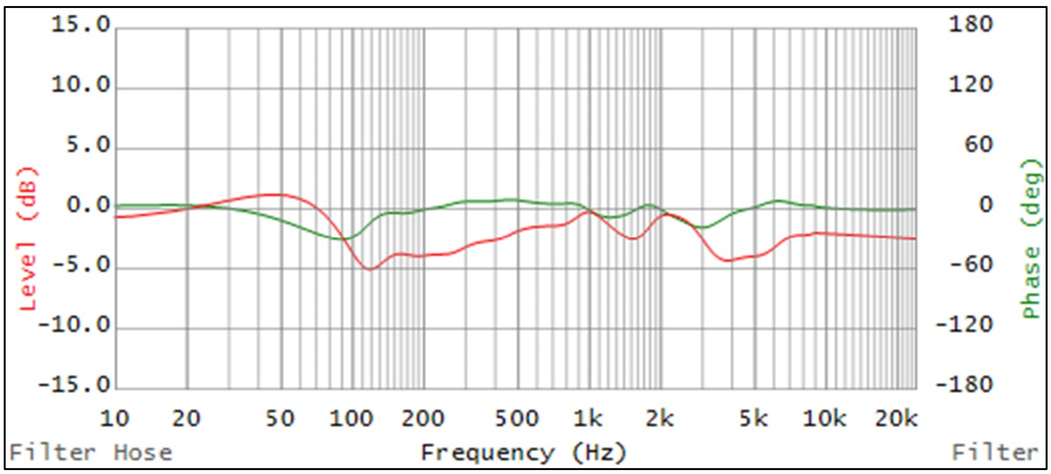

Figure 14 - First-stage FIR filter response for the IV6

Figure 14 shows the first-stage minimum-phase FIR filter we developed for the IV6 array. As previously mentioned, the loudspeaker does not need a significant boost or cut. The EQ correction in Figure 14 is approximately ±2.5 dB or 5 dB max cut above 100 Hz. From past experiences, the low-frequency accumulation (below 500 Hz) of some auditoriums can easily exceed +12 dB! In some rare cases, we remembered applying an 18 dB cut at frequency regions around 100-400 Hz for another auditorium project.

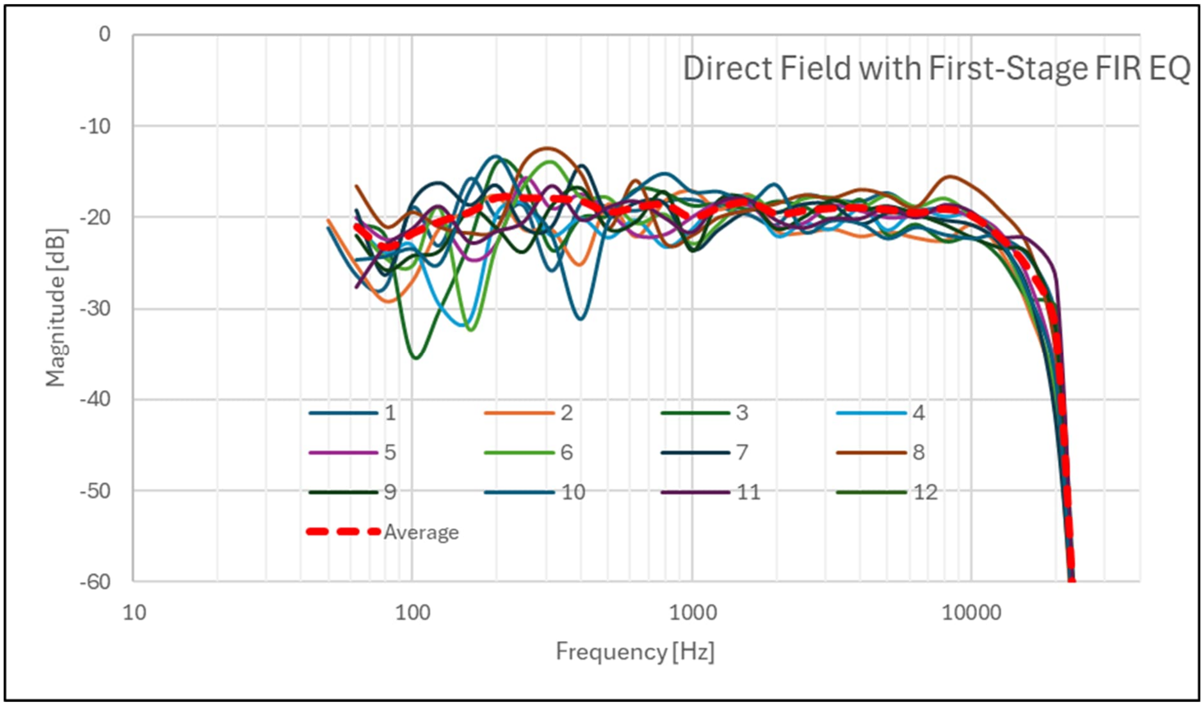

Figure 15 -IV6 Direct Field with the First-Stage FIR EQ

Figure 15 shows the direct field responses with the EQ (Figure 14) applied. The first-stage EQ was developed based on direct field responses, targeting an average response with less than ±3 dB variation above 500 Hz. With this EQ, the IV6 arrays begin to sound natural in the room, or in other words, blend well with the room. However, it is not all smooth yet, as there is still some polishing to be done. Before working on the second-stage filter, let’s see how the LVH performs in this room. To shorten the discussion, we will skip the initial measurement (without EQ).

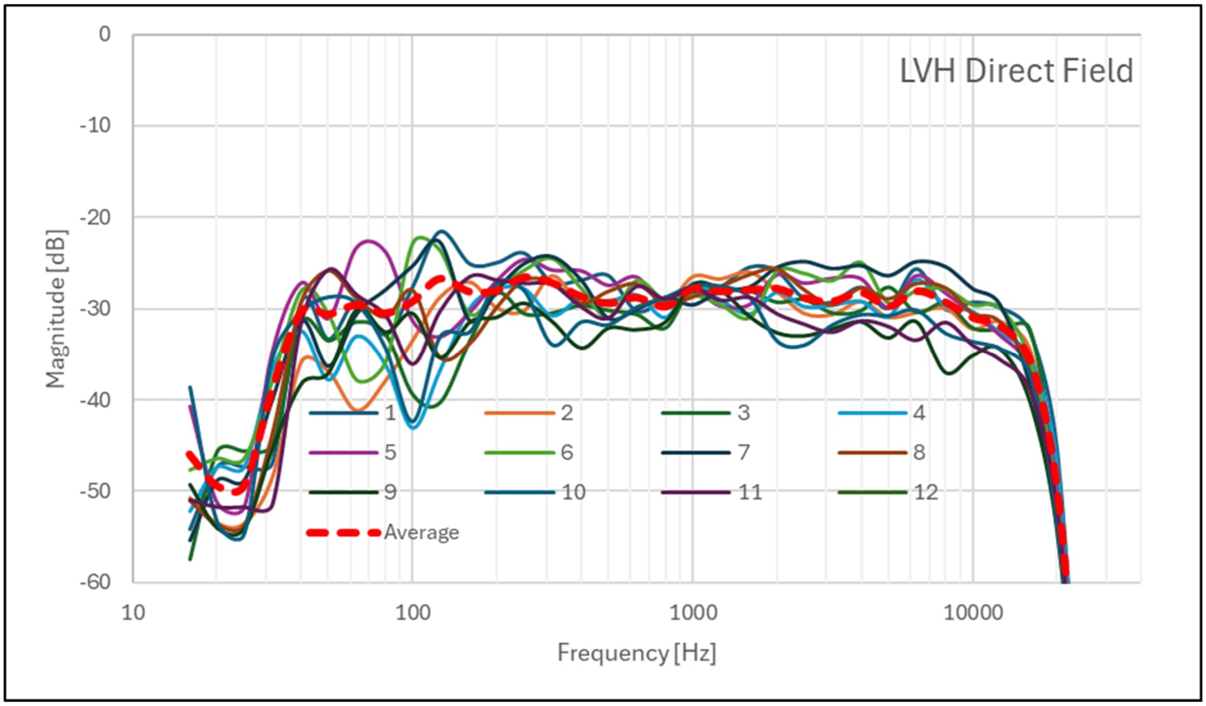

Figure 16 - LVH direct field responses with the First-Stage FIR EQ

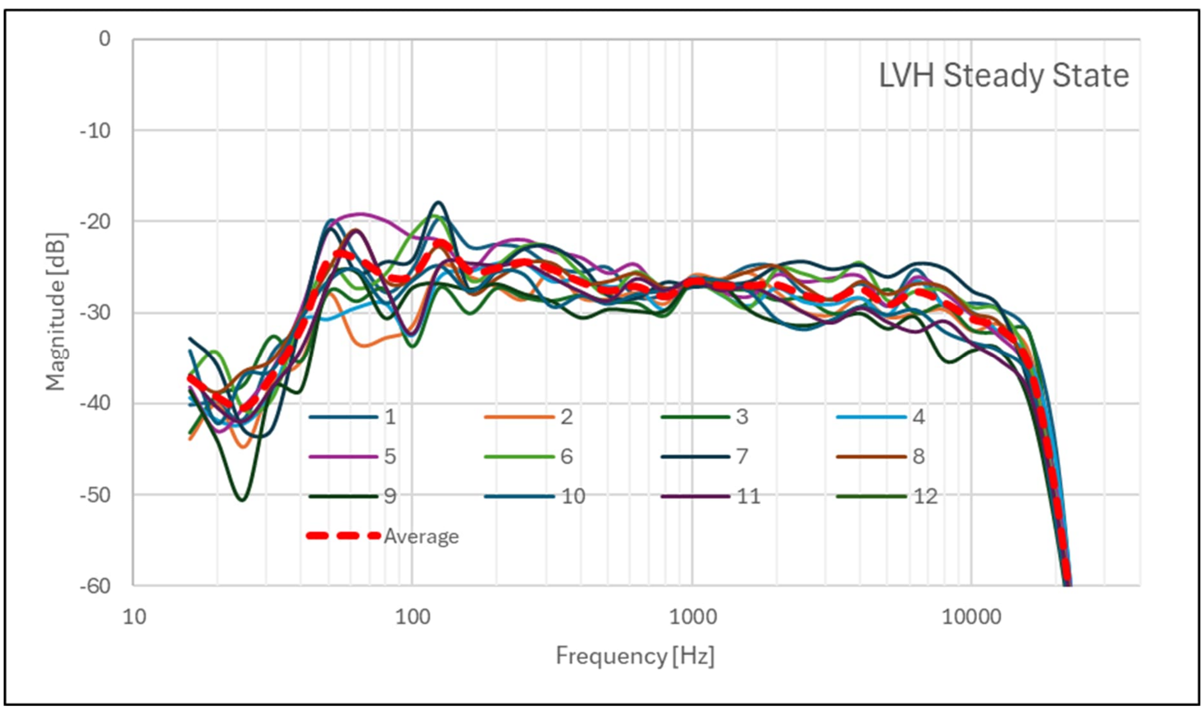

Figure 17 - LVH steady state responses with the First Stage FIR EQ

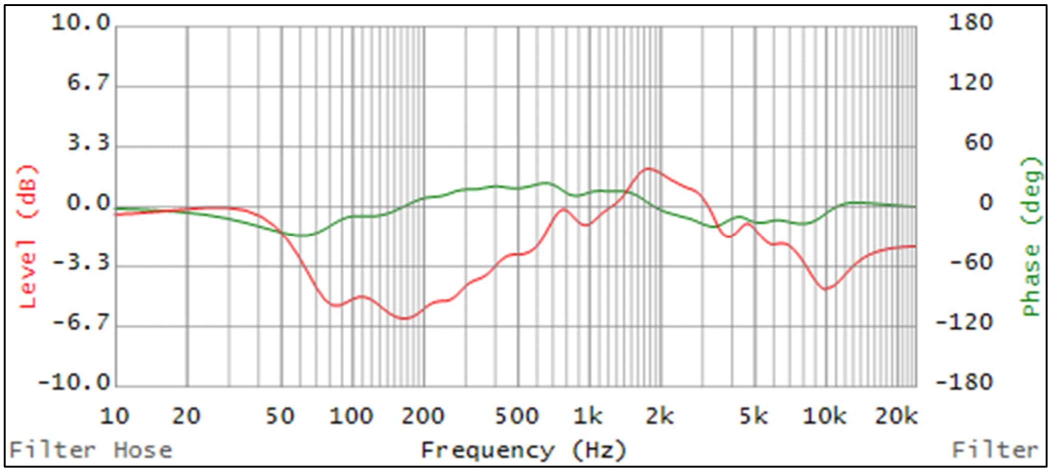

Figure 18 - First-stage FIR filter for the LVH

Figures 16 and 17 show the measured EQ'd direct field responses at the 12 exact microphone locations. The first-stage FIR filter we developed is shown in Figure 18. At a glance, it appears to have more cuts at low frequencies. This is caused by more powerful amplifier channels assigned to the woofers, as this loudspeaker is amplified and processed per driver. Comparing figures 15 and 16, the first-stage FIR filter for both systems created an average direct field response with a variation of ±3 dB. When switching between the systems, it was already hard to distinguish which system was being used (referring to the speech LVH system vs. the IV6 music system).

Although both systems already sound quite natural in the room, there are still minor colorations in the playback. Subjectively, we recalled there were one or two frequencies that could be further cleaned up. This is expected, as the first-stage EQ was created based on direct sound field responses, which remove many room reflections during post-processing. Another study was conducted to find these frequencies objectively. Please note that there are many alternative methods for developing equalization filters. This method is just one of the many. Figure 19 shows a comparison between the average of the direct field and the steady-state response, with the dashed line representing the IV6 array and the solid line representing the LVH system. The curves in this graph may differ slightly from those in the previous figures due to variations in the normalization during post-processing. Figure 20 shows the difference curves.

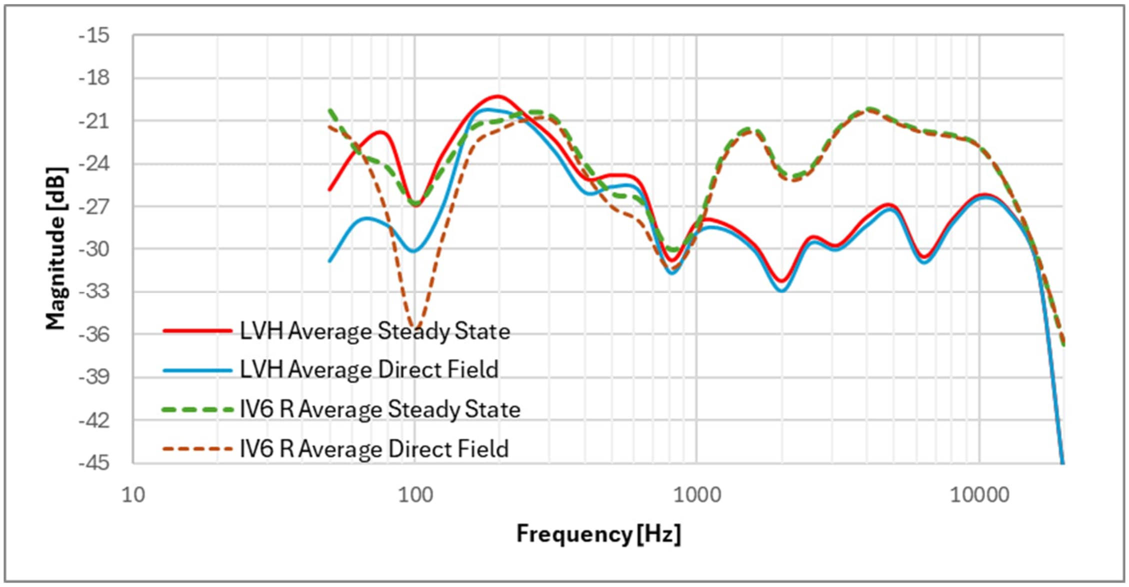

Figure 19 - Comparing the average of the direct field response and the steady state response

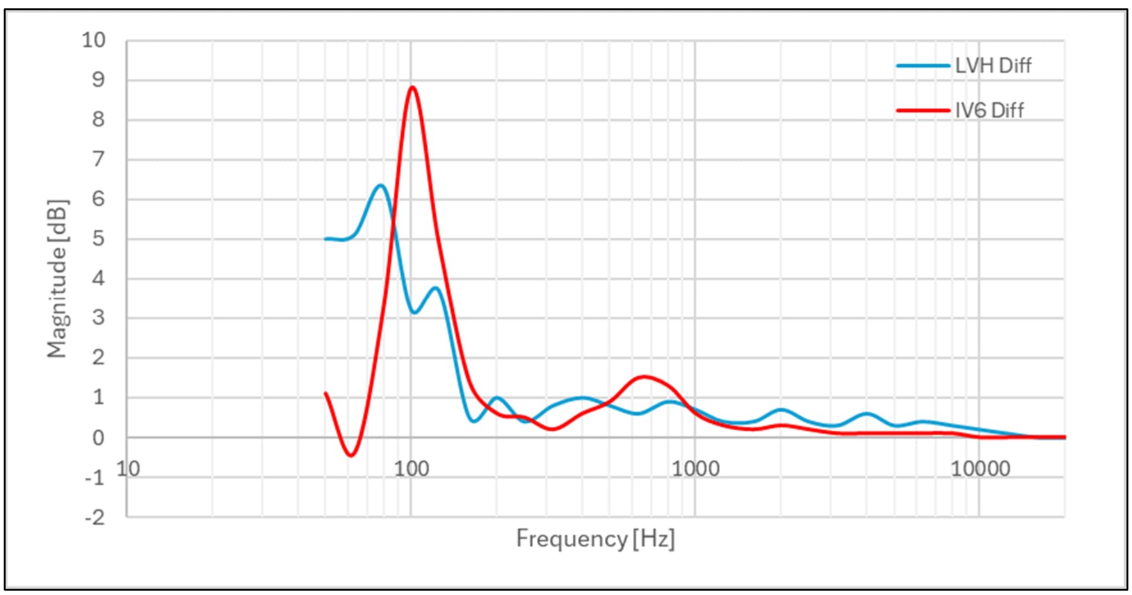

Figure 20 - The difference between the average steady state and the direct field

Due to the windowing used, Figure 20 has validity at frequencies above 75 Hz (also, the subwoofer will be more dominant at 80 Hz and below). In our findings, a bump in the difference curve of 1 dB or more can give coloration to the loudspeaker's playback. Whether it is considered a negative coloration or not is up to the optimizer's judgment. Please note that HX Audio Lab has developed this process independently, drawing on years of experience in system optimization.

In our case, we found out that the bump at approximately 650 Hz in the IV6 system (note that the red curve in Figure 20 shows a bump at 650 Hz) was the culprit of the negative coloration we heard. While in the LVH system, we attempted to add several EQ filters at frequencies of approximately 800 Hz, 400 Hz, and 120 Hz, as indicated by the blue curve in Figure 20, which shows bumps at these frequencies. The final second-stage EQ was implemented as IIR filters, and the exact frequency points were slightly moved based on subjective listening of known recordings. At the end, only one filter is added to each system. Although it appears to be a minor tweak, it has significantly improved the system's sound quality to the point that the two sound qualities are almost indistinguishable. With the future addition of the bass traps (after the system optimization is done), we believe the overall system playback quality is even better now.

To keep the article short, we do not delve into the alignment of the loudspeakers, i.e., subwoofer-to-full range, and the fill loudspeaker (lower IV6) or the details of the limiter setup. We hope this article provides some insight into the journey of EQ curve development, one of the major optimization processes. Lastly, we would like to share several key points to conclude the article. These points were all taken from previous discussions or frequently asked questions we received throughout the years.

- Understand the room and loudspeaker interaction.
  The room and the loudspeaker interact. To create a natural sound (note that the word 'natural' is a subjective term), adjustments can be made to both sides. Their interactions can be thought of analogically as a 'marriage,' where adjustments can be seen as a form of 'compromise.'
- Use mathematical function tools with an open mind.
  A common function available in modern tools is calculating the power average of the responses. While some argue that the spatially averaged response is not what is heard, we used it as a guideline along with the individual responses. We always observe both the spatial averages and individual responses. Impulse response windowing, a form of impulse response post-processing, is another common tool. Some practitioners adhere rigidly to the use of specific window types or lengths, and there is no absolute right or wrong in this approach. It's best to correlate what you hear and what you see on the screen. The system optimization process is highly fluid, so two venues that appear similar may yield a significantly different measurement result. Use mathematical function tools to create metrics that are relatable to your work.
- Understand that system optimization can't resolve all problems.
  System tuning or optimization refers to maximizing a system's potential. It is heavily tied to the design concept behind the system's installation. The result of the optimization process will be heavily restricted if the sound system design is flawed, such as incorrect loudspeaker placements, choices, layout, or orientations.

- Be flexible on the tools you use.
  Sound system optimization is a work that is full of surprises, especially when faced with unexpected technical issues in situ. Other frequently asked questions we received were about how and why we chose our tools. A simple answer is that we're accustomed to them, and it gives us the ability to improvise in many surprising situations quickly.

We would like to thank all personnel who collaborated on this fantastic room and sound system upgrade. Pine Grove now boasts a state-of-the-art sound system in their auditorium to support many years of student performance. HX Audio Lab is glad to be a part of their journey.

Figure 21 - From left to right: Paul Moyer of Moyer Electronics, Hadi Sumoro of the HX Audio Lab, and Mike Marr of Biamp

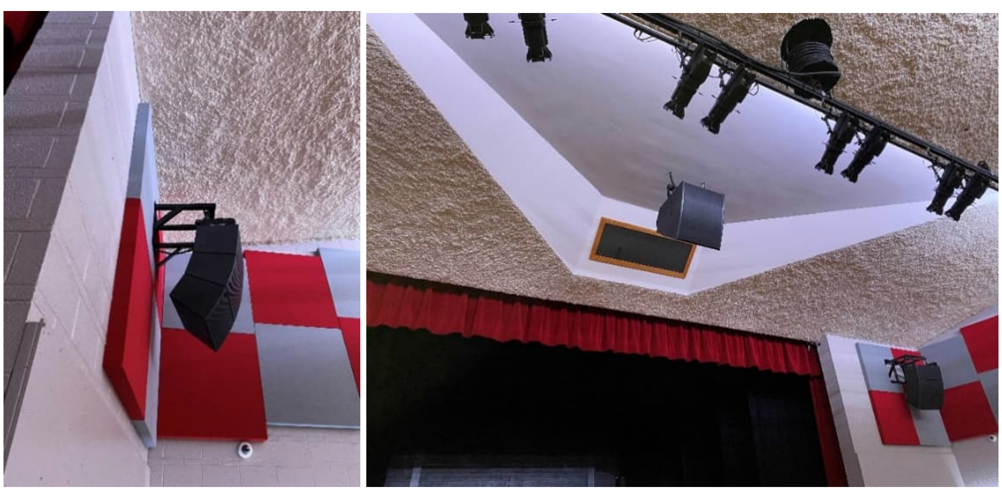

Figure 22 - More installation photos of the loudspeakers and front corner panels