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Speaker Innovation Happening Every Day

On December 10, 1915, Danish engineer Peter L. Jensen and Edwin S. Pridham presented a groundbreaking invention to the public in Golden Gate Park in San Francisco, CA. It was the first practical application of a moving-coil loudspeaker, using a horn to amplify the sound. They successfully took their invention to market, building speakers for public address systems and for radios under the Magnavox brand.

Much earlier, British physicist, inventor, and radio pioneer Oliver Joseph Lodge described the concept of the dynamic loudspeaker and was even granted a patent for a dynamic, moving-coil loudspeaker concept. Ten years after Jensen and Pridham successfully demonstrated a moving-coil loudspeaker, in 1925, Edward W. Kellogg and Chester W. Rice applied for the first of many loudspeaker-related patents. The "Sound reproducing apparatus" was issued in 1926, and the "Loud-Speaker" patent was issued in 1929.

Much later, Edgar Villchur (working with Henry Kloss in Acoustic Research) received a patent for the much improved "acoustic suspension" loudspeaker. When Kloss left to form KLH (with Malcolm Low and J. Anton Hofmann) in 1957, and started to sell loudspeakers under license from Villchur, only then did other manufacturers notice and adopt the technology. Altec Lansing even turned down the opportunity to buy the original patent from Villchur.

In our September 2023 issue, *audioXpress* published a thorough analysis of the C2S Concentric Coplanar Stabilizer (C2S) loudspeaker technology, developed by Dinaburg Technology. The article from Roger Shively (Shively Acoustics International) used today's state-of-the-art numerical simulation and measurements to illustrate the full benefits of this patented integral coaxial speaker cone with a stabilizing ring radiator. Besides providing powerful subwoofers in thin and compact boxes, Mikhail Dinaburg's invention enables lower distortion, extended frequency range, higher efficiency, and improved speech intelligibility in a multitude of speaker and even headphone applications. Today, as in 1957, 1925, and 1915, bringing innovations to market that are actually recognized by consumers, requires a huge engineering effort.

Coincidently, while working on Roger Shively's article, I came across a reference to an interesting earlier approach to the dual concentric cone—the ALTEC Lansing Biflex speakers. And I was curious as to why this approach seemed to have been forgotten. In a note to Roger Shively, he confirmed that he was extremely familiar with the concept, stating, "Neal House and I messed around with the 'biflex' cone ideas in our early days (1986 to the early 1990s) at Harman. We called it a cone decoupler, in an attempt to control the breakup better. After I wrote Harman's first FEA/BEM program (good ol' NASTRAN and FORTAN and a VAX machine the size of small bedroom...), I think I convinced myself it was futile. It didn't stop us from trying to get more into a simple package. There was a co-motional tweeter (an independently driven tweeter cone mounted to the apex of the cone). And, a version of the cone decoupler with a dual voice coil—one coil for mids and highs and the other for the bass."

It sounded extremely familiar. Even with all the resources of one of the largest audio companies in the world, some concepts are simply not feasible with the technology of the time.

When people say that "speaker technology has not changed much in more than 100 years," it's an obvious demonstration of how many times we fail to recognize how much happens every year in the speaker world. In the editorial I wrote for *The Audio Voice* in January 2022, "CES 2022 Audio Highlights," unknowingly I documented one of those key moments. As I wrote, at that show Swedish company Exeger—the company behind the revolutionary Powerfoyle solar cell material that enables self-charging consumer electronic devices—promoted a joint demo with an innovative Dutch startup, called Mayht (later acquired by Sonos for its Heartmotion speaker driver technology). They demonstrated a proof-of-concept, compact speaker using Exeger's Powerfoyle material applied to the cabinet, allowing the speaker to keep playing without any power source. A great example to showcase the benefits for consumer electronics brands.

Fast forward to 2023, and Urbanista, a Swedish lifestyle audio brand, announced the world's first self-charging wireless speaker with an integrated Powerfoyle membrane. The Urbanista Malibu is a portable, waterproof design that self charges whenever exposed to indoor or outdoor light. It was officially launched at IFA 2023 in Berlin and is available to order now.

I wonder what the early pioneers would say about a light-powered loudspeaker?

J. Martins
Editor-in-Chief

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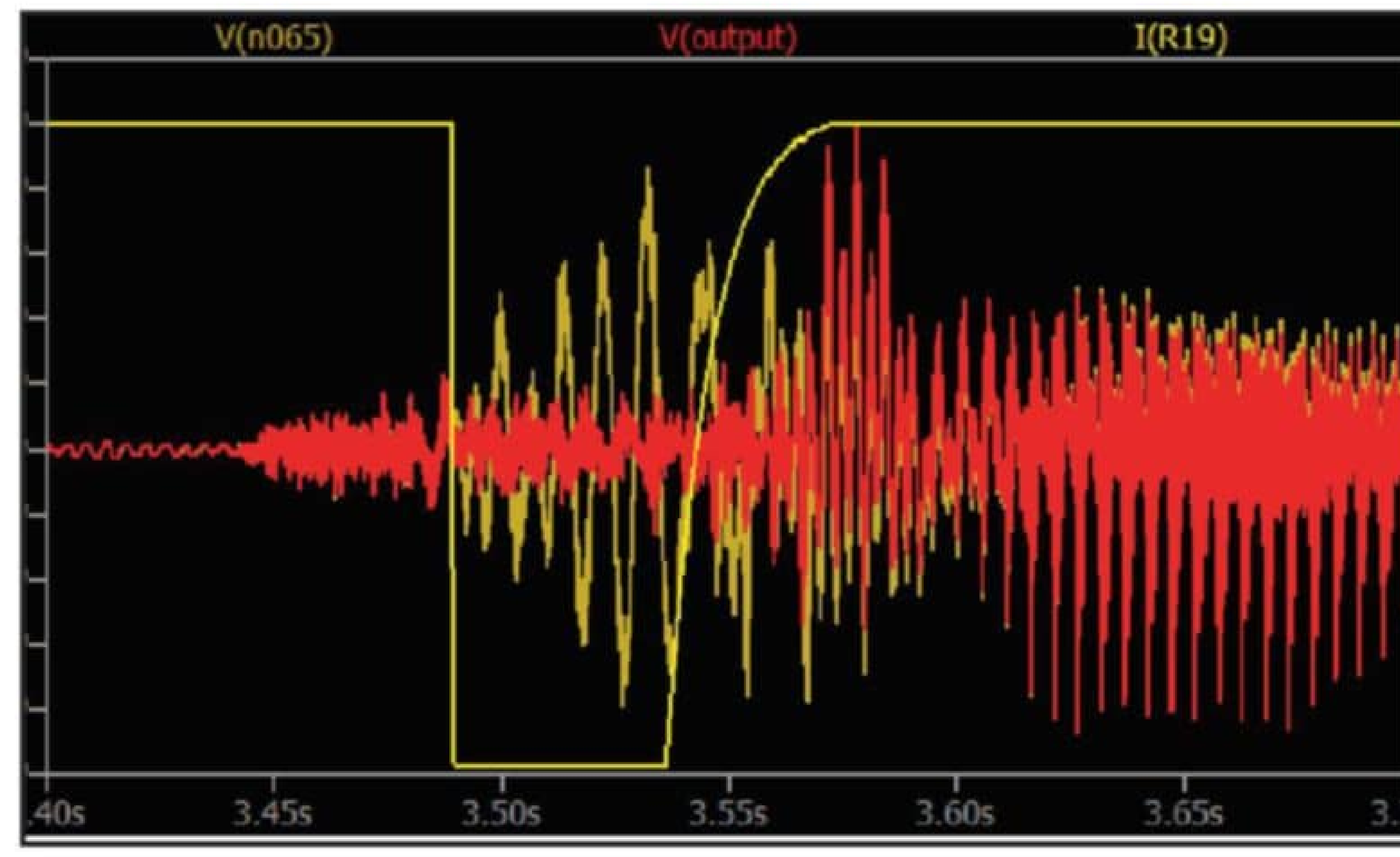

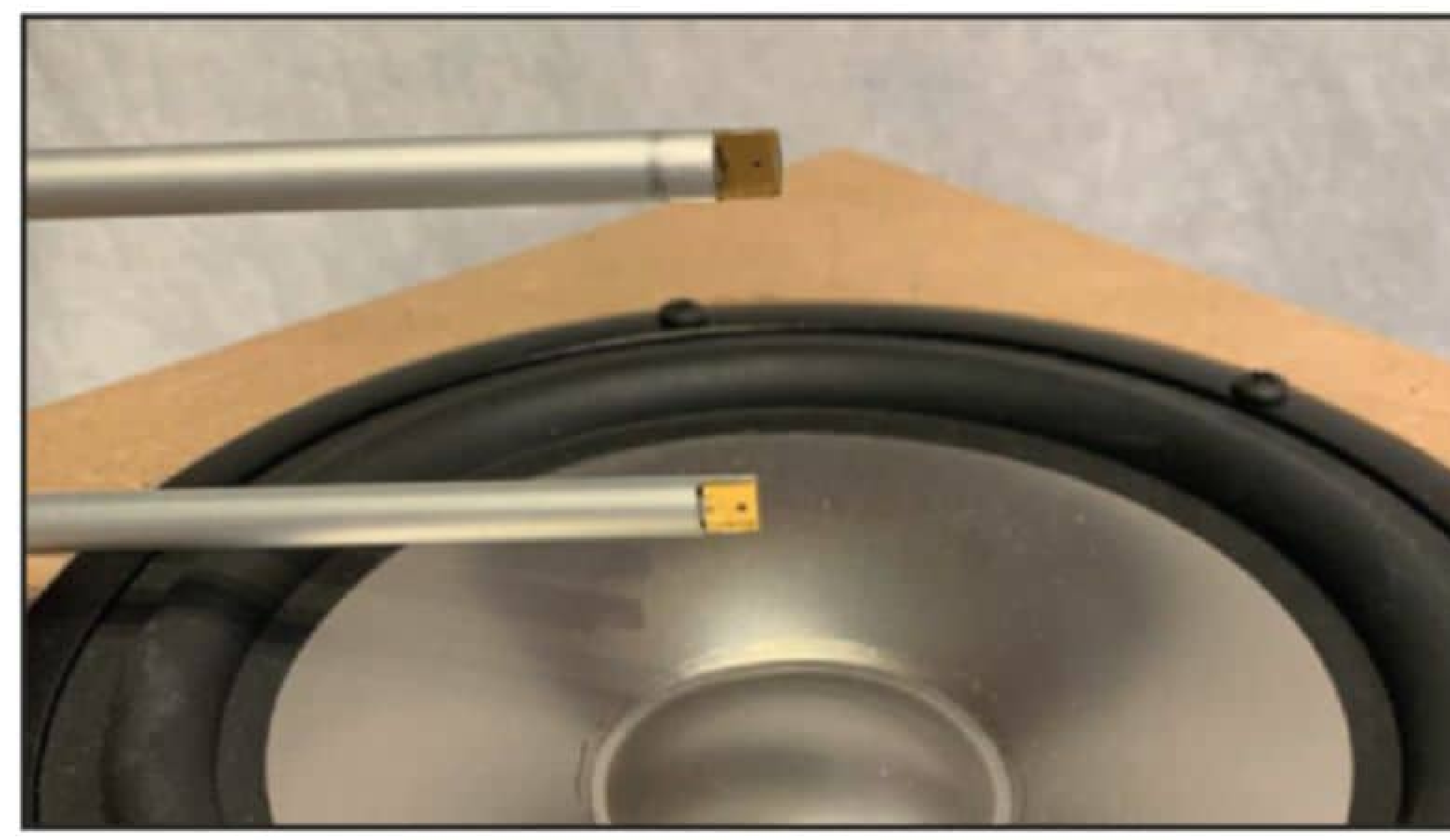

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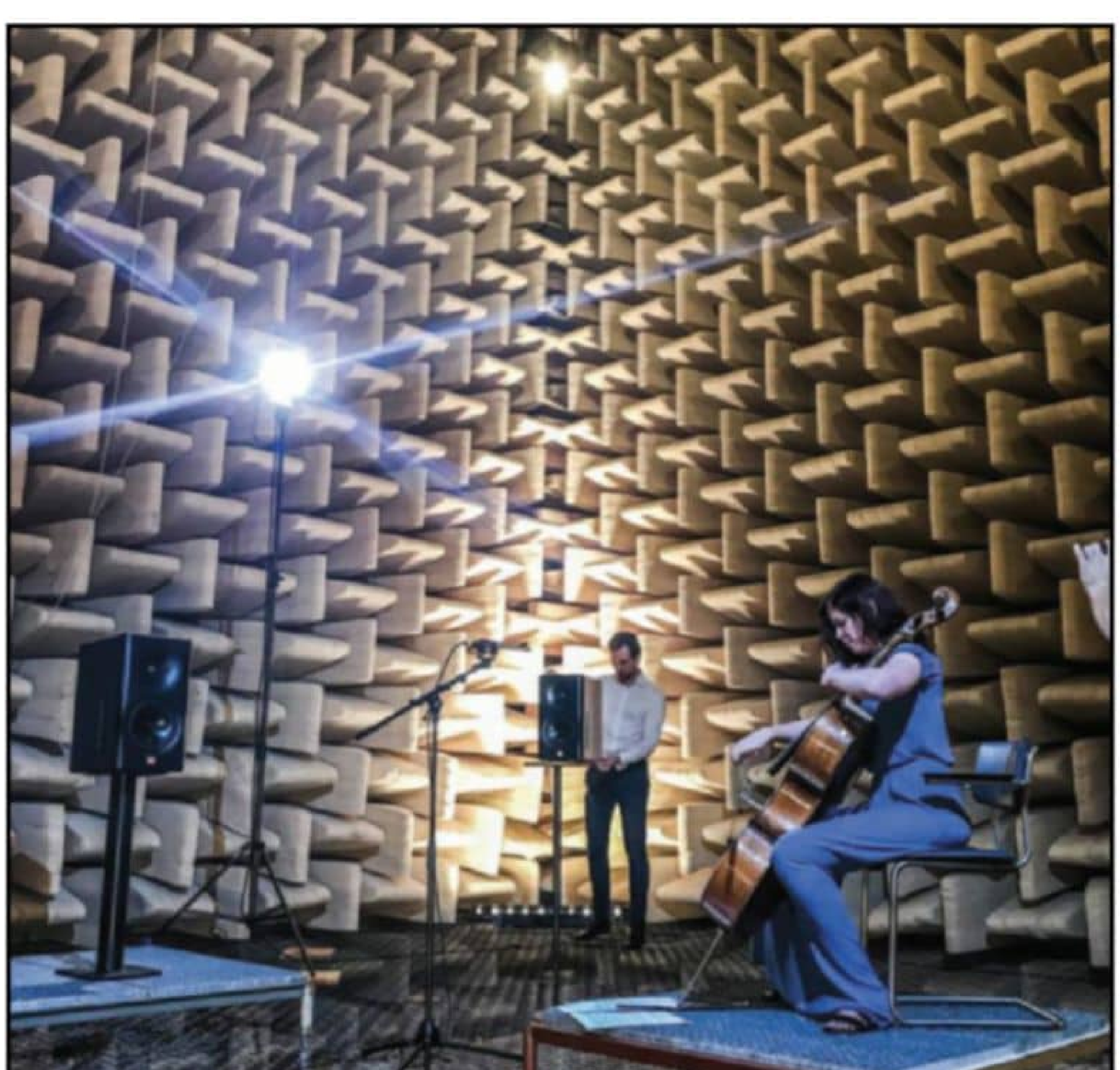
Getting to Know Dutch & Dutch

audioXpress visited Dutch & Dutch, a company founded in 2014 that managed to gain global recognition with a single product—the 8c speaker. As the company's founder Martijn Mensink explains, his philosophy of designing a no-compromise speaker that works together with the acoustics of the listening room is paying off.

By
Jan Didden
(audioXpress Technical Editor)

Everything in the design of the 8c was carefully considered to be part of the solution for room problems. From the proprietary waveguide tweeter with an acoustic cardioid midrange to the boundary-coupled bass drivers on the rear and active room matching.

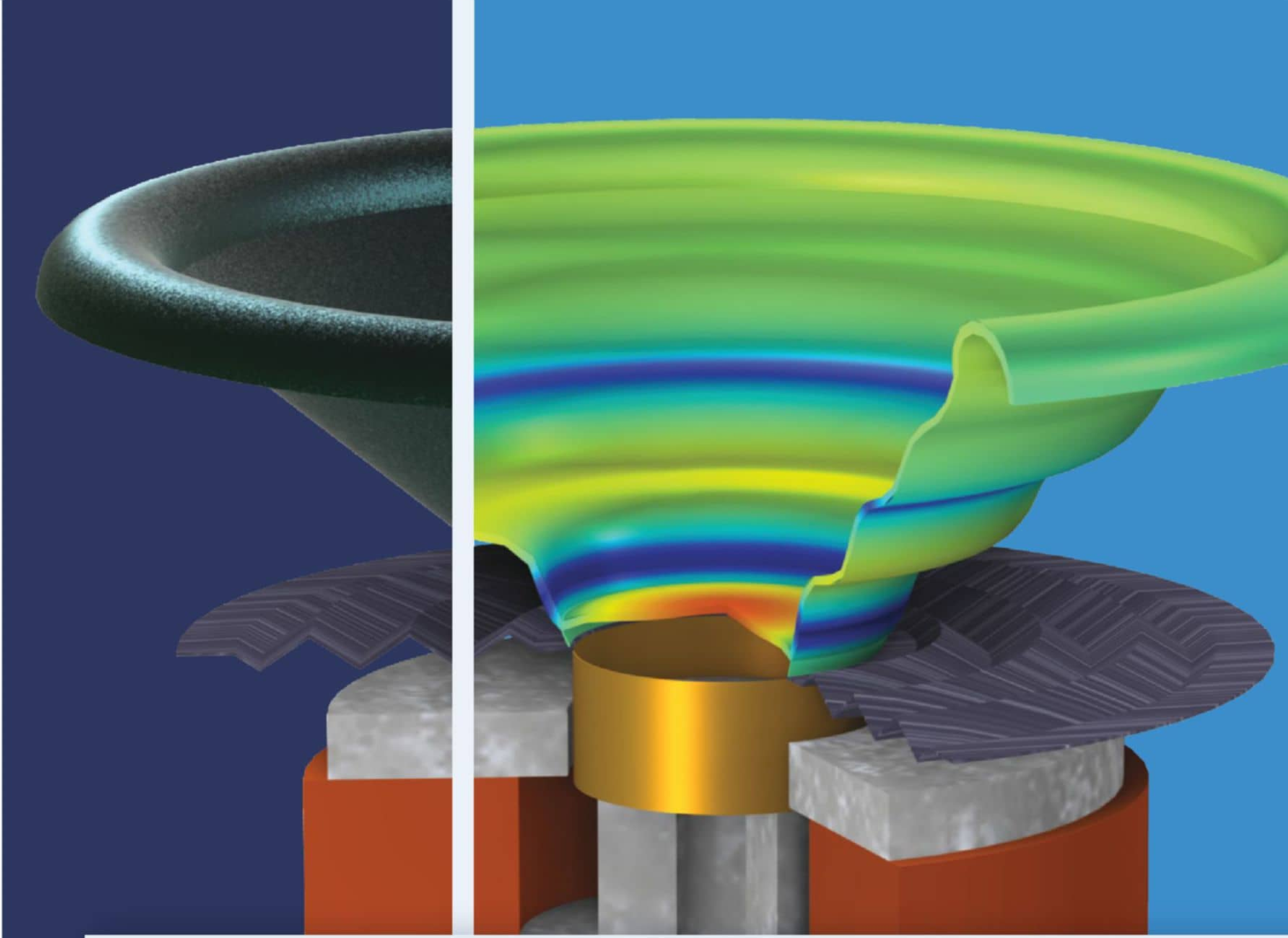


Early days for Dutch & Dutch (circa 2017) with Martijn Mensink testing the speakers at the Delft University (TU Delft)

Dutch & Dutch is a relatively young company that currently has a single product that is hugely successful and earned global recognition. The formula, as they describe it, is to design, develop, and produce next level audio systems that combine acoustical innovations, modern IT solutions, and Dutch design to deliver the most accurate sound reproduction in the world. The company's adage can also be summed up as: first study and make sure you understand the issues involved, then develop a product that solves them.

Located at the confluence of several major rivers in Rotterdam, The Netherlands, I spent a day with the team listening, talking, and philosophizing. I spoke directly with Martijn Mensink, the company's founder and "acoustic brain," whose passion for audio stems from his love of music, which he considers to be the most universal and emotionally stirring art form.

During his teenage years, Mensink had a transformative experience listening to his neighbor's high-end system, igniting his interest in audio. He pursued a dual education in physics and entrepreneurship, and throughout his university years, he immersed himself further in the audio world by working part-time at a high-end hi-fi store and starting his first audio company with friends from university. It was in this period that he discovered the concept of embracing room acoustics instead of fighting it.



The Foundation

At age 16 , Mensink learned that, when it comes to sound quality, the most important components of an audio system are the speakers. As he didn't have the money to buy better ones, he decided to build his own. The result was less than satisfying, and, with a group of friends from university, he decided to initiate a sort of collaborative competition to design the "best" speaker.

"At the time, I had my own listening room at my parents' home. We built a lot of speakers and one takeaway for me was that the audio electronics were not the primary determining factor in sound quality, but the speaker was," Mensink shares.

All Dutch & Dutch speakers are assembled and fully tested in-house. Here we see the electronics modules being assembled and units ready to be incorporated in the speaker cabinets, which are mostly assembled to order.

Another key insight came from the years he worked for an audio dealer in the East of The Netherlands where he did equipment delivery and demonstrations at the store and customers' homes. "I began to see a trend: Many customers started buying what they thought was the ultimate system. But after a time, they would get a feeling that something wasn't quite right, that something was missing. They would come back and purchase a new system, speakers, amplifiers, cables… hoping to find Nirvana this time. The company I worked for made a policy of accepting returns of previously sold equipment, to facilitate this."

"So, they took the new system that sounded so great in the showroom, set it up at home and, as is the way our brain works, would be very happy. Until, after some time they would again start to hear some shortcomings, and the cycle was repeated. Often, they were told that they should try a different speaker/amplifier combination, or exchange speaker cables, but I was skeptical about those seemingly simple solutions. Indeed, the new amplifier delivered much "faster" and cleaner lows! At first, at least, because, as I started to learn, after some time the doubts would inevitably reappear and another trip to the dealer was planned."

At the same time, this was a sort of business model for these companies. Encouraging a customer's hobby as a social ritual; to meet at the store, exchange stories, have a coffee together and have a good time. "This was always based on an underlying dissatisfaction, and at a certain point I asked myself, what's going on here? I came to the realization that the core basic issue was in the interaction of the speaker and the listening room. In the store we'd demo a nice floor stander, standing relatively free in a large space and it sounded fantastic. The customer then took the speakers home and positioned them where the space was available; a space that was almost always a lot smaller. The speakers ended up against a wall or in a corner and that didn't work, of course. This fundamental problem of how the speaker interacts with the room acoustics is almost universally ignored."

Room Judo

At that time, Mensink experienced the problem himself when he moved in with his girlfriend and got a pair of speakers that he thought were marvelous but couldn't get them to perform well in his living room. That's when he got back together with his friends from university and challenged them to try another approach.

Essentially, he realized that first, it's the combination of the speaker and the room that determines how it sounds. Second, the room is a given. Yes, a room can be treated for its most glaring acoustic problems, but it is not always possible in a family setting and its impact in the room aesthetics will hardly meet the "Spouse Acceptance Factor"—a very important criteria in every household. It was this realization that convinced Mensink to embark on the development of a speaker that would "adapt to the listening room, no matter how bad that room was" (within limits of course).

"My Dutch & Dutch business partner, Kevin Kleine, calls it 'room judo.' You use the 'power' of the room to work around its limitations."

"We succeeded in developing a system that worked quite well in my room. It was a complex system—satellite speakers and multiple subs, lots of wires and preamps and amps and DSP boards. We had a demo set, which we took around to prospective customers. There were a few challenges. First, we had to convince people that they had a problem. Then, we needed to convince them we had a solution. Then we showed up with a minitruck full of amps, subs, speakers, DSP boxes, and a lot of cables, taking a few hours to set it up. Next was tuning it with buggy DSP software—you get the point… It was what they call a 'hard sell'."

"We had given a lot of money to an outfit to develop tooling for series production, and they went bankrupt and took our money with them… That was a real low in my life, and I was not sure what to do next. But then I was approached by someone who was looking for somebody who a) knew a thing or

Cabinets in the woodworks ready to receive the finishing as ordered by customers two about speakers; and b) had some idea of how to run a business.

"I had just completed a master's in business and entrepreneurship, and already had the invaluable experience of what it means and how it feels to go bankrupt! After some wandering around, we pivoted (as it's often called in startups), and we decided to restart my previous project, as it had proven its basic soundness from an audio reproduction point of view. But it had to be simplified. It needed a lot of work to make it simple to use, easy to set up, and ideally just a pair of speaker boxes and nothing else. We needed to make it much more user friendly and get rid of the cable salad and separate amps and DSP boxes. That is how the 8c finally became the all-in-one speaker that it is, including all amplification, power supplies, A-D and D-A conversion where appropriate, and DSP processing. All electronics find their place in the bottom part of the enclosure, in an EMI-screened compartment."

An 8c speaker in the assembly with all drivers in place (1″ SEAS aluminum/magnesium alloy dome tweeters, 8″ SEAS midranges, and 2x8″ woofers), and now ready to receive the electronics and I/O module, and lastly the front baffle

Martijn Mensink inspecting one of the 8c baffles

The 8c

The Dutch & Dutch 8c is a speaker that is immediately striking in any of the available finishes and materials, which contributes to the appeal of the overall design—modern and clean. This appearance doesn't detract from the fact that it is a fully integrated system, more like a platform that combines a lot of technology. Mensink shared more details of that product development for us.

"I already knew from the previous project that the best way to shape the speakers' dispersion pattern was with a waveguide tweeter and a cardioid pattern on the woofer. The tweeter on the front panel is augmented with a midrange, which has its own internal enclosure and a pair of tuning slots on the sides. Those side vents produce a wavefront that is derived from the back energy coming off the midrange driver, to cancel out the midrange energy that tends to bend around the cabinet corners. In this fashion, the side and backside of the loudspeaker cabinets do not produce net energy and therefore do not excite the listening room with reflections," he explains.

The small box of the 8c holds a total of 1kW of amplifier power! There is an analog XLR input which can be adjusted for two sensitivities (+4dBu or -10dBV). Another XLR provides an AES3 digital input with a Left-Right-Mono selection switch. There's also an analog XLR output for further processing or if you wish to connect a separate subwoofer. The connection for the streaming functionality uses Ethernet though an RJ45 connector.

One of the key aspects of the 8c recipe is that there are a pair of woofers on the back. This allows the speaker to be placed relatively close to the rear wall. Not only can the owners of these speakers place them where it is usually more convenient for them, Dutch & Dutch recommends that they do it. The woofers radiate onto that real wall and their output is reflected, which acts as if there is an extra pair of woofers.

"We do adjust the timing in the DSP to correct for the shift of the virtual source of those woofers due to the reflection. This is an ideal setup, as the user can set the rear speaker-to-wall distance in an app. The speaker and the wall are time-aligned and become a single system. This so-called boundary coupling avoids the usual destructive interference between the direct sound and the reflected sound and increases the low-frequency headroom by up to 6dB. It also increases the directivity in the low-frequency region, to match that of the cardioid midrange that takes over above 100Hz," Mensink explains proudly.

Rather than fight the acoustics of your room as most speakers do, the Dutch & Dutch 8cs are designed to work in concert with the room. Thanks to technologies such as the cardioid midrange and boundary coupling in the bass, the 8cs can be placed close to walls, or even in the corner. They can be used in big rooms or small ones—the 8cs basically don't care and will deliver great sound regardless. And besides the acoustic technologies, the 8c has a DSP on board that is used to adapt the way the 8cs radiate their sound to the room and the placement in the room. Dutch & Dutch calls this RoomMatching. There are settings to adjust to nearby boundaries, and you can even correct the room modes.

So, there you have it, the waveguide tweeter operates above 1.25kHz, the cardioid midrange works in that range down to 100Hz, and the Boundary Coupled Bass woofer system takes over below that. And while the distance to the rear wall can now be compensated for as far as the sound pattern is concerned, you still do get an increase in available low-frequency power as you move the speaker closer to the rear wall. Room Judo—using the room's strength to conquer it! After the 8cs are automatically discovered on the local area network (LAN), using the control app users have full control of settings like gain, distance to front and side walls, updates and many parametric filters. Internet connectivity is not even required.

To optimize the whole system, Dutch & Dutch recommends doing acoustic in-room measurements in the listening area with Room EQ Wizard (REW), calculate correction filters for room modes and shoot those straight into the 8cs via the LAN.

To do this, Dutch & Dutch worked with John Mulcahy, the creator of the software, to integrate the 8c with REW. All that is needed is a computer with REW and a USB measurement microphone, such as the miniDSP UMIK-1. When RoomMatching is done for a certain listening position in your room, you won't have to touch it anymore. However, you can make multiple RoomMatching Profiles for multiple listening locations, or larger listening areas. Normally, users can create a RoomMatching Profile for the main listening position, one that sounds good throughout the room, and one that sounds great when working from home behind your desk.

The control app will allow you to add the proverbial salt and pepper, by changing the tonal balance to your liking with some EQ functions. Even better: users can select a particular "voicing" to

Fully assembled 8cs ready for testing in the assembly line

Finished speakers ready to ship. The product packaging is as thoughtfully designed as the Dutch & Dutch speakers contained within.

White finish Dutch & Dutch 8c speakers with floor stands support events such as parties or a romantic evening when neutral reproduction is not a requirement.

Dutch & Dutch also realized that this controlled cardioid dispersion would be very valuable in recording studios and other Pro Audio related settings. That explains why there are two slightly different 8c types: 8c and 8c Studio. The differences are mainly related to the available color combinations, streaming capability, and the availability of a flying mounting for the Studio type.

"And so, the circle has been closed," Mensink adds. "We have customers that have used an 8c for many years and are totally happy with it. Still, sometimes they visit my old dealership and hang out and drink coffee. People are interesting!"

Martijn Mensink explains how the 8cs adapt to the room they are in and can be placed (very) close to room boundaries without losing performance.

The 8c Studio cabinet is made of MDF wood instead of solid oak, which enables a series of optional mounts (on M10 threads) that are required in studio and professional environments. The Studio model also doesn't feature streaming audio over Ethernet of the home 8C.

Keeping the Speakers in Sync

Signal distribution to the speakers is a mix of wireless and Ethernet. The two speakers are connected through an Ethernet link to the customer's router. As mentioned, the user interface is running on an app on a smartphone, tablet, or PC. That opens the whole world of streaming, NAS storage, and local storage to the listener. A key issue in such a setup is keeping the two speakers synchronized, with all the unpredictable network delays and often variable latencies on the router/ Ethernet connections. Dutch & Dutch has spent considerable efforts to solve that problem. "We start by giving each audio frame a time stamp, so we can ensure that each sample is being replayed at exactly the correct instance in time," explains Kleine, Dutch & Dutch's software guru.

That is, of course, assuming the two speakers' clocks run at exactly the same speed, which in general will not be the case. The timing in the two speakers will slowly be drifting apart due to very slight differences in their internal clock speed. "We were able to detect that drift, using the sample timestamps and the internal local clock. We use that information to slightly tune the local oscillator through a PLL [phase-locked loop]. This is a simplified view of course but this system allows us to keep the two speakers synchronized to within one audio sample. During the development phase we build a diagnostic tool, which gives us a snapshot of what's happening on the network when things get out of hand. That was very valuable during the development process but I'm happy to say that nowadays it stays totally silent!"

The Sound of Controlled Dispersion

We spent some time listening to a pair of 8cs in Dutch & Dutch's listening room. I was able to play some of my own favorite music, including well-known tracks from Cassandra Wilson, Mark Curry, and Ulla Meinecke.

I am used to a well-resolving ESL-based system at home, but the 8cs had no problem convincing me with a very realistic, well-balanced, and wideband reproduction with impeccable placement of voices and instruments. The wandering of voices and instruments with changing frequency that so often happens in lesser systems was totally absent in the

8cs, confirming that the team's efforts to control and shape the dispersion throughout the audio frequency range was spot on.

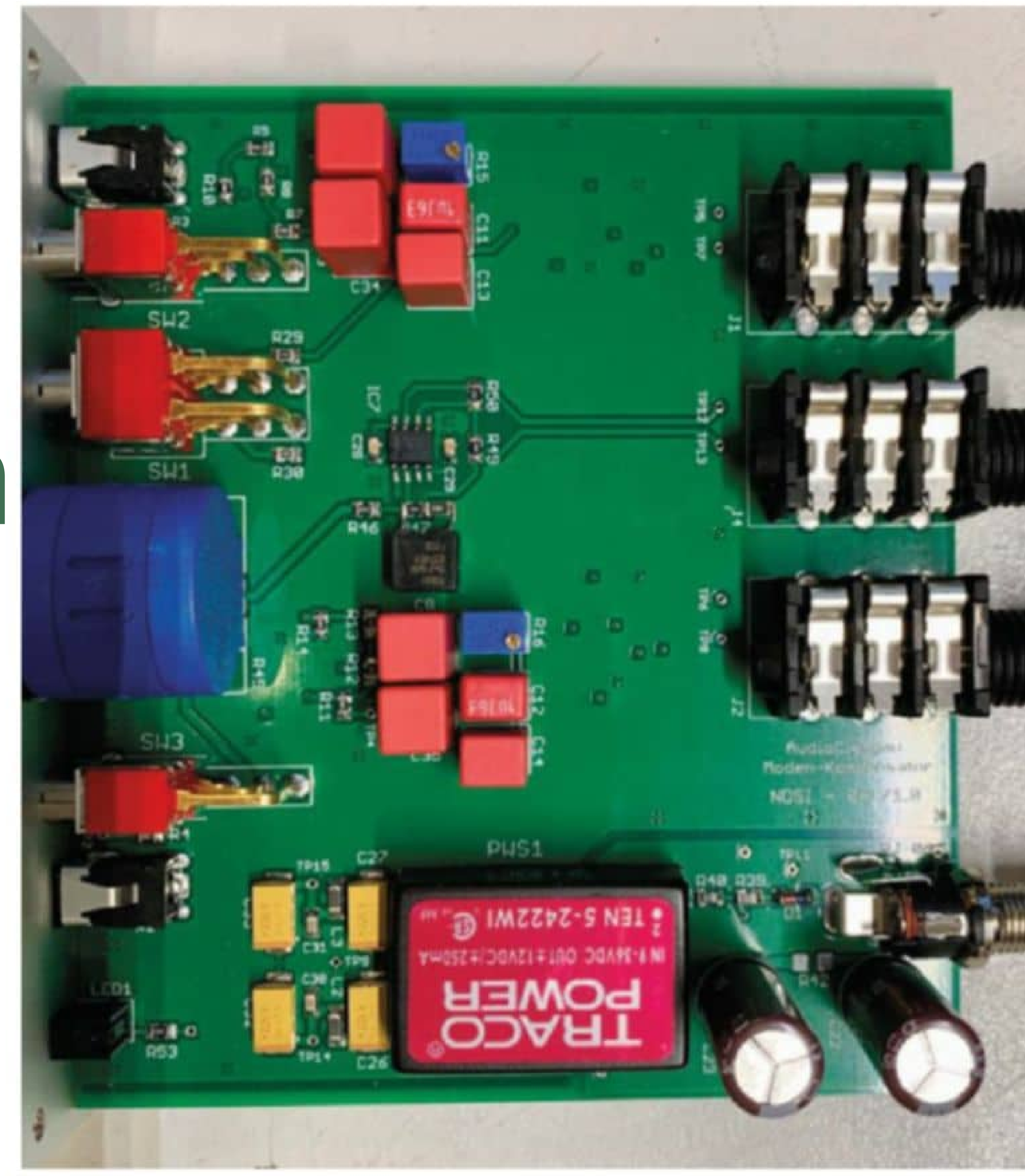
On my way out, after a very informative and enjoyable visit, I chatted with Mo and Mehdi, who were preparing another batch of 8cs to be shipped to waiting customers. In this company's case, they don't stop there. Dutch & Dutch offers (and strongly recommends) their Remote Commissioning Service (RCS), availability everywhere in the world.

After the new owner receives his speakers and places them in the intended room, one of the company's engineers will get in touch and set up a remote TeamViewer session over the Internet to make the necessary measurements. Customers who take the RCS option receive a professional measurement USB microphone. During the interactive session the customer is guided on the procedures, including moving the measurement microphone around the room to gather the data needed to make the right room correction settings.

Dutch & Dutch
www.dutchdutch.com

The connectivity in the 8c was also carefully planned, including the use of balanced analog, digital AES3 inputs, plus Ethernet LAN, allowing users to create groups of speakers in zones or multichannel configurations (e.g., Dolby Atmos). AES3 inputs allow balanced Type 1 connections or SPDIF with PCM up to 24-bit/192kHz.

Measuring Loudspeaker SPL Response and Harmonic Distortion at Low Frequencies



This article discusses how to measure loudspeaker SPL response and harmonic distortion at low frequencies in a standard laboratory environment, using a method that combines near-field measurements with two identical microphones.

By
Reinhold Lutz

There are three challenges when measuring the frequency response and the harmonic distortion of a loudspeaker in a standard laboratory environment: reflections from acoustically hard boundaries, such as walls, floor, and ceiling; room resonances (a.k.a. room modes) heavily influencing the lower frequency range; and ambient noise reducing the dynamic range and accuracy of the measurement.

Photo 1: AudioChiemgau has developed an affordable ModeCompensator comprising two calibrated MEMS microphones and the necessary electronics to perform measurement tasks.

Solutions to these problems are known. The most expensive solution is an anechoic chamber. However, at low frequencies these chambers need to have extremely large dimensions. A frequency of 20Hz correspond to 17m wavelength and absorbers are effective up to a quarter wavelength from the boundaries. That is the reason why most anechoic chambers have a lower corner frequency of around 70Hz.

Alternative techniques, such as quasi-anechoic measurements, using a ground plane, impulse testing, MLS testing and Log-Swept Sine measurements, are also well known.

In 2000, Farina introduced the Log-Swept Sine measurement [1]. Today, it is the most used technique for loudspeaker testing, as it shows a good dynamic range and allows the simultaneous measurement of frequency response and harmonic distortions.

A theoretical and experimental comparison of several techniques can be found in a 2002 Journal of the Audio Engineering Society article written by G. Stan, J. Embrechts, and D. Archembeau [2].

For a standard laboratory environment, the Log-Swept Sine technique needs to be combined with time gating of the measurement signal. Otherwise, reflections from hard boundaries, room resonances,

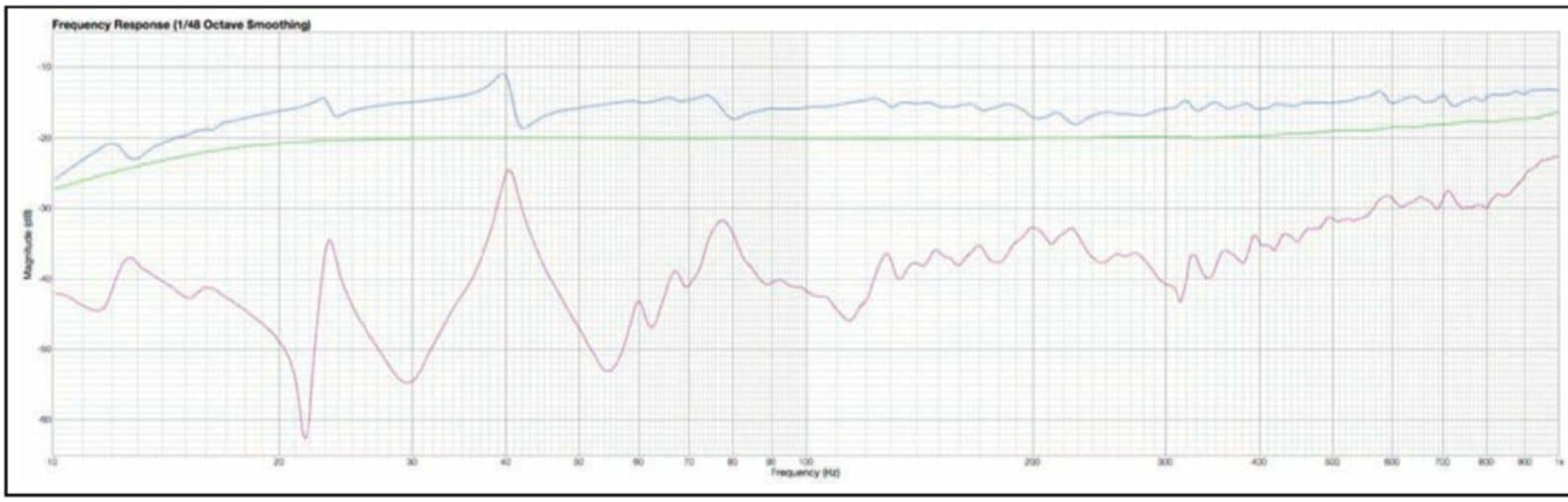


Figure 1: The sound pressure level (SPL) from the measurement microphone (blue) and the mode microphone (violet) are shown here. The derived SPL, excluding room effects, is shown in green.

and ambient noise would significantly impact the accuracy of the results.

However, time gating tackles mainly the first challenge. Sufficient short time windows limit the resolution of the frequency measurement. The time-bandwidth uncertainty [3] gives the frequency resolution as the inverse of the time window duration. In simple terms: When one wants to measure a frequency of 1Hz, one needs to measure for a time duration of 1 second. Also, windowing may truncate the impulse response before it has decayed to a sufficiently low level.

Most lab rooms with the usual distances from the device under test to the walls, ceiling, or floor generate reflections within 10ms. A time window set to 10ms limits the frequency resolution to 100Hz. Measurements down to 20Hz with a sufficient frequency resolution are impossible.

It is therefore sensible to find a solution to increase the time window to say 10s and solve simultaneously the second and third challenge. Such a solution needs to work in the time domain before applying the time window respectively the Short-Term Fourier Transformation (STFT). This article will describe a technique we developed to solve these difficulties.

The Technique

The basic idea is to perform a near-field measurement with two identical microphones. A Measurement microphone is positioned close (say 5cm distance) from the loudspeaker membrane and a Mode microphone is positioned at a short distance (say 5cm) from the measurement microphone along an axis perpendicular to the loudspeaker. **Photo 1** shows the geometry for two microelectromechanical systems (MEMS) microphones.

Both microphones record the sound pressure from the loudspeaker. The upper Mode microphone will record a lower level than the lower measurement microphone according to the one over distance law.

Both microphones record room modes, reflections, and ambient noise at nearly equal level, as those sound sources are relatively far away from both microphones. Subtraction of the recorded signals in the time domain eliminates (or reduces to an insignificant level) the room influence on the measurement.

Practical Measurements

The graph shown in **Figure 1** shows typical frequency responses measured with this setup. The measurement microphone was located about 5cm from the membrane and the mode microphone was located exactly 5cm from the measurement microphone.

This measurement technique solves all the three previously mentioned challenges. By subtraction of the two signals in the time domain, reflections and room modes are cancelled and as both microphones receive the ambient noise with equal level, that ambient noise is cancelled too. This results in a significant increase in measurement dynamics, especially at low frequencies.

Without reflections and room modes present in the time domain signal, the time window for the STFT can be freely set (e.g., to 10s corresponding to 0.1Hz frequency resolution). The graphs used in this article have been recorded using our ModeCompensator measurement test set (see **Photo 2**). The ModeCompensator comprises two calibrated MEMS microphones and the necessary electronics to perform the Time Domain subtraction and nulling of the signals.

Photo 2: The AudioChiemgau ModeCompensator enables the measurement of the SPL with very high frequency resolution and accuracy down to 10Hz. It is intended to be used in combination with measurement equipment such as a computer with audio interface and an appropriate evaluation software.

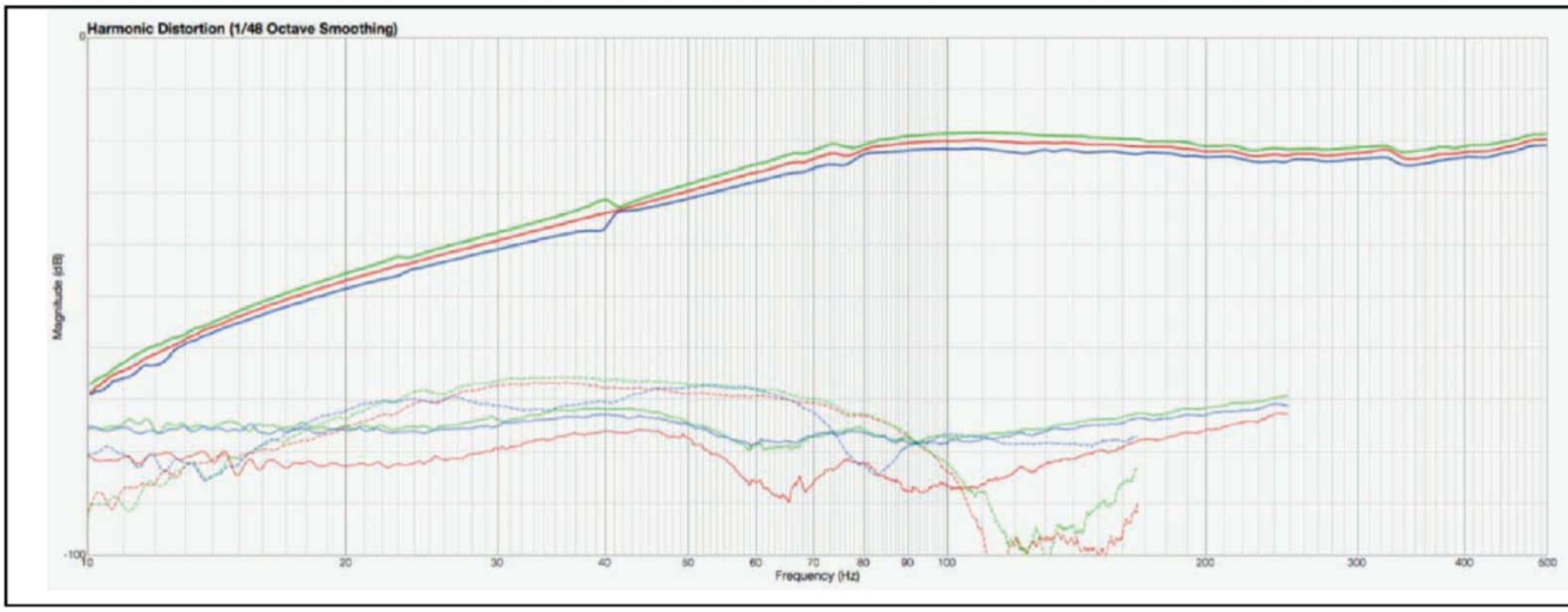


Figure 2: Sound pressure level frequency responses for three different adjustments. (Blue: under-compensation of the room modes; Green: over-compensation of the room modes; Red: correct compensation of the room modes. The harmonic distortion curves are shown at the bottom.

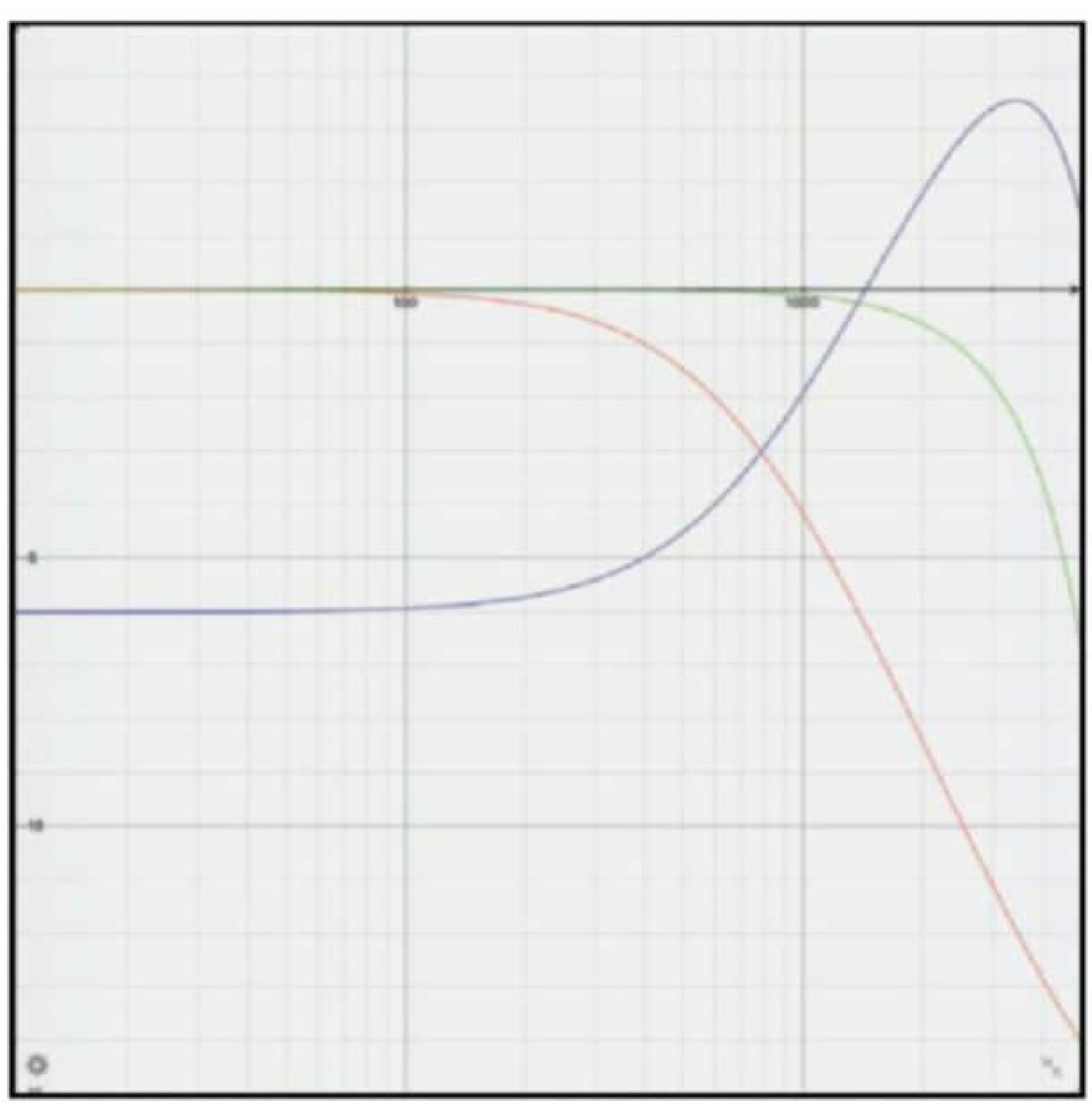


Figure 3a: The combined frequency response of the two microphones are shown in blue. (Also see text.)

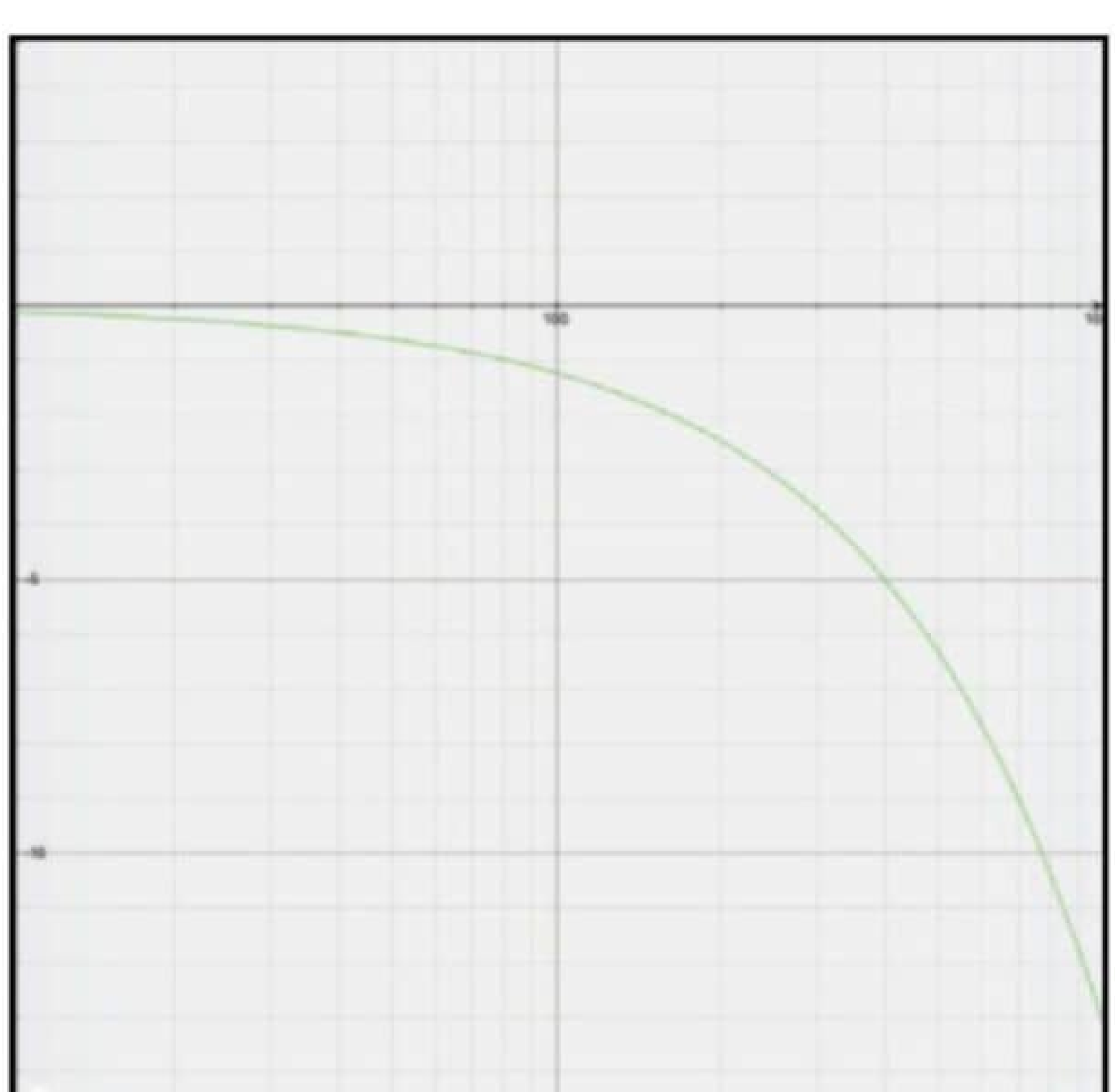


Figure 3b: This is the inverse low-pass filter.

The time domain signal subtraction is illustrated in **Figure 2**. The ModeCompensator adjustment dial is adjusted until the room modes become invisible. The inversion of their phases at this point is a clear and accurate indicator.

Two microphones at approximately 5cm generate a comb filter frequency response as shown in **Figure 3**. In Figure 3a, the combined frequency response of the two microphones is shown in blue. The subtraction causes a 6dB loss at low frequencies due to the one-over-distance law and the chosen distances. The distance between the microphones leads to a frequency proportional phase shift.

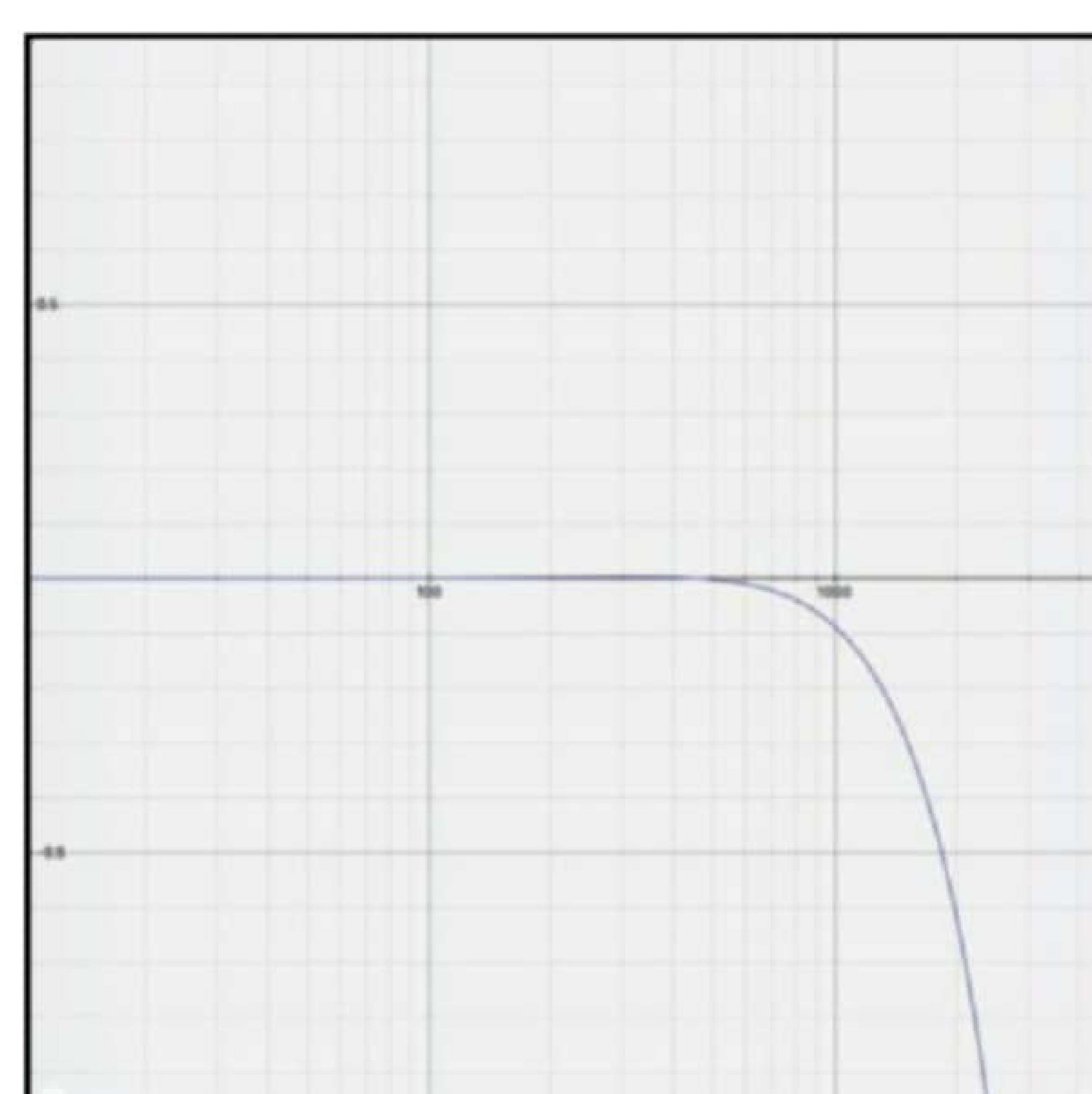


Figure 3c: This graph shows the resulting frequency response. It is accurate within +0 to -0.1dB up to 1000Hz.

Thus, at higher frequencies, the subtractions turns into an addition resulting in a level increase with frequency. That frequency response can be corrected in the time domain by an inverse low-pass filter (red). The resulting frequency response (green) is accurate up to 1000Hz. Figure 3b shows a closer look to the inverse low-pass filter. Finally, Figure 3c shows a close look at the resulting frequency response. It is accurate within +0 to -0.1dB up to 1000Hz.

The Method

The measurement technique presented here can be performed easily with two identical microphones, a dual input audio interface and software to perform logarithmic sweep measurements. However, such a setup will usually not allow the necessary inverse filtering as shown in Figure 3. Therefore, AudioChiemgau has developed an affordable ModeCompensator comprising two calibrated MEMS microphones and the necessary electronics to perform the tasks as shown in Photo 1.

Figure 4 shows a block diagram of the ModeCompensator. A user guide with further details can be downloaded from the AudioChiemgau website (www.AudioChiemgau.de).

Christopher Struck and Steve Temme presented a method whereby the full-range frequency response of a loudspeaker is measured by combining its separately measured low-frequency response with its mid and high- frequency response [4]. Our Mode Compensation technique now enlarges the frequency range of the near field measurement down to 10Hz with 0.1Hz frequency resolution, only limited by the microphones used. As shown in Figure 3 the upper frequency limit is determined by the distance of the two microphones and reaches 1kHz at 0.1dB accuracy.

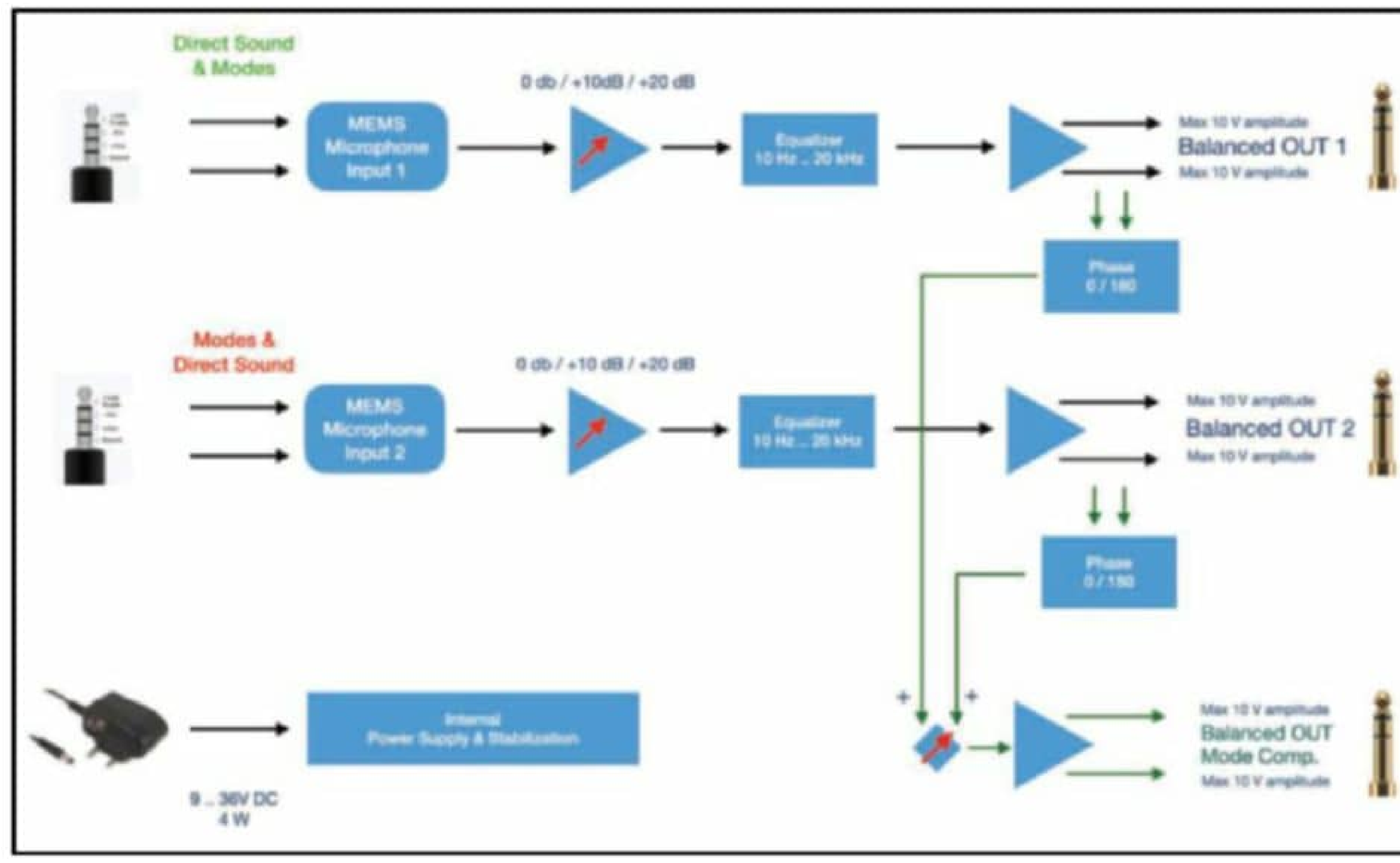


Figure 4: This graph shows the Analog Signal processing of the AudioChiemgau ModeCompensator.

Summary

Appropriate signal processing in the time domain enables users to overcome the well-known limitations of time gated near-field measurements. With two microphones at defined distances to the loudspeaker membrane and difference building in the time domain before applying the time window/STFT, accurate measurements of the frequency response of the sound pressure level and harmonic distortion become possible in a standard lab environment. Level resolutions of 0.1dB and a frequency resolution of 0.1Hz can easily be achieved between 10Hz and 1kHz. In addition, the suppression of ambient noise results in a high measurement dynamic range required for the measurement of low-level harmonic distortions.

About the Author

Reinhold Lutz studied Communications Engineering at the Technical University of Munich and received a doctorate degree (summa cum laude) in engineering. He subsequently spent 30 years in the Space industry supporting German, European, and American companies. He served as Senior Vice President Airbus Defence&Space in Europe and the US. Reinhold lectured on system engineering for optical and SAR remote sensing space systems at the University of Delft and was co-author of the Space Mission Analysis and Design (SMAD) textbook. Reinhold also served as managing director for Astrium GmbH and on several boards of directors. He retired in August 2016 from Airbus DS and now develops high-end loudspeaker systems for audio enthusiasts.

References

[1] A. Farina, "Simultaneous measurement of impulse response and distortion with a swept sine technique," presented at the 108th Audio Engineering Society (AES) Convention, Paris, France, 2000.

[2] G. Stan, J. Embrechts, and D. Archembeau, "Comparison of Different Impulse Measurement Techniques," *Journal of the Audio Engineering Society* (JAES) Volume 50, No. 4, 2002.

[3] R. Hoffmann, "Grundlagen der Frequenzanalyse, Eine Einführung für Ingenieure und Informatiker," 2. Auflage, Expert Verlag, Renningen 2005.

[4] C. Struck and S. Temme, "Simulated Free Field Measurements," presented at the 93rd Audio Engineering Society (AES) Convention, 1992.

The AudioChiemgau ModeCompensator

A Quick Basic Test

By
Rens Tellers and Jan Didden

The principles used in Dr. Reinhold Lutz's article included in this issue are applied in a solution called the ModeCompensator for precise measurement of the sound pressure frequency response in the presence of room modes, available from Lutz's company, AudioChiemgau. Jan Didden and Rens Tellers performed a basic functional test to find out how well they could make it work.

When we received Dr. Reinhold Lutz's article proposal, we were surprised that we hadn't previously heard of the ModeCompensator. Most *audioXpress* readers will be familiar with the issues of trying to make a good, reliable measurement of low-frequency response and distortion in loudspeakers and drivers. One of us has even been paying user fees for the local university's anechoic room to do such measurements.

MLS excited low-frequency measurements or close mike low-frequency measurements and splicing that response to the mid/high response to get an audio band response are known techniques. Swept measurements need careful attention to finding a compromise in windowing the impulse response to exclude reflection without throwing away valuable low-frequency response information. So, naturally we were quite curious to find out if a simple, low-cost unit like the ModeCompensator would take care of all that, so we asked for a sample and decided to do a basic functional test.

Photo 1: Measurement setup with microphones positioned 5cm and 10cm above the cone

The Setup

We set up the unit with the two microphones as described in the User Guide using a guitar amp woofer on a baffle as shown in **Photo 1**.

The two microphone outputs were connected to the inputs of an Audio Precision APx525. After the setup, we connected the ModeCompensator difference output to one of the AP inputs and conducted the measurement sweep. All traces (individual mikes and the difference output) were recorded in the same graph for comparison. Initially, we couldn't get it to work as it should. We would do the measurement run and then using the AP time domain measurement adjust the precision dial on the unit for a minimum at the difference output. We just got some low-level chaotic curve. After some communication with Dr. Lutz, we realized our procedure was incorrect. You do not want a perfect subtraction of the two curves to zero output. Consider, mike 1 has as output: V(c) + V(a&m), while mike 2 receives: V(c/2) +V(a&m) where V(c) is the direct signal from the driver cone, and V(a&m) is the signal resulting from ambient noise and room modes. When you correctly subtract those two levels you get V(c/2), not zero! Obvious in hindsight, but since this is a very unusual unit, this could be explained more clearly in the User Guide [1]. In the end, we chose an attenuator setting very close to the factory-recommended setting, which is provided for each unit.

Does It Work?

To start with the conclusion: Yes, it definitely does work. The idea behind it is sound, and we could reliably measure the low-frequency response of the woofer with almost no ambient noise or room reflections visible. But let us put it in perspective, with **Figure 1**. The top and bottom curves are measured by the two microphones, and we see some low-frequency noise and minimal reflections below 80Hz or so. More reflections and modes can be seen up to 300Hz. If you look at the difference response (middle curve), you see that the difference processing has neatly canceled those ripples and produced a nice smooth frequency response for this woofer. We were surprised that we did not see more reflections and modes below 100Hz, as shown in Dr. Lutz's article elsewhere in this issue. Our measurement space wasn't particularly large or damped and had ample angles and edges to cause reflections (**Photo 2**).

So, let us assume we had performed this measurement with just a single microphone, and gotten the top curve shown in Figure 1. We would have seen a reasonable correct response at the low-frequency side down to 20Hz, albeit noisy, mixing noise and room modes with any driver resonances. With the ModeCompensator, possible driver resonances can be easily identified.

But there's one more aspect to this. We did our measurements at a relatively low SPL. We found that the ripples and modes were a bit more pronounced at low levels. The positive side of that is that you can do these measurements without having to grab the hearing protection all the time, and even without disturbing fellow workers close by. Especially in a production environment, that is a great plus.

The Verdict

We met a lot of skepticism from colleagues when we mentioned an article and review of this unit, varying from "it won't work" to "nobody needs that." But being sure that resonances and ripples seen in the driver response do not come from ambient noise and room modes is a valuable result. There's also the substantial advantage of being able to do these tests at very low levels without losing information. So, we feel that, especially at the reasonable cost of around $500 including a pair of calibrated electret microphones, this is a useful unit if you routinely do low-frequency woofer or speaker measurements. For more information, visit https://audiochiemgau.com/modecompensator-01

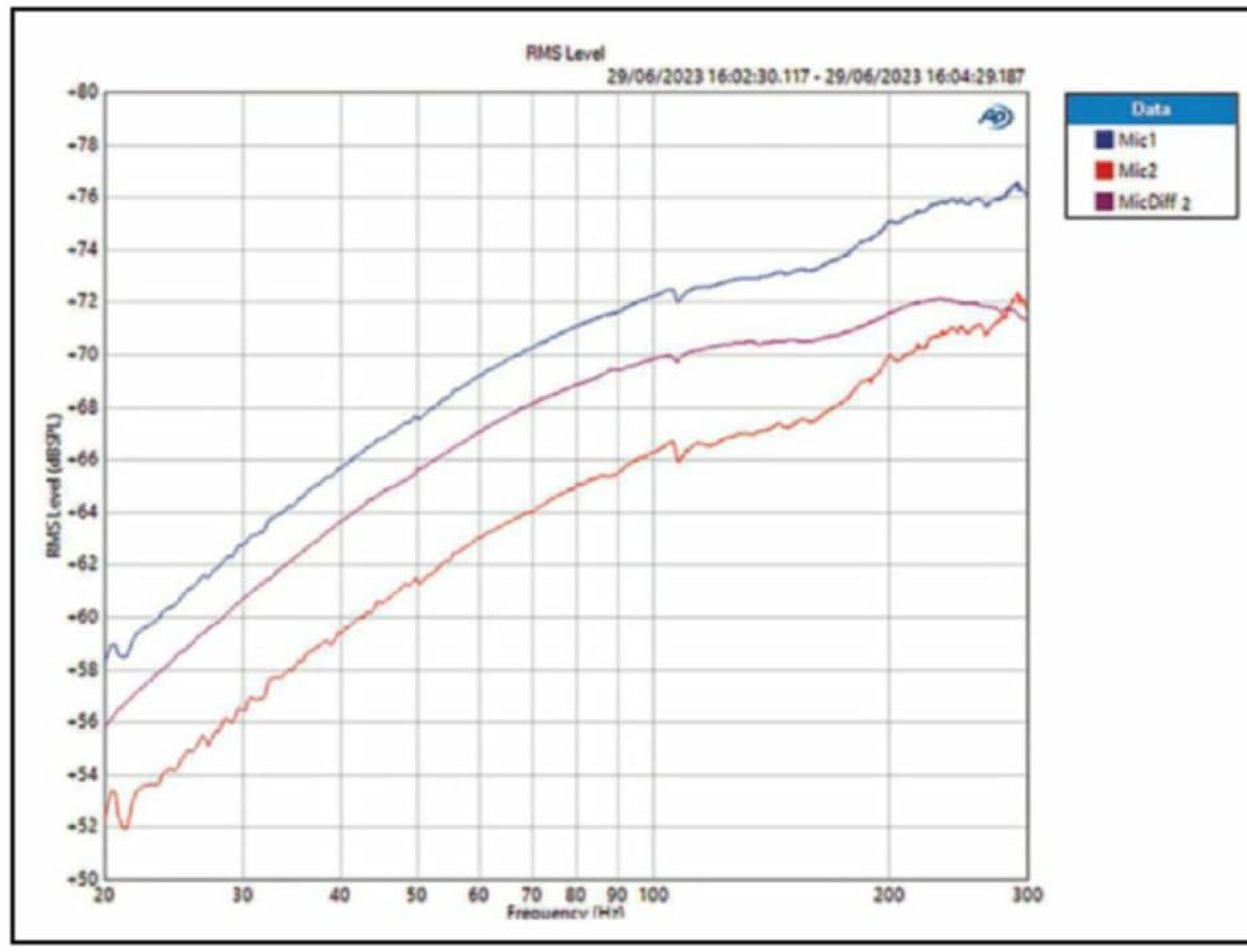


Figure 1: Representative measurement curves; measurement mike (top, blue), compensation mike (bottom, red) and difference response (mid, mauve).

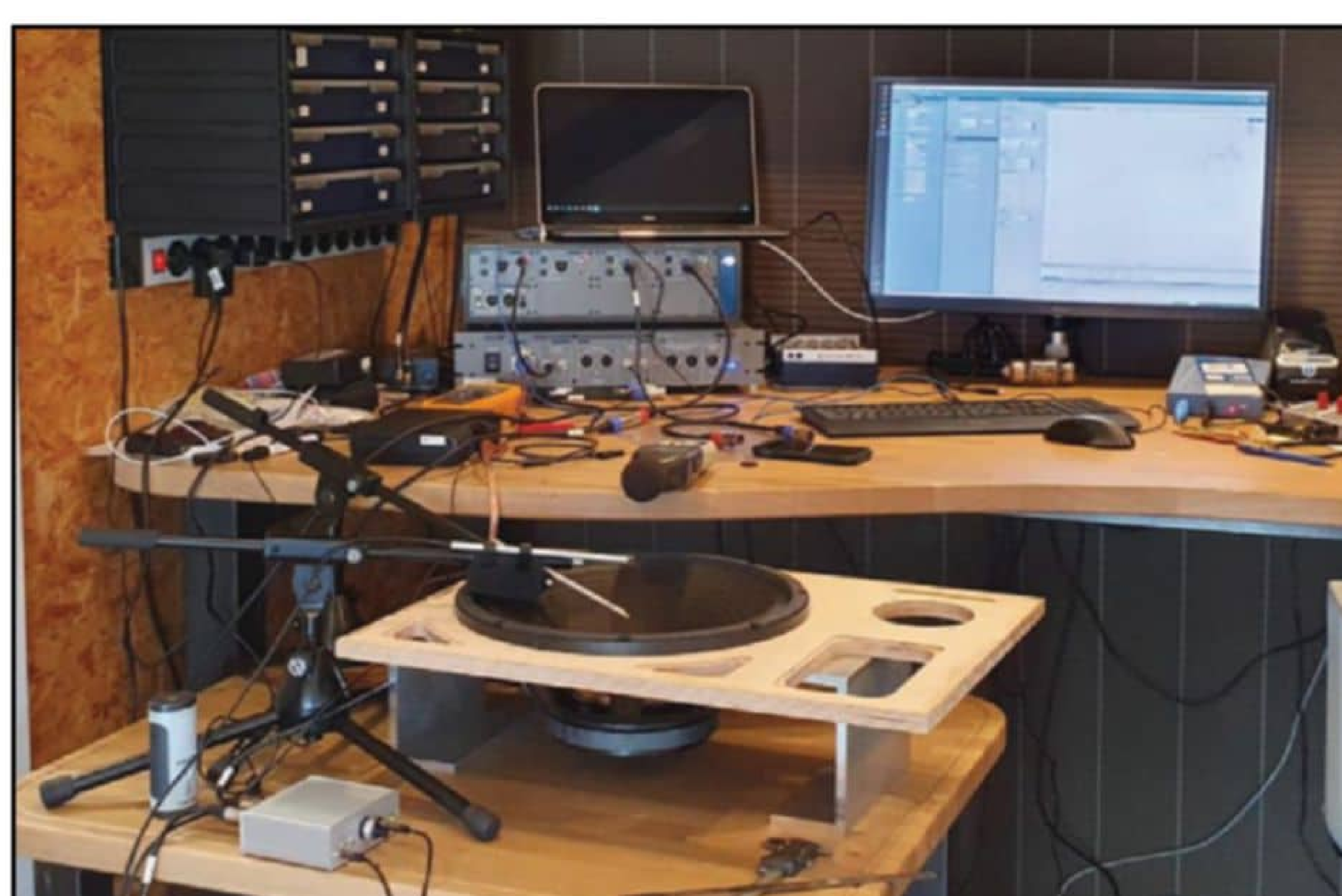


Photo 2: The measurement room has ample edges and corners causing reflections.

About the Author

Rens Tellers runs a company on electrical automation, with more than 15 years of experience in electrical engineering that he translates into the audio world. He currently focuses on improving audio amplification and measurements thereof, which has earned him member status of the Audio Engineering Society (AES). Rens has been an active musician (bass guitar and piano) and music enthusiast his whole life, with extensive work in the recording and mixing of music. One day, he hopes to dedicate his entire time to making music sound better in all aspects.

References

[1] With the measurement microphone 5cm above the cone, and the mode mike 5cm above the measurement mic, the mode microphone receives half of the cone level see the User Guide from the AudioChiemgau website, https://audiochiemgau.com.

Passive Variable Acoustics

The Elusive Optimal Reverberation Time

By
Richard Honeycutt

For decades, competent acousticians have been able to design performance spaces optimized for specific program material. Now it has become possible to tweak the acoustics of a space for different program materials! This technique is called "Variable Acoustics."

In my very first Sound Control article, "We Really Have (or Want) Great Acoustics…" published in the November 2013 issue of *audioXpress*, the point was made that great (or good) acoustics is an inadequate description of the desired outcome for any venue: Everything depends upon the purpose of the venue.

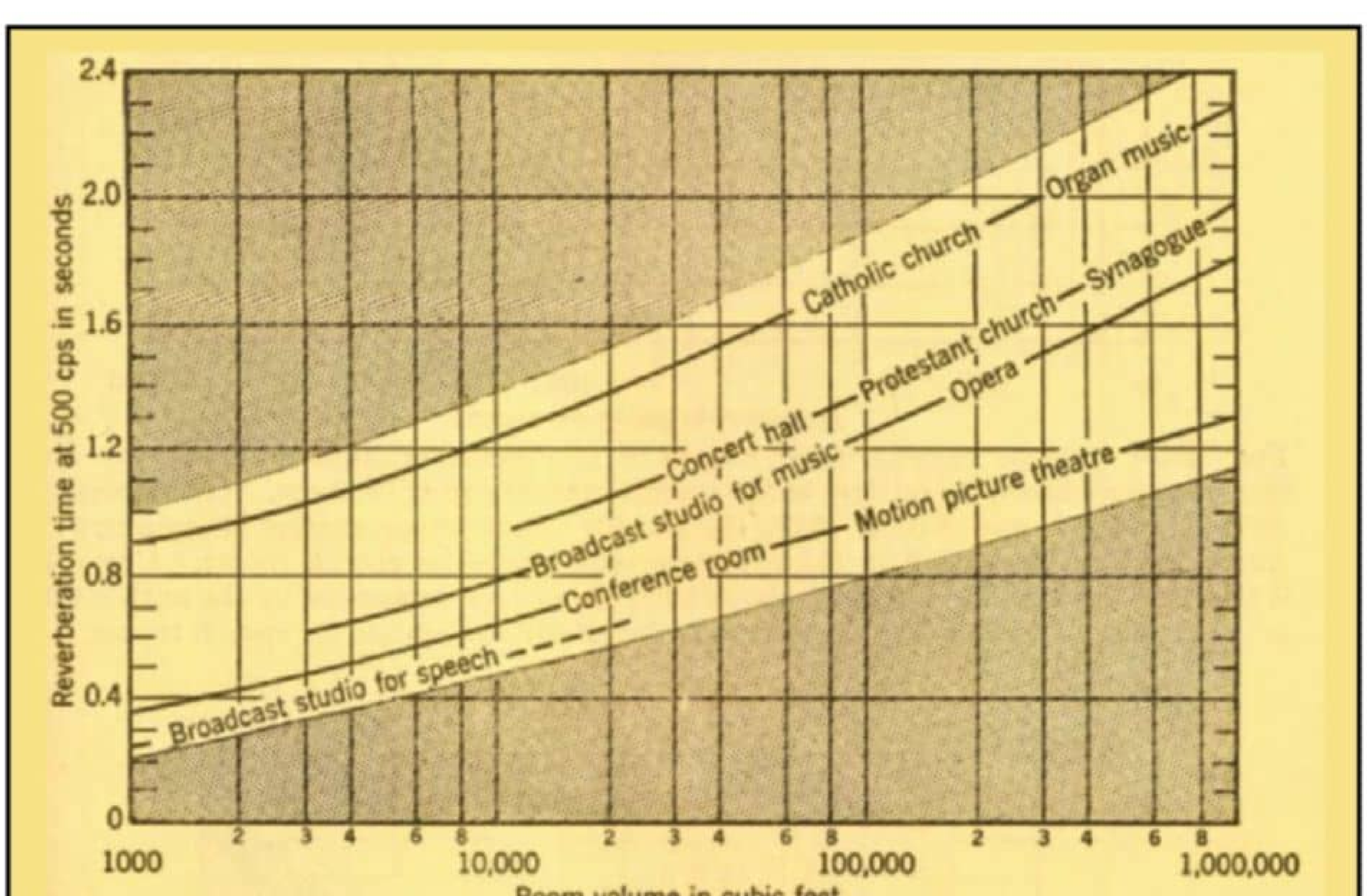


Figure 1: The optimum RT for a room depends upon its cubic volume and its intended use.

At least as early as the publication of Vern Knudsen's acoustics textbook in 1932 [1], acousticians recognized that room size, reverberation time, and the provision of beneficial early reflections must be taken into account in the acoustic design of any venue, and that optimum values for all of these conditions depended upon whether the venue was to be used primarily for speech or music, and if music, whether it was solo instruments, symphony orchestras, or oratorio (perhaps involving pipe organs).

Leo Beranek's landmark text [2] included a valuable chart (see **Figure 1**) specifying optimum RT values for various types of venues. Note that this chart shows the dependence of optimum RT upon the cubic volume of a hall. Presumably, the human ear/brain system, which can estimate the size of a room based upon the time delay of the initial reflections after the first arrival of direct sound, expects larger rooms to have a longer RT. Rooms not meeting this expectation tend to sound unnatural. This dependence of optimum RT upon cubic volume applies in addition to the variation of optimum RT upon program type.

Table 1 [3] indicates optimum RTs for typical venues, organized by the purpose of the room. While the information shown is less specific and comprehensive than that shown in Figure 1, it does give an idea of the typical cubic volume for venues intended for each of the purposes covered.

Beranek published a paper [4] in which he pointed out other important factors besides RT in user ratings of music venues. These factors include the shape of the hall, the distribution of acoustical absorption, sufficiency, and direction of arrival of early reflections, sound strength G, total sound energy, and hall dimensions. All the rated halls are well-respected venues usually used for standard concert repertoire. The ratings were compiled from perceptual evaluations by multiple listeners. The paper did not include information on the style of music being performed. We can probably assume that the performing groups were symphonic orchestras, chamber ensembles, and/or choirs. Rock music was probably not among the styles for which the halls were evaluated. I believe it is safe to assume that in each concert, the audience was seated.

While Beranek did not address the frequency dependence of the RT of these halls, presumably because of this factor having been addressed in the literature many times previously, this factor certainly must not be ignored if the acoustics of a hall are to be optimized. **Figure 2** shows curves representing the optimum variation of RT with frequency for a 500,000ft^3 hall to support the most reverberant (choral music) and least reverberant (rock music) music styles, and for lecture use. While this represents an unusually large volume for a school auditorium or civic center, it is not particularly large for a cathedral or a gym.

During my 43-year career (so far) as an acoustician, most auditoria on which I have consulted have been classed as multi-purpose halls: they were designed to serve for all the uses listed in Table 1, plus occasional rock, country music, and folk music concerts, as well as occasional uses for cinematic presentations. Some of these have been "gymnatoria": gyms that are also used as auditoria. Ideally, the RT of a multi-purpose hall should be variable. Using the data from Table 1 plus that from Niels Werner Adelman-Larsen's papers (discussed in the Sound Control article published in the February 2023 issue of *audioXpress*), we can deduce that ideally, the RT of these rooms should be able to be as low as 1 second, ranging as high as 2.5 seconds. Furthermore, the frequency dependence of the RT should be able to be varied from the "rock" curve.

Variable Acoustics

A variety of methods are available for varying the acoustics of a venue; these can be categorized as active or passive methods. Active variable acoustics may begin with an acoustically deadened room in which a highly specialized sound system has been installed to artificially create reverberation whose parameters can be adjusted by the sound system operator. Alternatively, a different approach to the reverberation enhancement system can be used in which the natural reverberation of the venue is picked up by carefully positioned microphones,

Function	Range of RT60	Range of Typical Volume
Lecture, meetings, drama	<1.1s	<140,000ft^3
Multipurpose halls, music, drama, sports	1.25s–1.5s	150,000ft^3–800,000ft^3
Opera, chamber music	1.5s–1.7s	250,000ft^3–700,000
Symphonic music	1.75s–2s	350,000ft^3–900,000ft^3
Organ, oratorio	2.1s–2.5s*	400,000ft^3–1,000,000ft^3
From Table 5A: Preferred Reverberation Times		
*Duke University Chapel in Durham, NC, has an RT60 of almost 7 seconds – suitable for a Gothic Cathedral, but useful only for specific types of music. Adequate speech intelligibility requires exceptional care and a large budget for sound-system design.		

Table 1: Typical ranges of RT and cubic volumes for venues of various uses

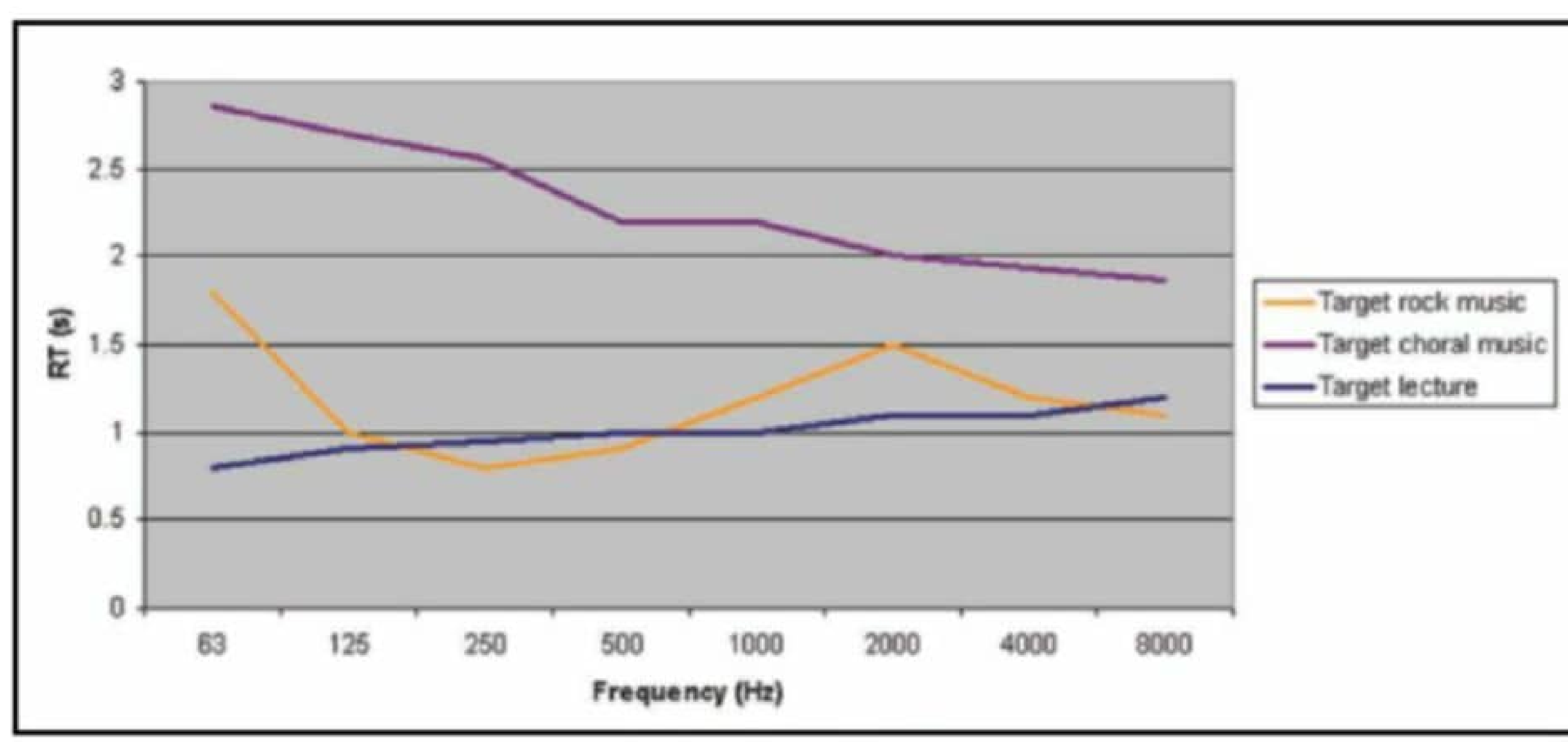


Figure 2: The target RT's variation with frequency depends upon the use of the hall.

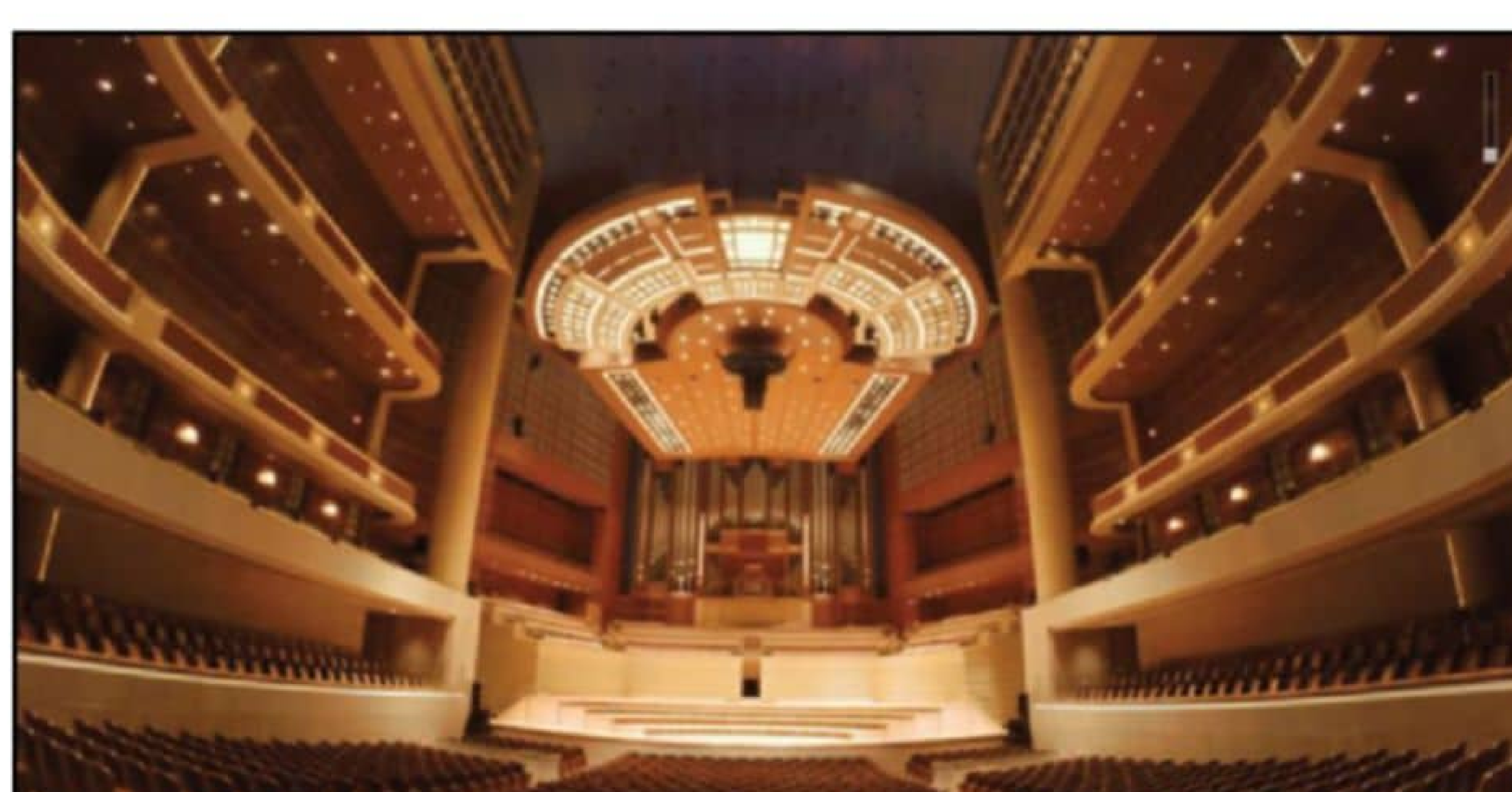


Photo 1: A horseshoe-shaped opening between the upper Myerson Hall and its large reverb chamber can be coupled to the hall, varying the hall's cubic volume.

Photo 2: Mechanically operated doors control the acoustic coupling between the hall's volume and the reverb chamber.

Photo 3: Mechanically operated curtains (look to the left of the stairsteps) covering a variable fraction of the reverb chamber's volume allow frequency voicing of the reverb character.

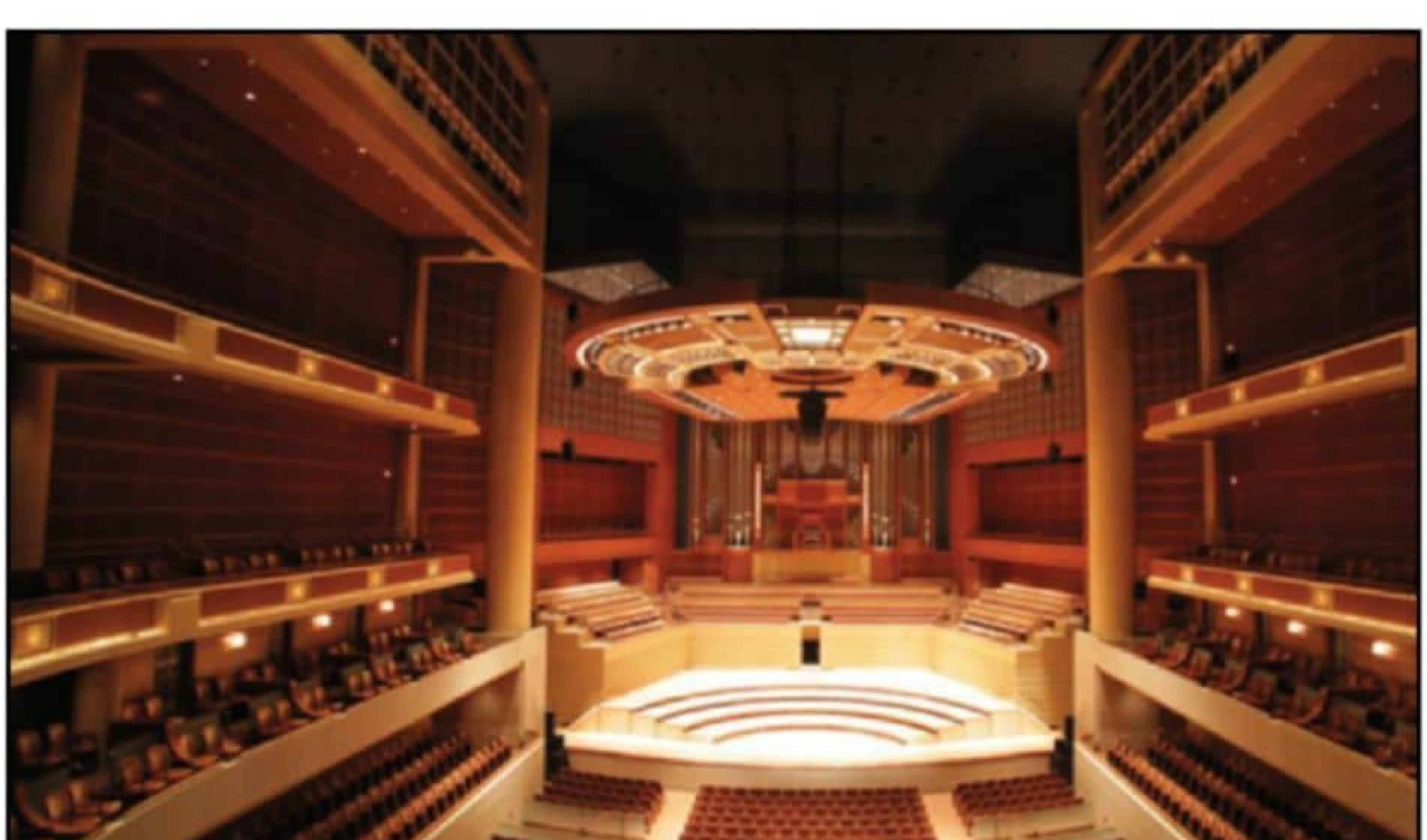


Photo 4: The reflective cloud (or "canopy") can be raised or lowered to control the listeners' perception of room size.

digitally processed, and re-radiated by an extensive, carefully designed speaker system. While many listeners find that the reverb enhancement approach delivers a more natural result, both types of systems have their proponents.

Passive variable acoustics use one of two methods to vary the RT and its frequency dependence. In a room having perfectly diffuse reflections, the RT can be found from the Sabine equation:

$$RT = \frac{0.161V}{\Sigma S\alpha}$$

where V = the volume of the room in cubic meters, and the denominator of the fraction signifies the addition of the products of the area (S) and absorption (α) of each material.

The product Sα is the acoustical absorption, measured in sabins.

So, we see that the RT can be controlled by changing the cubic volume of the room or the total acoustical absorption. In a real room, reflections are virtually never completely diffuse, so the location of the acoustical absorption also affects RT, which varies from seat to seat.

Probably the most successful example of varying the acoustics of a concert hall is the Myerson Hall in Dallas, TX. (See The Sound | Secrets of The Meyerson (kera.org).) This venue incorporates a large reverberation chamber wrapped in a horseshoe shape around the upper portion of the room (**Photo 1**). The opening between the chamber and the hall can be varied by opening or closing mechanically operated doors (**Photo 2**). Part of the chamber can be curtained off to change the frequency character of the reverberation (**Photo 3**). The large, massive acoustically reflecting cloud or canopy above the stage (**Photo 4**) can be raised or lowered in order to change the timing of early sound reflections that listeners hear—effectively varying the Early Decay Time (EDT) of the hall, which controls listener perception of hall size. The overall RT can be adjusted from about 0.3 seconds to over 3 seconds.

The first variable acoustics design accomplished by varying the acoustical absorption was Wallace Sabine's research project in which he investigated the perceptual effect of RT. To achieve this, he and his research assistants moved Oriental rugs, people, and seat cushions between the Sanders Theater and the Fogg Lecture Hall, repeatedly measuring the RT of the venues. Audiences are often the most abundant acoustical absorption in concert or lecture hall, but frictional absorbers such as fiberglass, cotton (felt, in earlier years), or membrane absorbers are often needed as well. A membrane absorber can consist of plastic film, thin wood, or thin gypsum board mounted over a springy substrate such as a thin layer of air, and such an absorber functions by air viscosity—sometimes abetted by friction, when the air space is partially or fully filled with fibrous material—changing acoustical energy into heat. In actual practice, three methods have long been available for varying the acoustical absorption of a room: hinged panels having one absorptive side and one reflective side, absorptive panels partially covered by perforated panels whose holes can be closed by sliding a solid panel over them, and heavy drapes whose wall coverage can be varied by pulling or extending the drapes.

A limitation applies to the use of hinged panels and manually adjustable perforated panels over a frictional absorber in that the maximum absorption that can be added is limited to half of the available wall space. Both methods, as well as operable drapes, are less effective at absorbing low-frequency sound than in the midrange and above.

To compare the performance of different approaches to passive variable acoustics, I built an EASE model of a multi-purpose (MP) hall of 400,000 cubic feet volume, designed for the three types of events for which target RTs are graphed in **Figure 2**. In designing a hall using passive variable acoustics, the hall must exhibit the longest optimum RT required by any event: in this case, choral music. The choice and location of acoustical absorption must be made with a view toward optimizing the frequency dependence of the RT, usually calculated with an occupancy of two-thirds of the maximum audience size. Control of RT variation with changes in occupancy is implemented by proper choices of seating design.

To support choral music, the RT should range from about 2.7 seconds in the 125Hz octave band to 1.87 seconds in the 8kHz band. I was able to approximate this range by using a combination of unpainted masonry blocks, acoustically diffusive/absorptive masonry blocks, fiberglass on the rear wall to prevent echoes from the sound system speakers, 18oz/square yard stage curtains folded to half area, and a reflective ceiling to help project sound from the stage into the audience. **Figure 3** shows the modeled RT vs frequency for this condition.

If we cover all available wall space with hinged panels having the absorptive side toward the audience, our computer model shows how closely

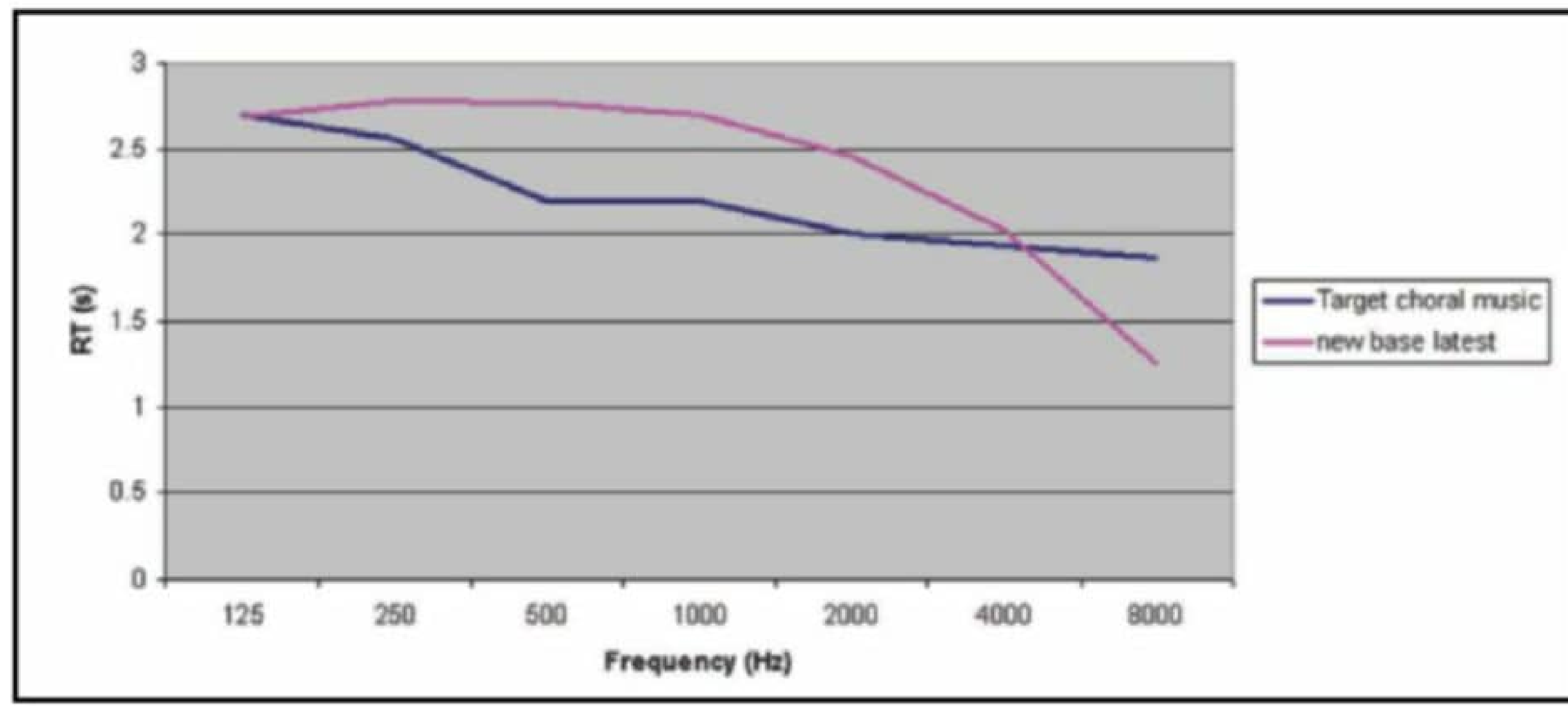


Figure 3: The RT vs. frequency plot for the MP hall approximates the target. (“New base latest” identifies the final interaction of the design for the highest RT condition.)

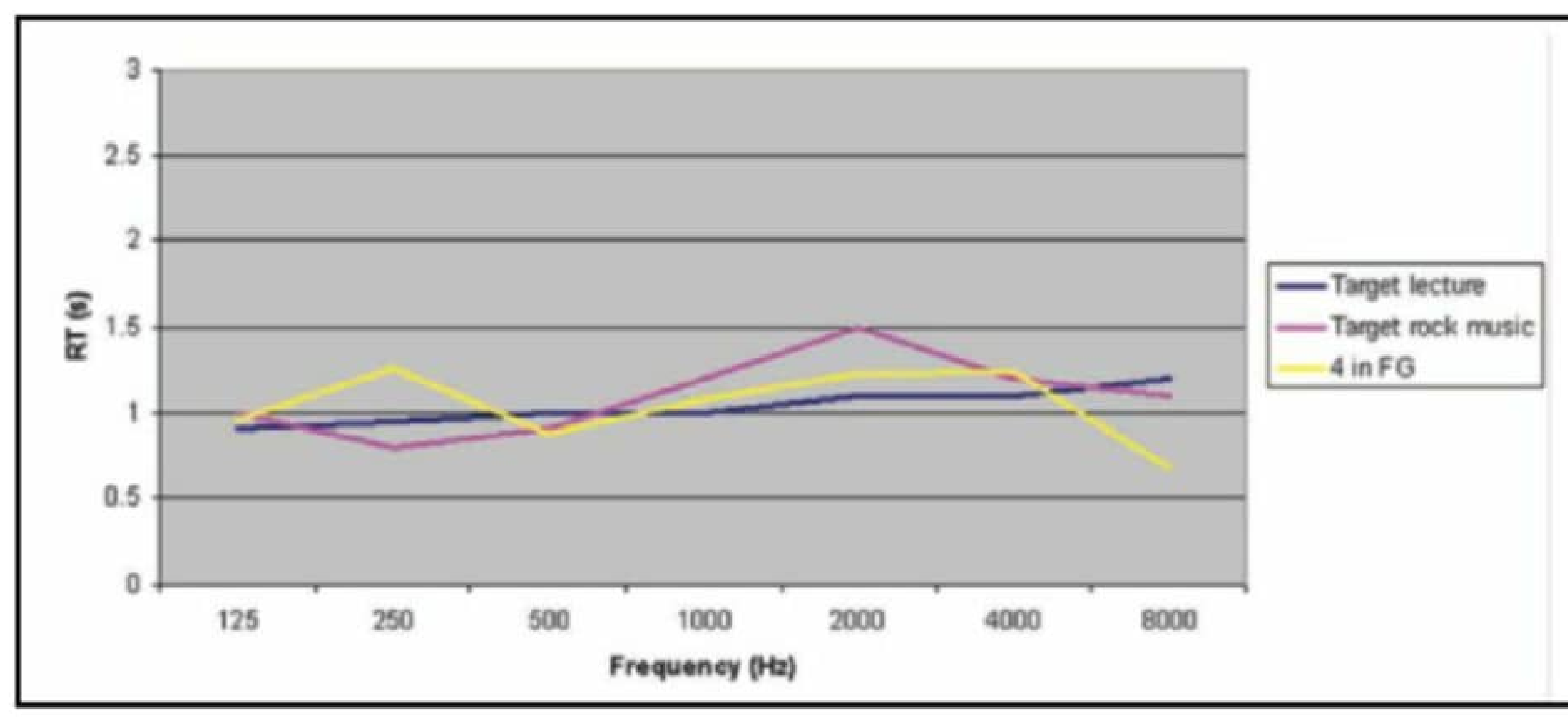


Figure 4: Using hinged panels of 4″ fiberglass over plywood certainly reduces the RT.

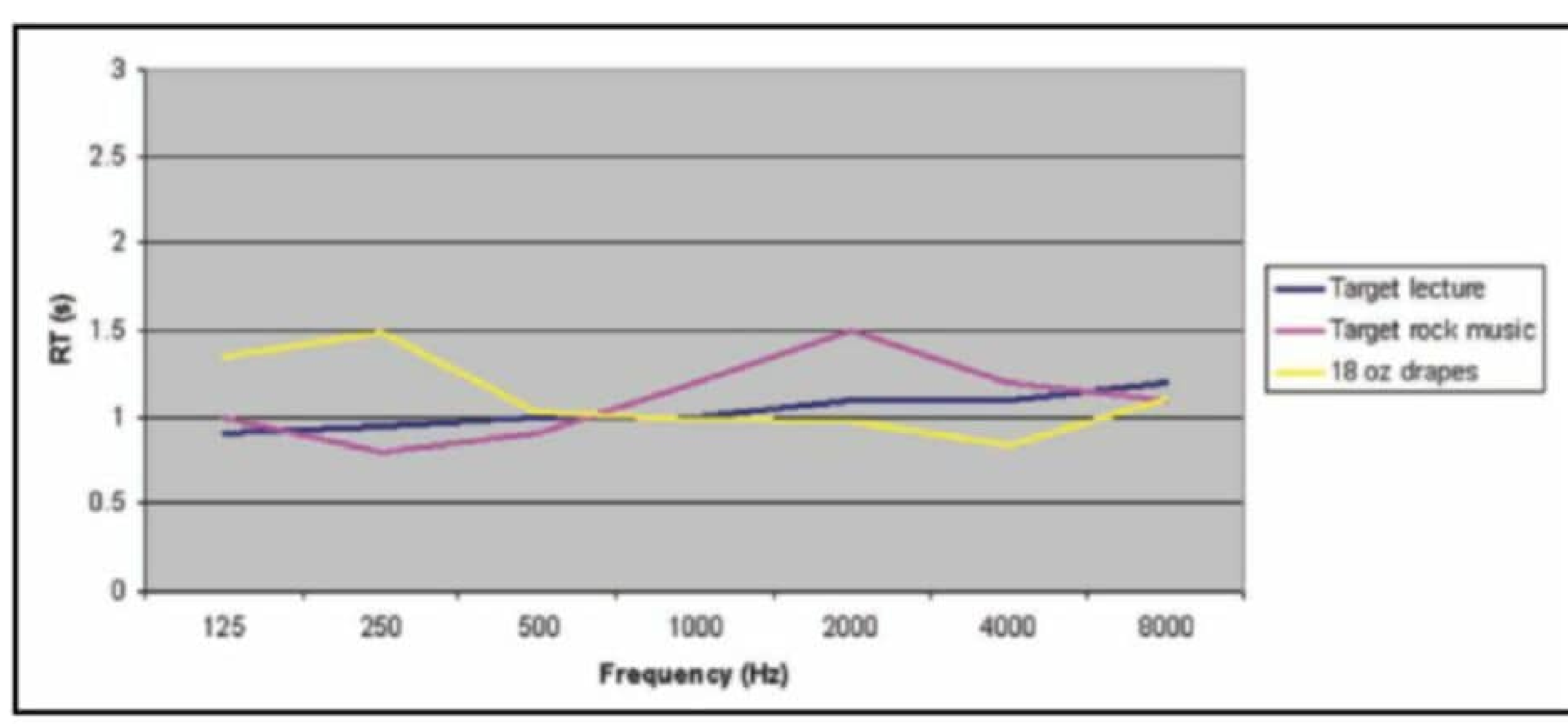


Figure 5: Using heavy drapes on all walls reveals low-frequency absorption issues, particularly in the 125Hz and 250Hz bands.

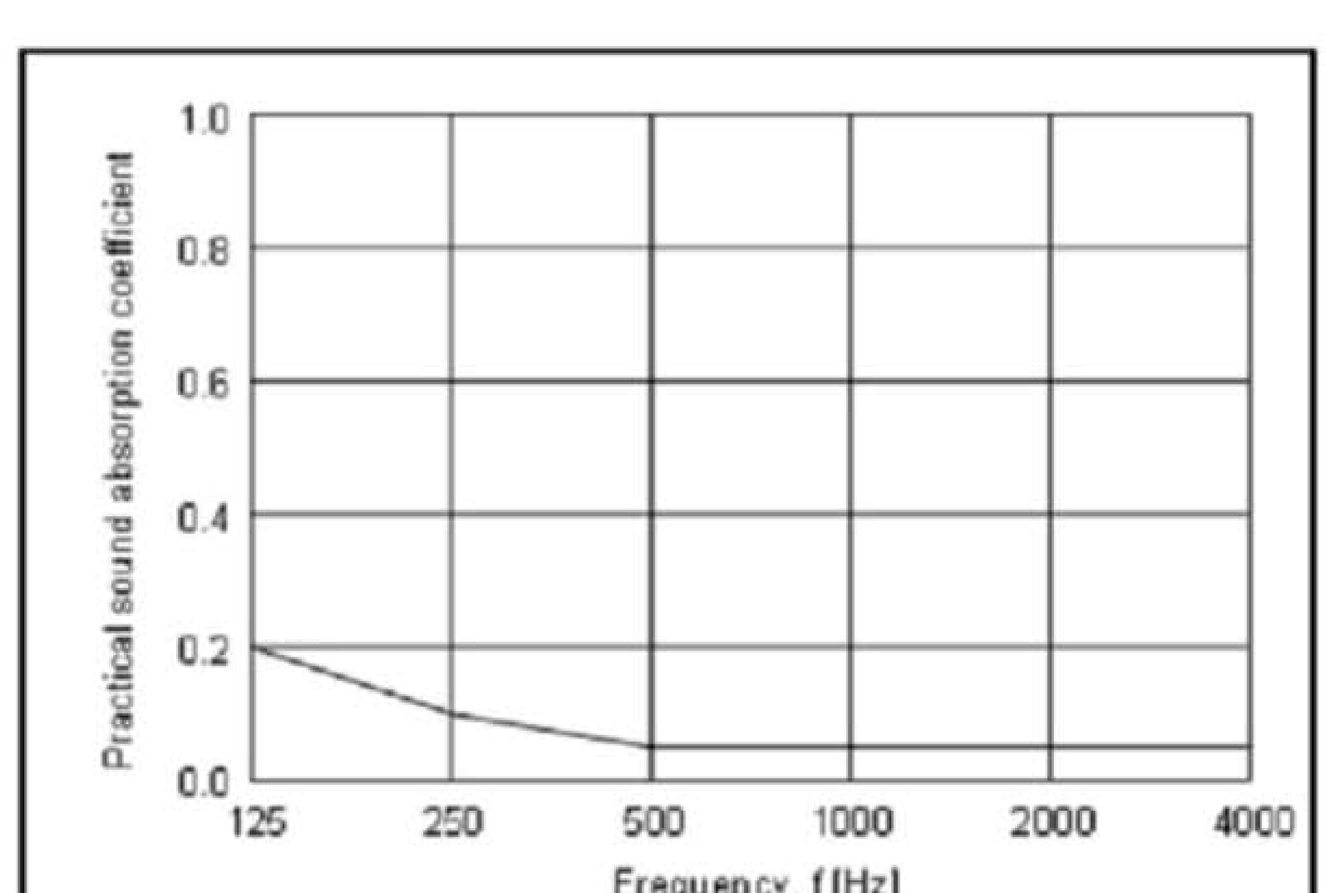


Figure 6: Flex Acoustics’ Evoke panel absorption with perforations closed (minimum absorption condition)

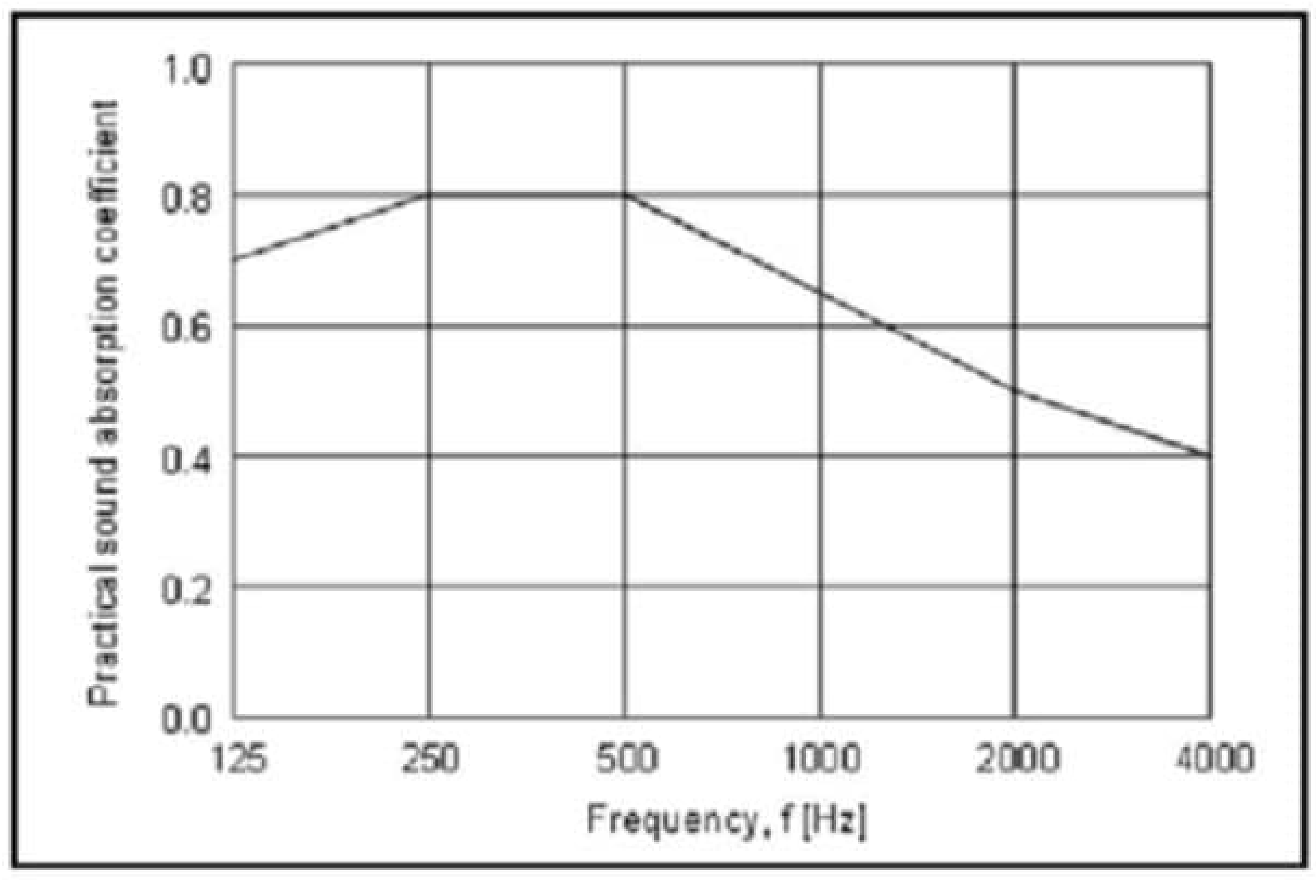


Figure 7: Flex Acoustics' Evoke panel absorption with perforations open (maximum absorption condition)

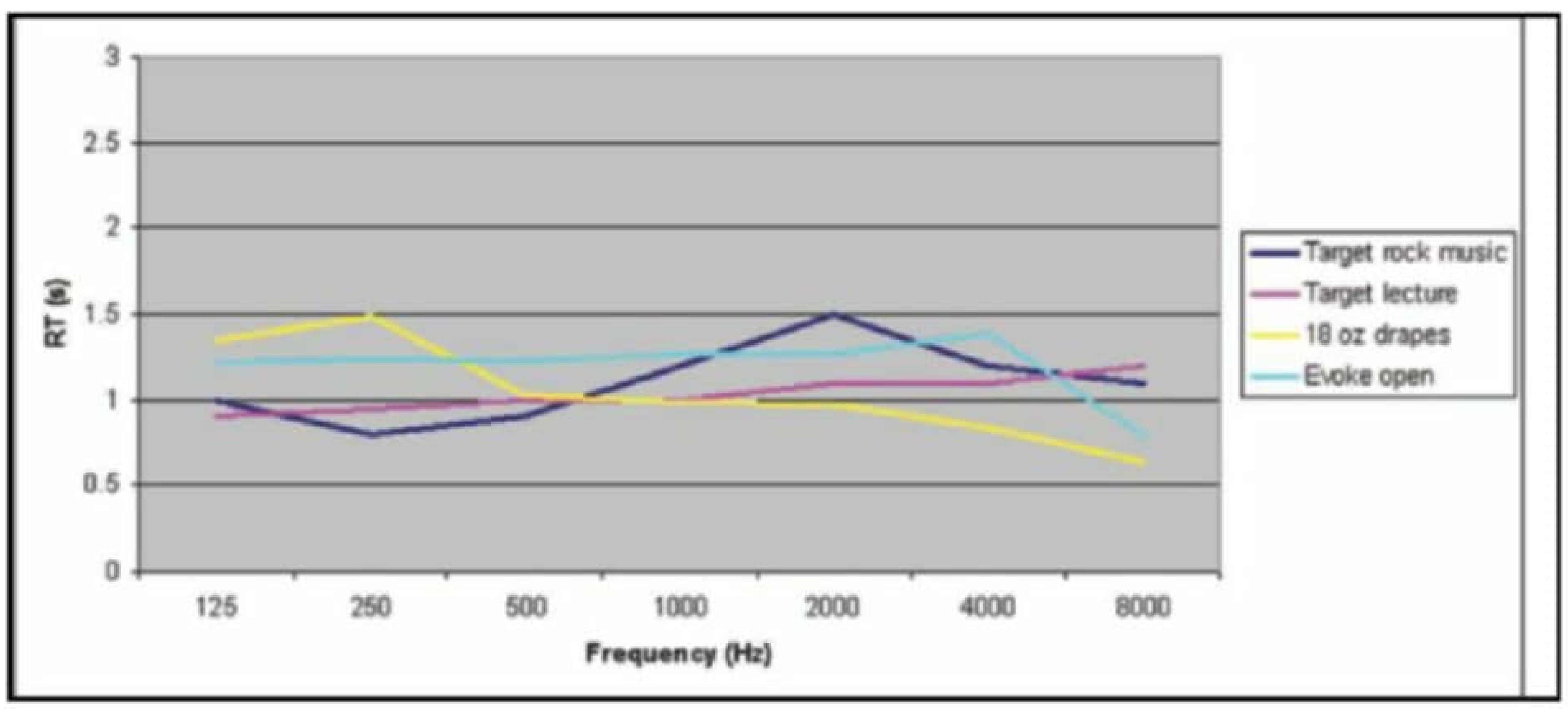


Figure 8: The Evoke system in the open condition closely approaches the target RT profile for rock and lecture events.

About the Author

Dr. Richard Honeycutt fell in love with acoustics after his father brought home a copy of Leo Beranek's landmark text on the subject when Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

References

[1] V. O. Knudsen, *Architectural Acoustics*, Wiley and Sons, New York, 1932.

[2] L. L Beranek, *Acoustics*, McGraw Hill, New York, 1954, p. 425.

[3] R. A. Honeycutt, *Acoustics in Performance*, London, *Elektor*, 2015, p. 28.

[4] L. L. Beranek, "Concert Hall Acoustics: Recent Findings," *Journal of the Acoustical Society of America*, (JASA) B139, 4 (April 2016), pp. 1548-1556.

[5] N. Adelson, E. Thompson, and A. Gade, "Suitable Reverberation Times for Rock and Pop Music," *Journal of the Acoustic Society of America* (JASA), 127 (1) January 2010.

Resources

Evoke, "Installation in Sound Labs," Flex Acoustics,
https://flexac.com/wp-content/uploads/2022/08/Evoke-CASES-SoundLab-Flyer-arrow-kopi.pdf

Evoke, "Installation in Concert Hall, Flex Acoustics
https://flexac.com/wp-content/uploads/2022/06/EvokeAarhus-Musikhus-qrcode.pdf

we can approach the target low-RT targets (lecture and rock concerts), as illustrated in **Figure 4**. Although the hinged panels reduce the RT, they do not really allow us to match the target RT vs. frequency characteristic for either lectures or rock concerts. An additional disadvantage of this approach to passive variable acoustics is that changing the acoustics of the hall is not really feasible between selections in a program.

Many "black-box theaters," which require lecture-type RT profiles, use heavy drapes on all walls to vary the acoustics. A computer model of our MP hall showing the results of this approach is shown in **Figure 5**.

As you can see, the limited low-frequency absorption of the drapes results in longer than optimal RTs for lectures or rock concerts. Drapes, however, can be repositioned to modify the acoustics during an event, particularly if motorized drapes are employed.

Flex Acoustics, using a design developed by Niels W. Adelman-Larsen [5], has developed the Evoke system comprised of variable acoustic modules, each based upon two layers of acoustic linen separated by 90mm of mineral wool. The acoustical performance of each module can be controlled by motors that open or close perforations in the panel's housing.

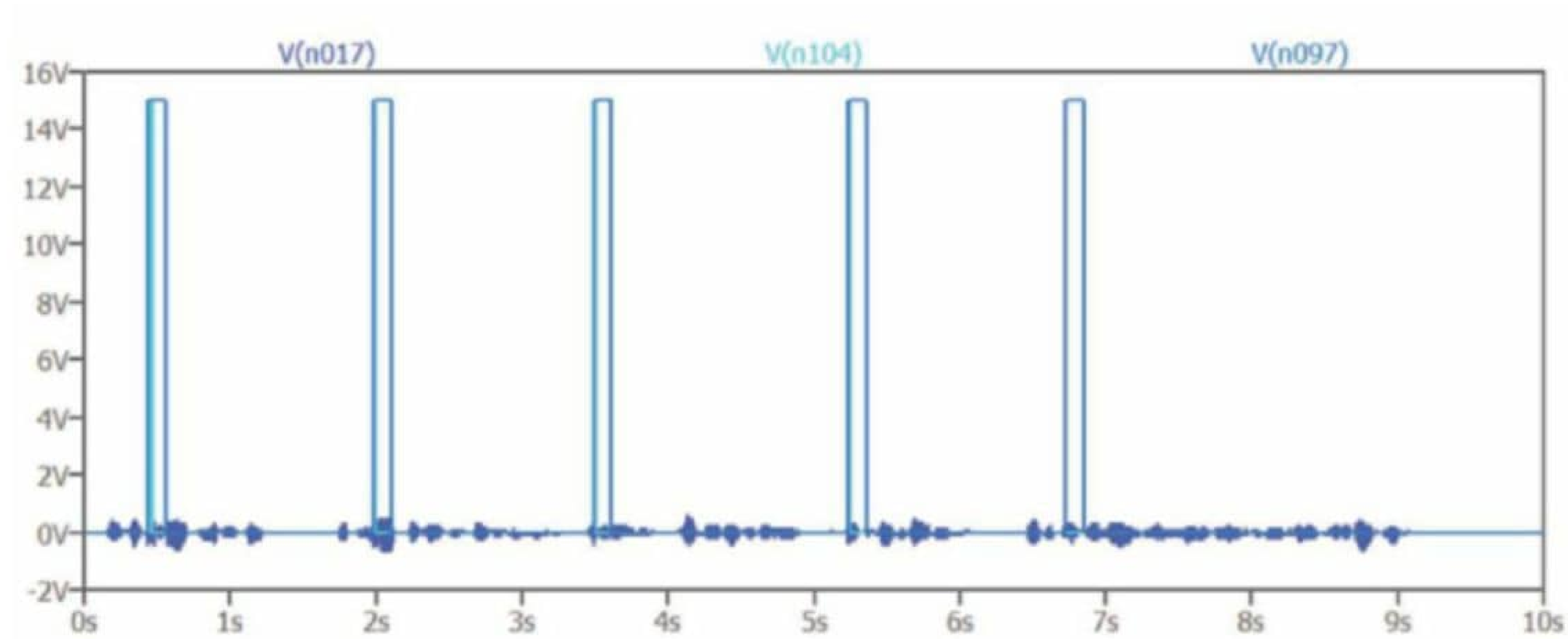
Figure 6 and **Figure 7** show the acoustical absorption coefficients by octave band for Evoke panels in the closed and open conditions, respectively. **Figure 8** shows our MP room with Evoke modules covering all walls, and the audience standing, as is typical for a rock concert. In fact, the room could be further deadened" using these modules, because the ceiling could be covered with modules—a condition not usually desirable for acoustic music concerts, drama, or lectures. Reflected sound from the ceiling is an important part of the acoustical experience for such events, though not for rock concerts.

Evoke modules provide a solution to passive variable acoustics that can easily be changed to support different types of program material during an event. The system consists of individual 240cm × 60cm modules, each with its own low-voltage motor. The modules can be set up to be individually controlled, or they can be controlled in groups.

Since virtually no real room is truly acoustically diffuse, the locations of various open and closed modules can affect the profile of RT vs frequency, allowing additional control of RT characteristics. Further information can be found from Flex Acoustics (see Resources).

Build the Popping P Filter

Popping Ps are low-frequency impulses caused by air blasts from a person speaking or singing into a microphone. This problem can also occur with other "plosive" letters such as B, T, K, and D. In this article, Ethan Winer shows an analog circuit that removes these irritating sounds from a live audio stream in real time.



By **Ethan Winer**

The best way to avoid popping Ps and Bs is at the source. Many people use a mesh screen filter in front of the microphone, but that's not really necessary. Popping Ps are caused by air blasts from the performer's mouth, so simply placing the microphone a few inches higher pointing down, or angled off to one side, will avoid this problem. But audio engineers often work after the fact and aren't always able to control mic position. I hear popping Ps all the time on the radio and TV, especially from live on-the-scene reporters when the microphone is too close to their mouth.

What Is a Popping P?

In a recording or podcast studio that pre-records content, popping Ps can be removed in an audio editing program. Software is available that can detect and remove popping Ps automatically, or you can do it manually. As shown in **Figure 1**, the basic method is to highlight just the pop sound, then either drop the volume about 15dB or apply a high-pass filter. This figure shows the word "Paris" three times in a row: First as the word exists in the file, then again with just the pop sound highlighted, and then after reducing the highlighted area's volume by 15dB. When the file is played, the third (corrected) version now sounds like a normal P.

There's no commercial device I know of that can remove popping Ps from a live audio stream. The circuit I developed and describe in this article can be used in recording or broadcast studios to repair the audio passing through their mixing board. As it turns out, removing popping Ps in real time is very difficult! The main problem is identifying the pop before it happens, so it can be fixed before even the first bit comes through. The low-pass filter that's part of the detection circuit has 2mS of group delay below 300Hz. So, letting even that much of the pop sound through is unacceptable. Without this filter, loud normal speech might trigger the pop suppression. Therefore, the main audio stream needs to be delayed slightly so the pop detector can identify and suppress the pop starting just before it happens.

I used a 24-stage all-pass filter to delay only the lower frequencies of the main audio stream, but not the mids and highs. Creating the delay digitally would avoid having so many filter stages, but that raises several issues. With a digital delay, the mids and highs would also lag. If the person speaking also monitors through the device, hearing their voice delayed even a few milliseconds could be a problem. And if a crossover was used to delay only the lows, combining the delayed lows with the not-delayed mids and highs later could make a real mess of the response. As you may know, all-pass filters shift the phase delaying select frequencies, but they don't change the response. Further, there's no practical way I know of to include digital delay in the LTspice simulation program I used to develop this circuit.

I've described LTspice in several of my previous *audioXpress* articles, so I won't belabor its many features here. A link to the download page for this terrific free program is listed under Resources, along with links to my previous articles that explain more about this program's features.

Circuit Overview

Figure 2 shows a block diagram of the circuit's operation. I think it makes sense to describe the circuit's basic operation here, then later I'll discuss specific circuit details in the schematic. The input buffer presents a high input impedance to the audio source and isolates the source from affecting the low-pass and all-pass filters that follow. The all-pass filters delay frequencies below 300Hz to 400Hz by around two cycles, though it's difficult to determine the exact amount of delay. Cumulative overlap between stages will increase the delay at some frequencies, and a given P pop might not align exactly with any stage's center frequency yielding slightly less delay. The voltage-controlled high-pass filter does the real work of removing popping Ps by suppressing frequencies below 360Hz rolling off at 12dB per octave.

Detecting P pops isn't too difficult. The low-pass filter helps to isolate the pop sound, then a full-wave rectifier converts that to only positive voltages. This way pops will be detected no matter which polarity dominates. Finally, a voltage comparator is set to trigger only when the filtered audio exceeds a given level indicating a popping P. It's not shown here, but the schematic includes a "Pop Sensitivity" control to adjust for different signal and popping P levels.

The last stages create the control voltage that reduces low frequencies from the main audio stream. When the comparator's output switches from 0V to the 15V supply it triggers two timers. The first timer starts the second one, holding its own output high for 125ms so the second timer can't quickly re-trigger. Some popping Ps contain a single cycle that comes and goes quickly, but others will flip the comparator several times, even beyond the 50ms low-frequency mute time. So, the first timer locks out repeated triggering by the comparator for 1/8th second.

Finally, the peak follower is needed to allow ramping down the control voltage that drives the transistor mute switches. Switching the transistors on quickly to mute low frequencies isn't a problem because the bulk of the audio—the pop—is being muted. But when the filter switches off letting the main audio through, a sudden DC level shift on that main audio would create a nasty pop of its own! You can see this level shift by temporarily removing capacitor C7, then probing the output of op-amp U8.

All-Pass Filters

Before we get to the complete schematic, let's first consider how an all-pass filter works. In **Figure 3**, you can see that both the plus and minus inputs of an op-amp are driven at equal levels. The formula for the filter's center frequency Fc is also shown. Frequencies well above Fc are shunted to ground through C1, so little signal gets to the plus input and instead goes through the op-amp's inverting path at unity gain. Frequencies well below Fc are unaffected by C1, so they go through both the inverting and non-inverting paths. But the non-

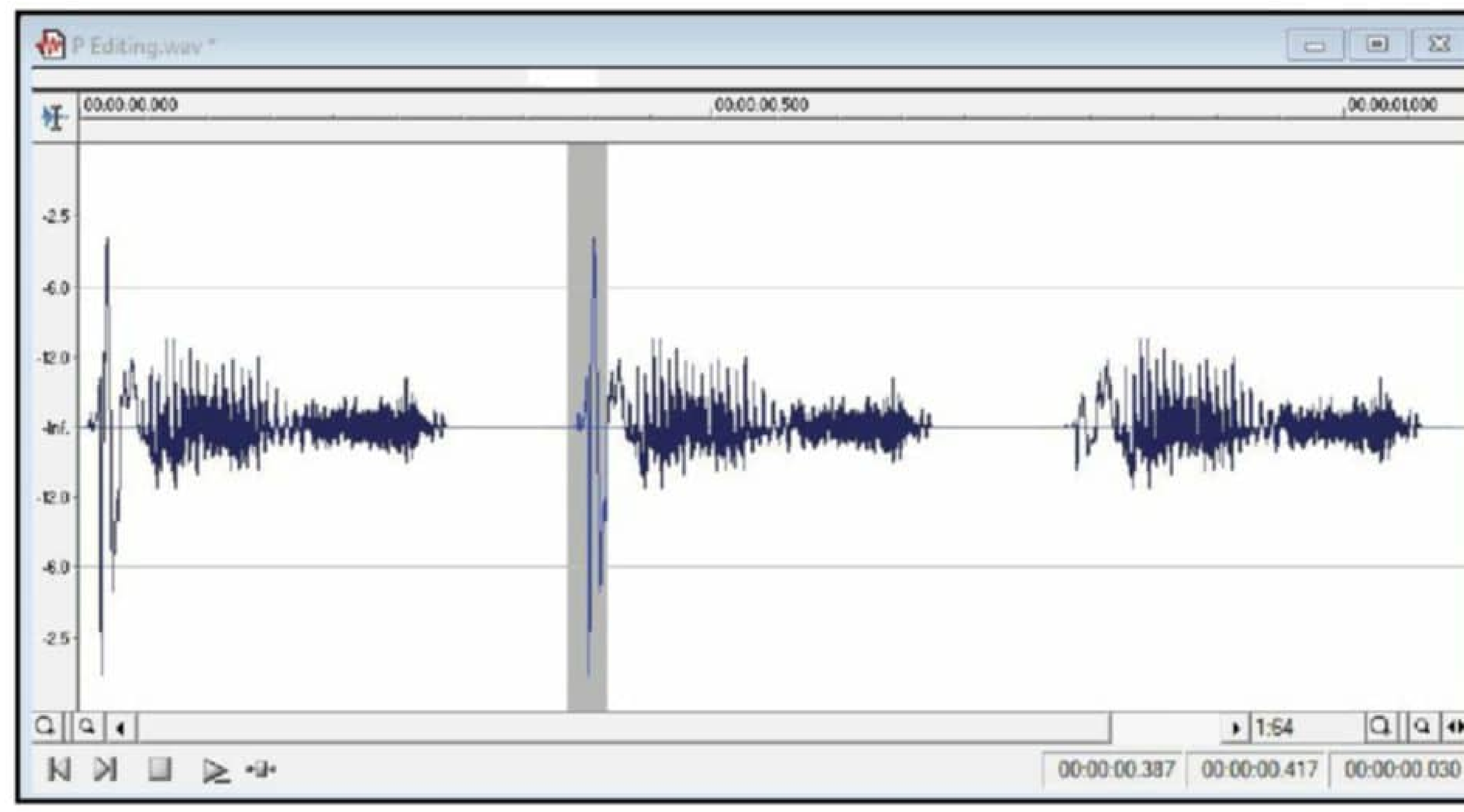


Figure 1: Popping Ps are easy to identify in an audio file by their large amplitude. The wide spacing between the initial wave cycles is another clue to the presence of low frequencies.

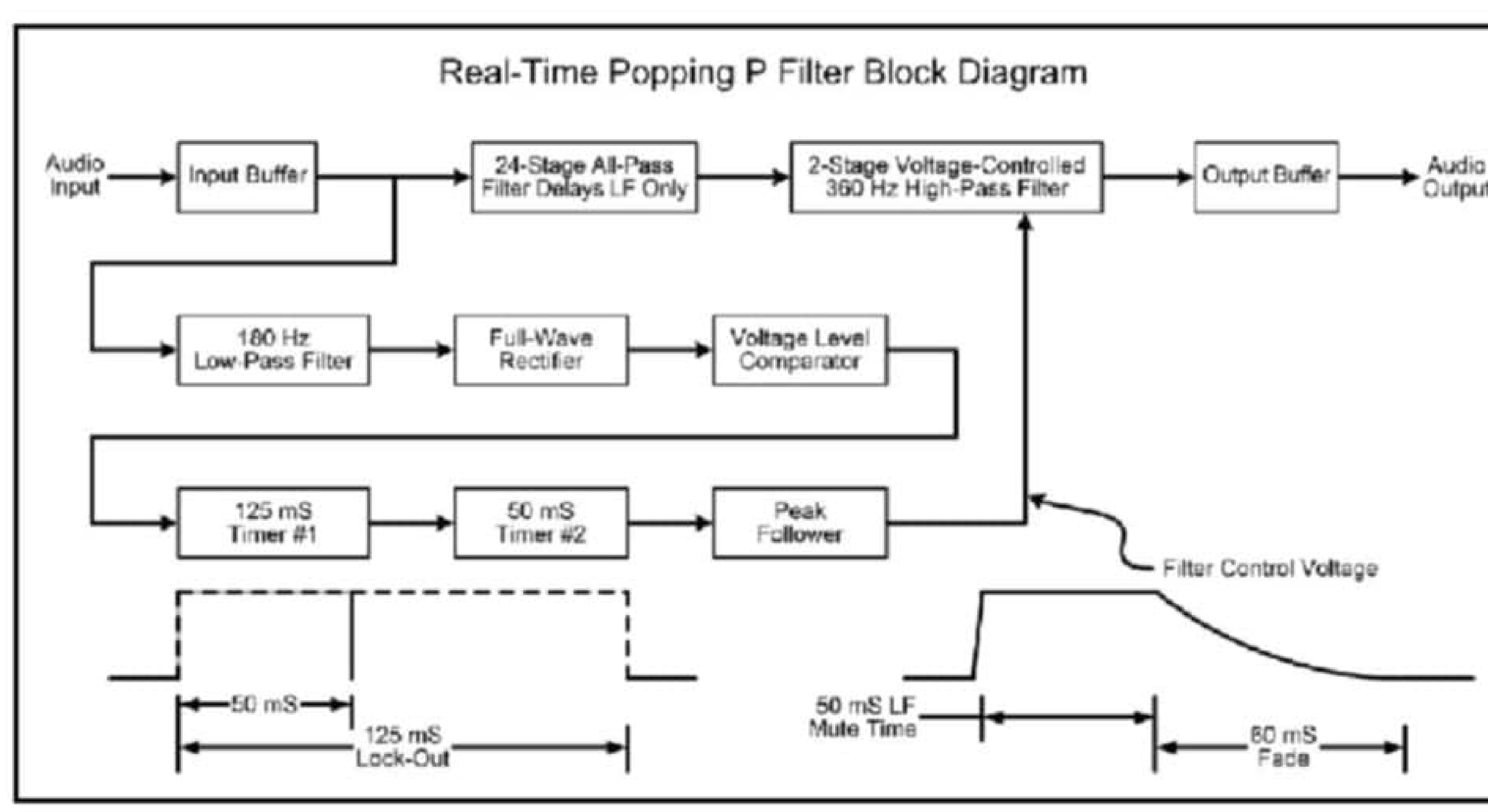


Figure 2: This block diagram shows the circuit's general operation, with three separate filters used to detect popping Ps, delay only low frequencies from the audio stream, and reduce the level of those low frequencies.

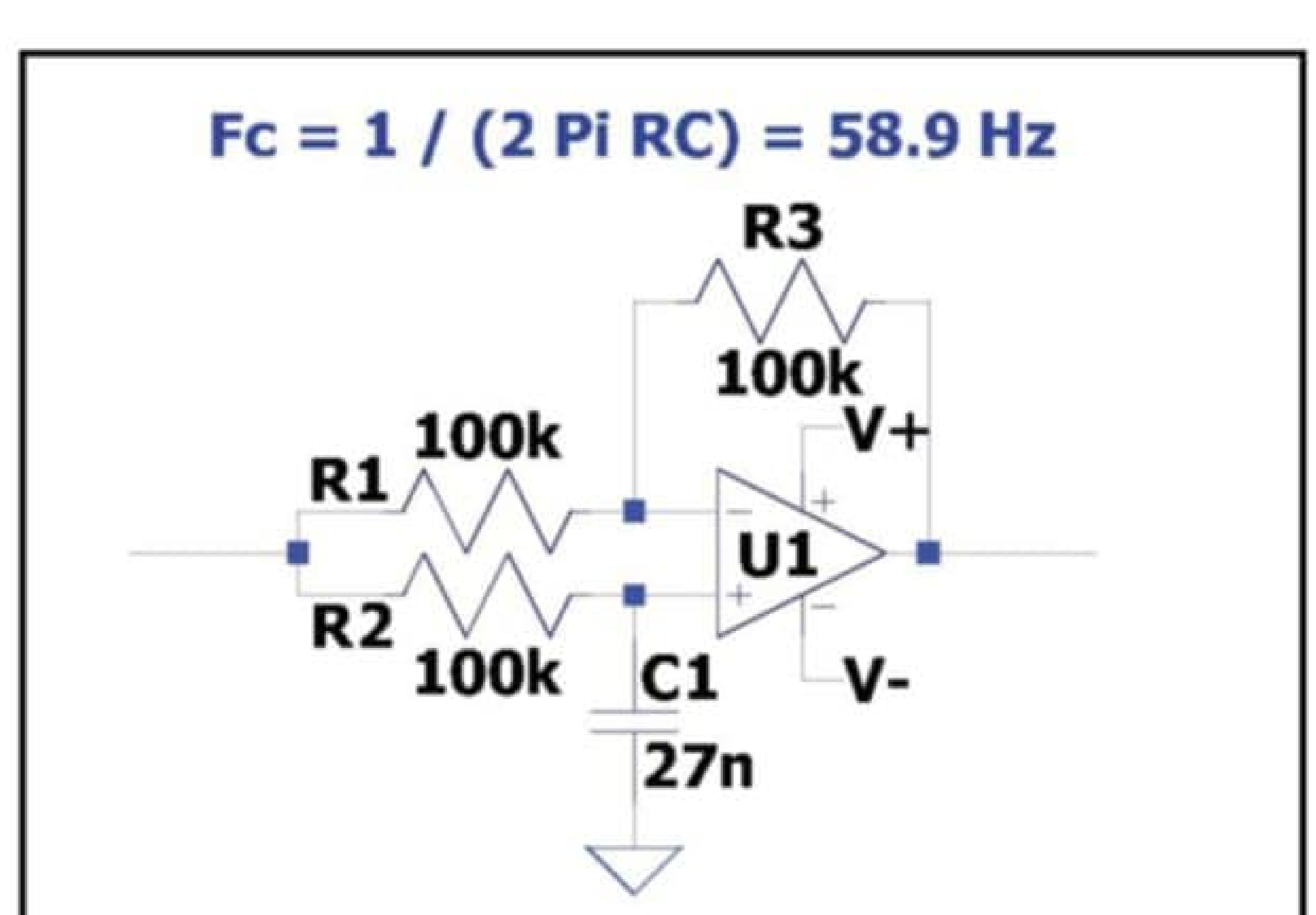


Figure 3: An all-pass filter sends the input signal to both the inverting and non-inverting inputs of an op-amp.

inverting path has a gain of 2 as set by R1 and R3, where the inverting path is unity gain. So the non-inverting path "wins" and also exits the op-amp at unity gain.

The "magic" happens for frequencies at or near Fc, because C1 takes time to charge through R2 creating phase shift that manifests as a time delay. All-pass filters are at the heart of musical instrument "phaser" effects to create the swooshing sound of comb filtering as the filter frequencies sweep up and down. Understand that this sound is not the phase shift itself! Even in fairly large amounts phase shift is inaudible. Rather, the hollow sound occurs

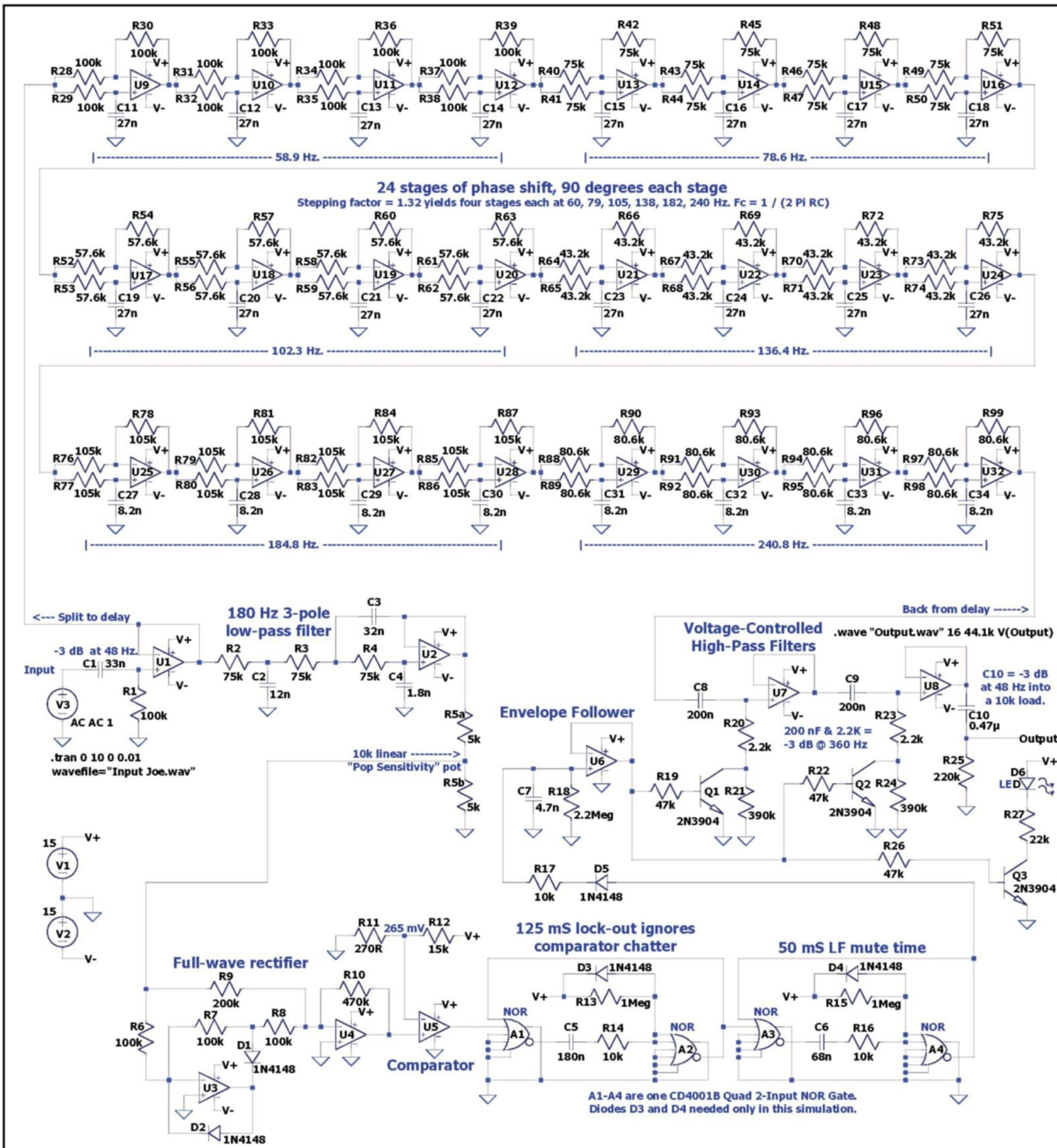


Figure 4: This is the complete schematic for the Popping P filter. (Copyright © 2023 by Ethan Winer. All rights reserved.)

when the source and phase-shifted outputs are combined. This creates a series of peaks and deep nulls in the response, and that is what you hear.

Figure 3 shows a single filter stage, though all-pass filters are typically used in pairs to avoid reversing the polarity of content above Fc. That is, two passes through an inverting op-amp restore the original polarity. Just as important, one stage can shift the phase no more than 90°, so two stages are needed to create a complete comb filter null in the response when used for a phaser effect. Most phaser effects use either six or eight stages, which creates either three or four peak-null pairs. You can optionally combine the source and phase-shifted outputs in opposite polarities to synthesize stereo from a mono source. That's the basis of my DIY stereo synthesizer article from the June 1979 issue of *Recording-engineer/producer* magazine linked under Resources.

Details, Details

And now we get to the complete schematic shown in **Figure 4**. I formatted the schematic to fit nicely on a single magazine page (Figure 4 has also been included in the Supplementary Materials section located on the *audioXpress* website). As you read the descriptions below, use the component numbers R1, C2, U3, and so forth to follow the logical progression of the signal path. The top three rows are the all-pass filter stages, with the input just below on the left. We'll start with the input rather than at the top of the page.

Input voltage source V3 is set inside LTspice for a function of None rather than the usual sine or pulse wave. The wave file Spice directive tells V3 to take its input from the file Input Joe.wav. There's no reason to allow DC or subsonic content into the circuit, so C1 and R1 form a simple high-pass filter at 48Hz. As long as the source feeding this device has a low output impedance, which is typical, those filter component values should hold. A higher output impedance would do no harm, merely lowering the cut-off frequency a little. The output of U1 sends the wave file input to the all-pass filters above, and to a steep 180Hz low-pass filter as part of the pop detection.

LTspice doesn't have a native potentiometer component, so resistors R5a and R5b substitute for the Sensitivity control. LTspice treats audio from a Wave file at 0dBFS as an input level of ±1V, or 2Vpp. Combined as a potentiometer, the R5 pair allows for signals half that amplitude when fully clockwise. If you expect to work with smaller input levels you can raise R10 or R12, or better, change U1 from a unity-gain follower to whatever amount of gain is needed. More gain at U1 puts the input signal that much higher above the all-pass filter's noise floor. Another useful modification would be to configure the input to be fully balanced.

The full-wave rectifier in U3 and U4 is a standard circuit, as is comparator U5. When a pop larger than 265mV gets through the low-pass filter, the comparator's output flips from 0V to 15V. This triggers the one-shot timer comprising NOR gates A1 and A2 to go high, which also starts the second timer. As mentioned, a loud pop might continue to trigger the comparator for longer than the 50ms low-frequency mute period. So, the first timer prevents any false triggers from muting the audio. Note that op-amps U4 through U6 connect their negative power feed to ground rather than to V-. These outputs never need to go below ground, and comparator U5 drives a logic chip whose input shouldn't be allowed to go below ground.

One advantage of the NOR-based timers used here is that they can't be re-triggered during their active timing period. So even though comparator U5 flips back and forth several times while a popping P is present, the timer's output always remains high for 125ms. Diodes D3 and D4 prevent the reverse spikes from C5 and C6 from exceeding V+ by more than 0.6V. When A1's output initially goes from V+ to ground, capacitor C5 follows. But 125ms later A1 goes high again, which sends C5 above 22V possibly damaging the input to A2. That spike also affects the timing for subsequent popping Ps because C5 starts its initial descent from 22V rather than from V+ which is 15V. However, the CD4001B NOR gates have protection diodes built in, so D3 and D4 are needed only for this simulation. At only 1% of the 1MΩ timing resistors R13 and R15, the 10kΩ current-limiting "safety" resistors R14 and R16 are too small to worry about.

There are thousands of logic ICs available, so rather than provide all those component models LTspice offers generic versions that are highly flexible. The AND and OR gates each have five inputs, with both normal 1 and reversed 0 "Not" outputs. This way OR gates can also serve as the NOR (Not OR) gates used here. In Figure 4, the outputs with the little circles are the NOR "Not"

About the Author

Ethan Winer has been an audio engineer and professional musician for more than 50 years. His Cello Rondo music video has received nearly 2 million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press, now in its second edition, is available at www.amazon.com and on Ethan's own website www.ethanwiner.com. Ethan is also a principal at RealTraps, a US manufacturer of high-quality acoustic treatment.

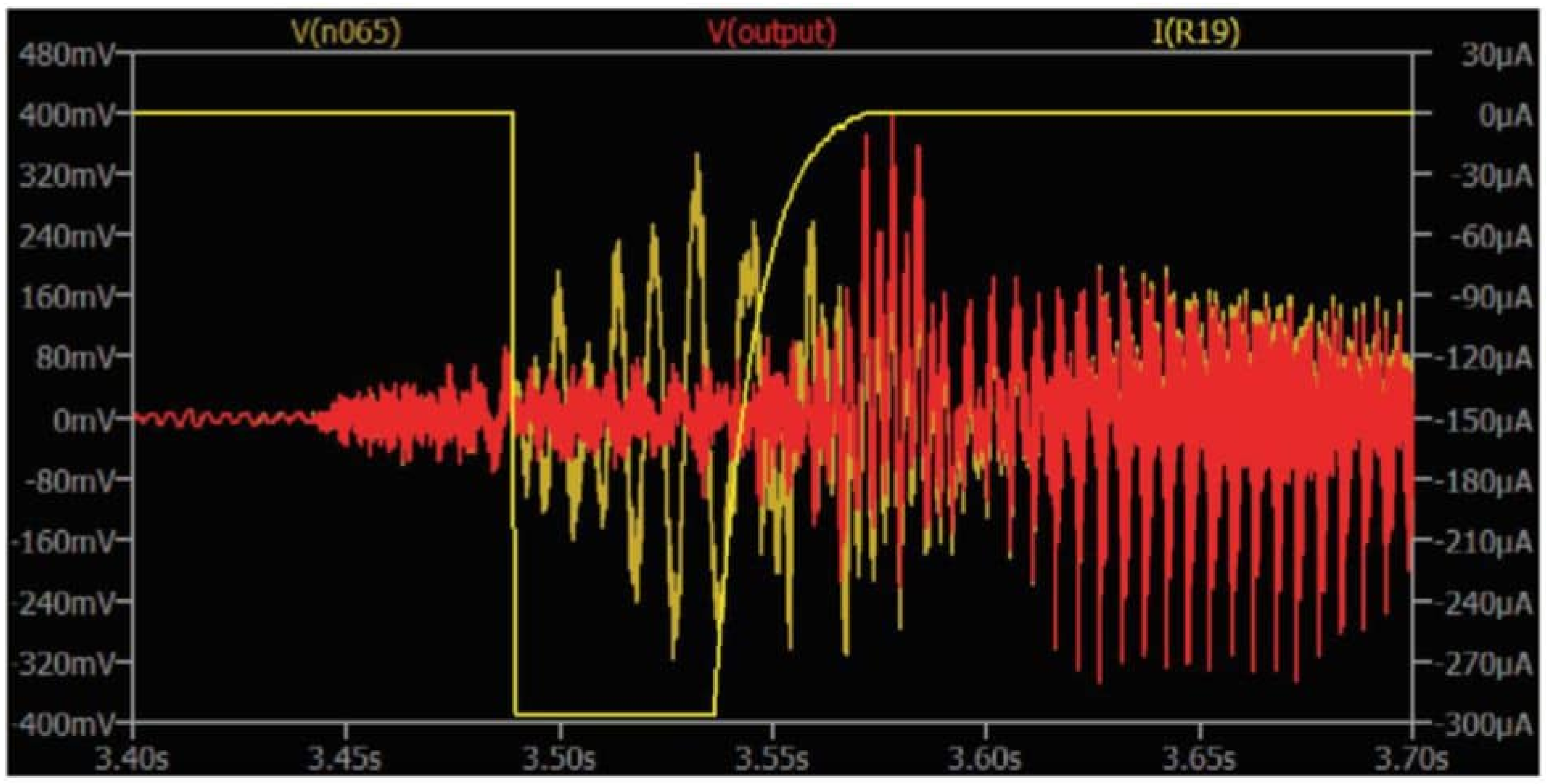


Figure 5: This oscilloscope view shows the circuit in operation as it filters a popping P.

output connections, and the OR outputs are grounded to show they're unused. With LTspice digital components, unused inputs or outputs should be grounded. This tells LTspice not to waste time computing those connections. Of course, in a real circuit you'd never connect an unused output to ground!

Op-amp U6 serves as an envelope follower, though there's no real envelope to follow. You can see in Figure 2 how the positive output from NOR gate A4 is "sculpted" to mute the low frequencies quickly through R17, then recover back to a normal response at a slower rate through R18 to avoid creating its own popping sound. Diode D5 is also needed to avoid pulling the control voltage back down when the 50ms timer completes. U7 and U8 are simple high-pass filters, one pole each, tuned to reduce frequencies below 360Hz. When a popping P is detected, transistors Q1 and Q2 pull the 2.2kΩ resistors down to ground which enables the filtering through C8 and C9. Q3 is also triggered to light an LED showing that a pop was detected. Seeing when the pop suppression occurs can help when setting the Sensitivity control. I'll also mention that if a source voice is very bassy and requires EQ, it's better to reduce the level of low frequencies before sending the audio through the Popping P device rather than after.

The LTspice file you can download under Project Files is set for oscilloscope view via the .tran directive. I encourage you to probe the various input and output points to better see how this circuit works. As one example, the brown trace in **Figure 5** is the delayed audio as it exits the all-pass filter bank. The device's final output is shown in red, and the yellow trace shows the current through R19 into the base of Q1. The brown input within the 50ms filter window shows several cycles of low frequency content at a large amplitude—the popping P. But that's mostly missing from the red output signal.

I set this image to show the current through R19 rather than the voltage output from the envelope follower because the follower's output is much larger than the audio. This view keeps the vertical range to under half a volt making the audio large enough to see its details. When possible, displaying

Project Files

To download additional material and files, including Figure 4, the LTspice file and demo input wave files, visit: https://audioxpress.com/page/audioXpress-Supplementary-Material.html

Project Files for this article include:

Figure 4, Input Ethan.wav, Input Joe.wav, Input Rob.wav, Popping.asc, and Popping Demo.mp3

Resources

LTspice, www.analog.com/en/design-center/design-tools-and-calculators/ltspice-simulator.html

E. Winer, "Build a Stereo Synthesizer," *Recording-engineer/producer* magazine, June 1979, http://ethanwiner.com/St-Synth.html

Previous articles by Ethan Winer that describe LTspice features:

E. Winer, "Building a Guitar-Controlled Synthesizer: Part 1 — The Sample & Hold Time Machine," *audioXpress*, April 2022, https://audioxpress.com/article/you-can-diy-building-a-guitar-controlled-synthesizer-the-sample-hold-time-machine

E. Winer, "Building a Guitar-Controlled Synthesizer: Part 3 — LFO and ADSR," *audioXpress*, June 2022, https://audioxpress.com/article/you-can-diy-building-a-guitar-controlled-synthesizer-lfo-and-adsr

E. Winer, "Building a Guitar-Controlled Synthesizer: Part 4 — Input Section," *audioXpress*, July 2022, https://audioxpress.com/article/you-can-diy-building-a-guitar-controlled-synthesizer-input-section-pick-detector

E. Winer, "Building a Guitar-Controlled Synthesizer: Part 5 — VCA and VCF," *audioXpress*, August 2022, https://audioxpress.com/article/you-can-diy-building-a-guitar-controlled-synthesizer-vca-vcf

current rather than voltage can keep everything on screen clearly when monitoring both small and large voltages at the same time.

Because popping Ps can span a wide range of low frequencies, as high as 180Hz or possibly higher, I used six groups of all-pass filters having four stages of phase shift each. The correct way to divide the filter frequency range of 60Hz to 240Hz is logarithmic rather than linear. If the filters were spaced at equal 36Hz intervals they'd be too far apart at the low end and too close together at the top. I applied a "scaling factor" of 1.32 to determine each successive frequency. That is 1.32 times 60 is about 79, then times 1.32 again is about 105. This is similar to multiplying a musical note frequency by 1.0595 to get the frequency of the next note a half step higher. The comment in Figure 4 under the top row of filters shows the ideal target frequencies I calculated, and the numbers under each group show the actual frequencies based on the closest available 1 percent standard resistor values.

Note that I haven't actually built this device. So far it exists only as an LTspice file. But the circuit is very straight-forward, so there's no reason it won't work the same when built. The only potential problem I'm aware of is the NOR gate timer periods could vary if the CD4001B inputs don't transition at half-supply like LTspice gates do. If you build this, probe the input and output of both timers to confirm their active time spans. You can change either the capacitors or the 1MΩ resistors slightly if needed.

Demo Audio Files

Listed under Project Files are three sample wave files you can run through the circuit inside LTspice. Input Ethan is a file I created with the most exaggerated P pops I could muster. Input Joe was captured from a recent broadcast stream, and Input Rob is from a friend's comedy show I videoed years ago. I also included a short MP3 demo showing Input Joe before and after removing the popping Ps, so readers can hear how well the circuit works without needing the LTspice program.

Author's Note: This circuit is copyrighted but not patented. It's offered for free for non-commercial use, and I welcome licensing offers for commercial manufacture. Contact me through my website: http://ethanwiner.com

The Bitches Brew Open Baffle Live Edge Speakers

Highly praised at the 2022 Speaker Design Competition sponsored by Parts Express, these large dipole speakers named after the classic Miles Davis 1970 album have surprised for the stunning imaging and 120dB dynamic range. The article from the author describes the full three-way system construction made from book matched slabs of spalted sycamore with a natural finish.

By
Perry Marshall

Photo 1: This system is the "ultimate design," which also aims for simplicity and elegance—a 15" three-way system with two SB Audience 15OB350 woofers and one B&C 15CXN88 constant directivity horn coaxial.

Photo 2: The speaker boxes are made from book matched slabs of spalted sycamore.

I've been designing speakers for 40 years as a hobbyist, including a three-year stint at Jensen's automotive division, where I designed OEM drivers professionally for Honda, Mazda, Chrysler, and Acura. I've built almost every type of speaker and enclosure type that exists except electrostatics.

This system embodies the very best of what I know. An "ultimate design," which also aims for simplicity and elegance—a 15" three-way system with two SB Audience (SB Acoustics) 15OB350 woofers and one B&C 15CXN88 constant directivity horn coaxial (**Photo 1**).

These speakers are made from book matched slabs of spalted sycamore (**Photo 2**). Spalting is when fungi has grown in the wood, and in right proportions dramatically emphasizes the grain. I picked out the wood at Schroeder Hardwoods in Harvey, IL. Seth Cothron of @studio38designs performed the carpentry.

Of course, achieving flat frequency response is important. But merely accomplishing that is minor league. Today with modern crossover design software and DSP, anyone who can't achieve that isn't trying. What separates the men from the boys is: how many other priorities you can simultaneously achieve.

The Design

When I invite friends over, they are in awe at the transparency, realism, impact, and imaging. Most people have never heard sound like this (**Figure 1**). I chose to build an open baffle system because nearly all "box" designs sound… boxy. Open baffle speakers present a huge sound stage, a room-filling sense of space, an open, effortless stereo image and a palpability that must be experienced.

Open baffle speakers, being dipoles, spill near-zero radiation to the side. That offers you many interesting possibilities in room placement. If you're a vinyl aficionado, place your turntable in the null point and you reduce acoustic feedback by 10dB. You can also place the speakers close to side walls without side reflections.

Most people have never heard of a speaker that has "dynamic range to burn" (i.e., it can easily exceed 120dB even if you never turn the volume above 90). You haven't heard drums until you've heard them this way. Bill Duddleston of Legacy Audio says, "If you can see the cone moving, it's distorting." With three 15" cone surfaces, you have to push them past 100dB before cones move visibly. Distortion is incredibly low at any reasonable volume level. Compression is negligible. High efficiency: These play as loud as most people want with flea-watt tube amps. My amps are 60W per channel (bi-amped, four channels total) and I can't drive them into clipping with any of my albums at any sane listening level.

This design simultaneously prioritizes several aspects of speaker design including constant directivity across the entire audio spectrum. This provides amazing precise imaging everywhere in the room—even when you're literally standing right next to one speaker! I've hosted listening parties with 20 to 30 guests—every person in the room hears a great stereo presentation. You can walk all around the room and the stereo image is stable. You're not confined to a sweet spot (**Figure 2**).

Constant directivity says that the speaker gives flat frequency response on axis, and as you go off axis, the level drops evenly (and not just at the top of the woofer or tweeter's range). Response remains even everywhere. With proper speaker positioning it sounds great all around. At 45° off axis, the level drops 6dB. At 75° it drops 12dB. The 90° side radiation is down 15dB or more.

The late Siegfried Linkwitz devoted considerable focus to open baffle designs. He insisted one of the reasons open baffle speakers sound so great is that reflections from room walls are consistent with the direct sound. This is seldom the case with conventional speakers, which have strong high-end in the sweet spot but radiate a bass-heavy sound in most other directions. This is unnatural, and your ears can tell. It is apparent when you're not in the sweet spot, or even when you're in an adjacent room. A speaker that's dipolar across the entire range radiates frequencies evenly throughout the whole room (**Figure 3**). Some of the finest horn and studio monitor designs achieve constant directivity above 500Hz, but this system achieves constant directivity across the entire spectrum.

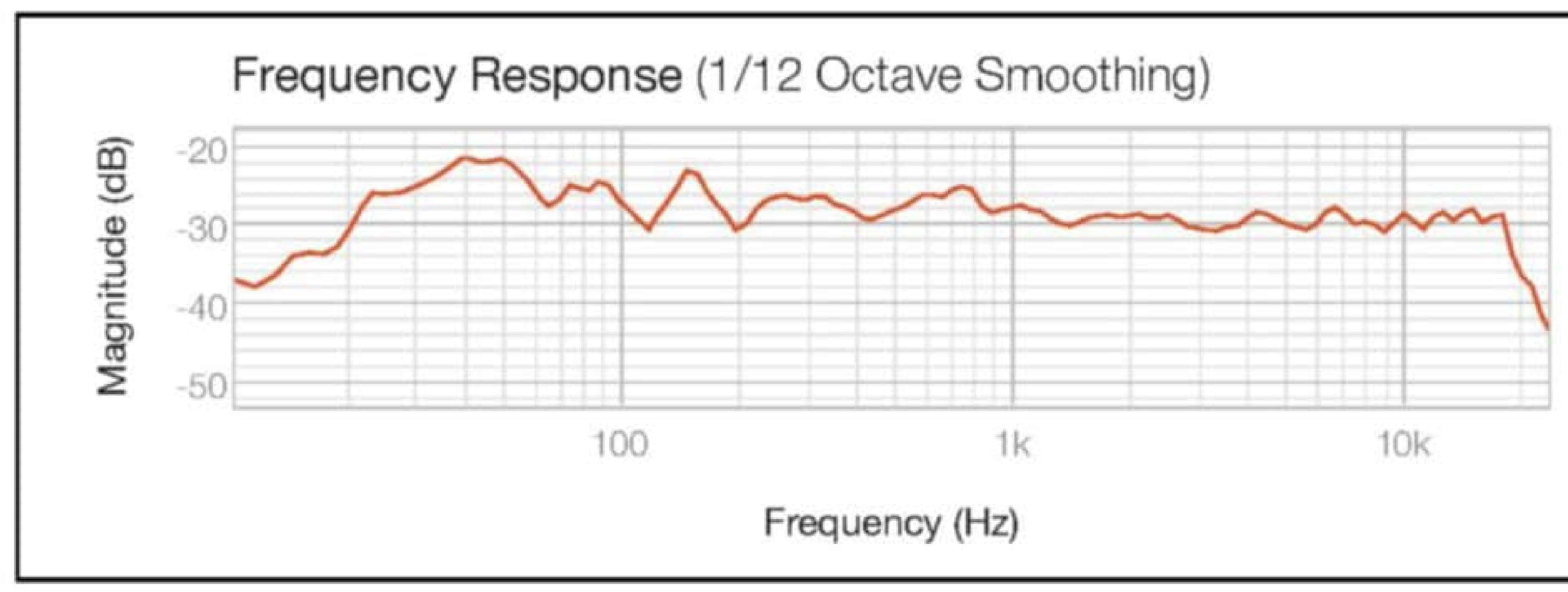


Figure 1: Frequency response in a real room, mostly at listener positions, averaged across both speakers and 12 locations

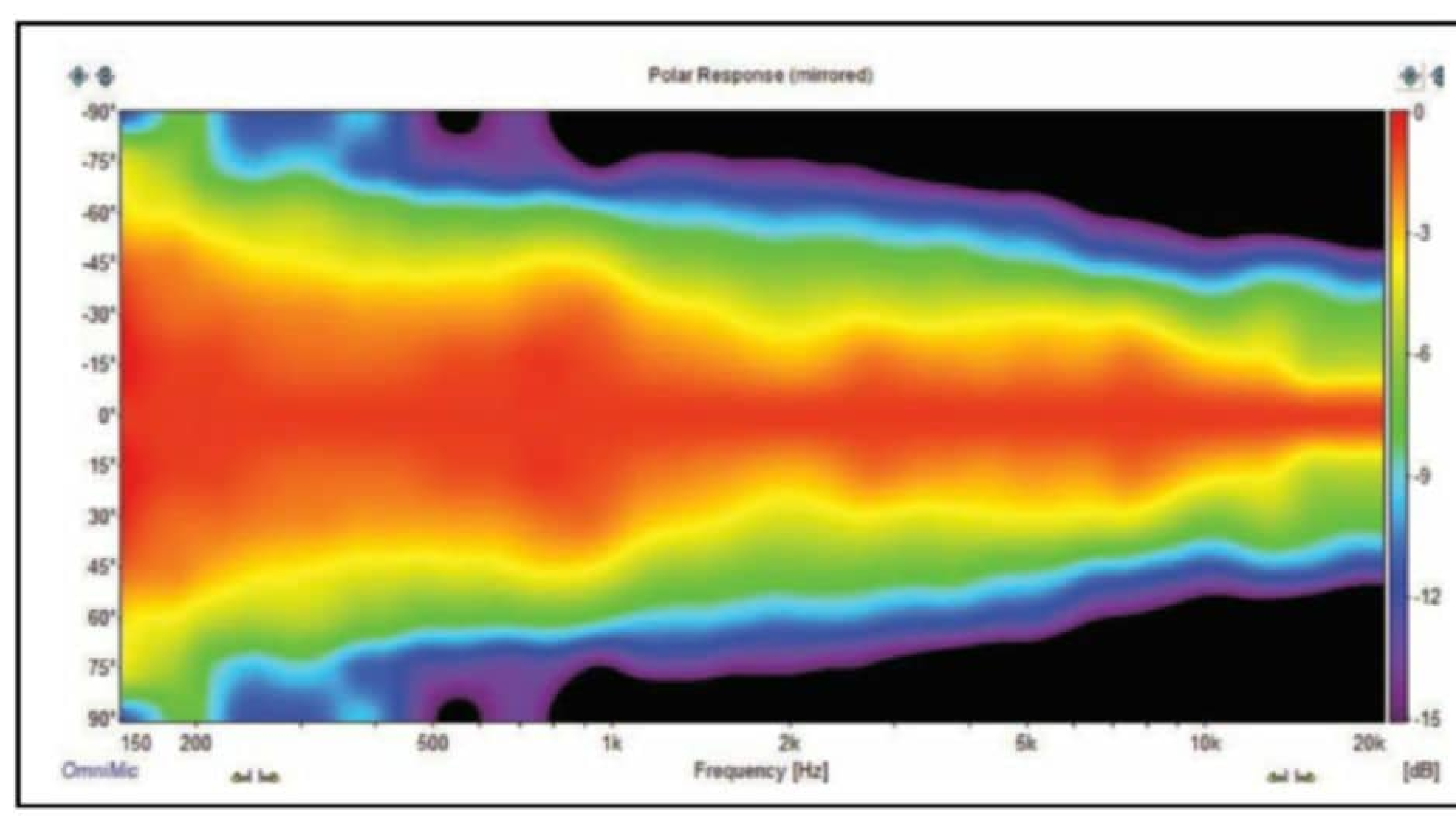


Figure 2: Speaker directivity pattern, 150Hz to 20kHz, shown with the vertical scale in 15° steps

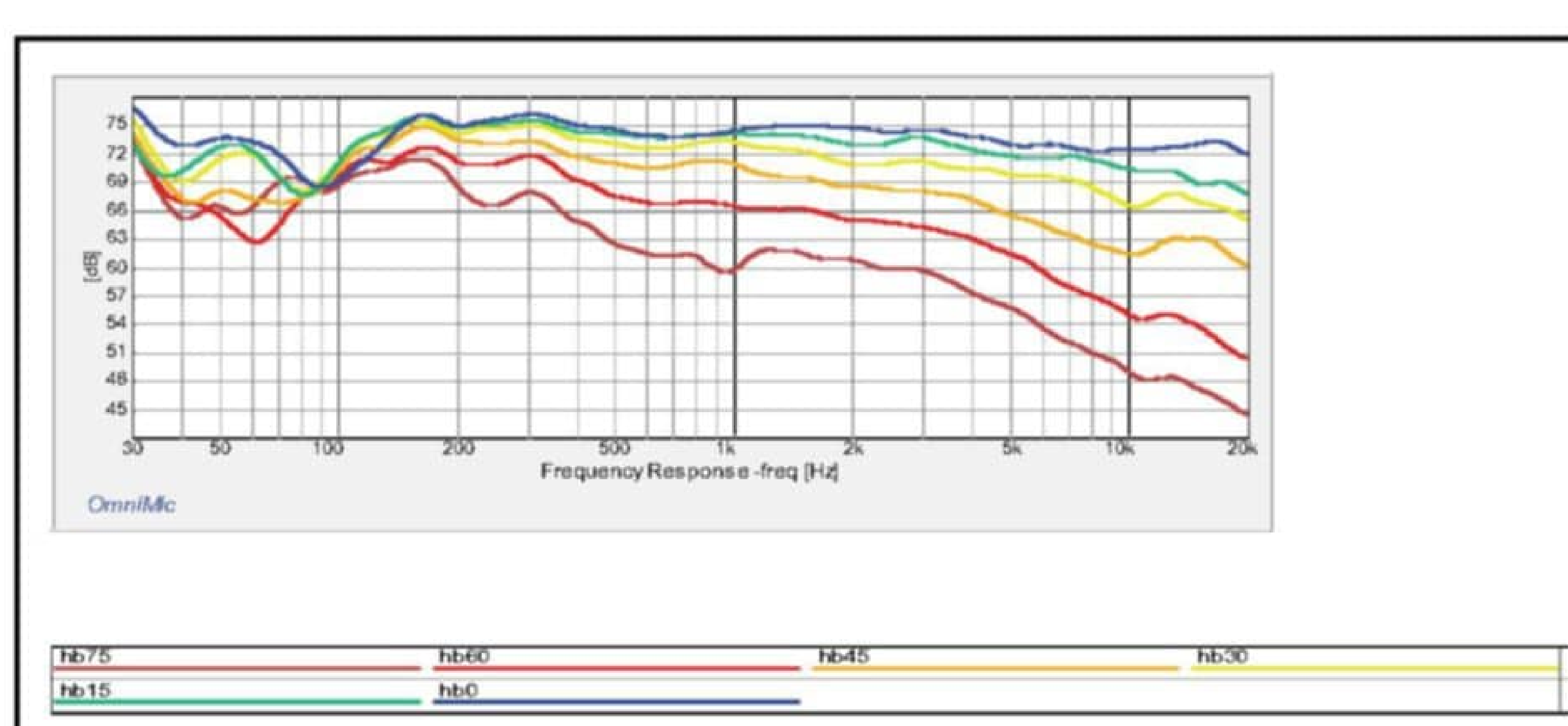


Figure 3: The near-field frequency response is shown at 0°, 15°, 30°, 45°, 60°, and 75° off axis. I don't have a good way to measure these at low frequencies, so you see odd behavior around 100Hz in the nearfield that is not present at greater distances.

Linear phase further contributes to incredible precise and stable stereo imaging. It also delivers incredible slam with percussion because the wavefront is steep with no phase inversions or delays. This is a huge key to soundstage and it enables you to resolve very fine details in recordings that were previously obscure (**Figure 4**).

As you can see, the impulse and step responses shown in **Figure 5** and **Figure 6** are excellent. There is no overhang on the step. Paying attention to the "time domain" does pay off in speaker design. Some audiophiles debate whether this is audible. I can't prove phase response is audible as such, but I believe it affects stereo imaging. Since phase is one of the most readily measured aspects of a speaker, any competent designer ought to optimize it whenever possible. **Table 1** lists some key specifications for these speakers.

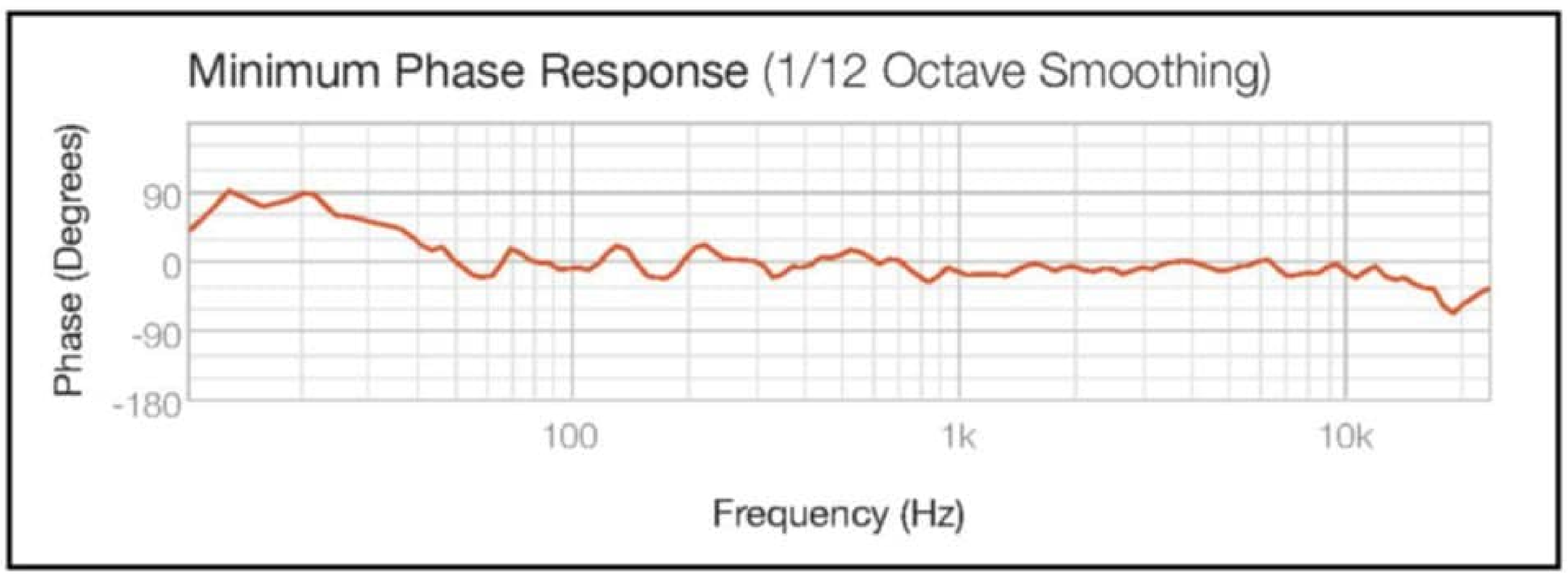


Figure 4: Phase response at listener position, averaged at five locations around my couch

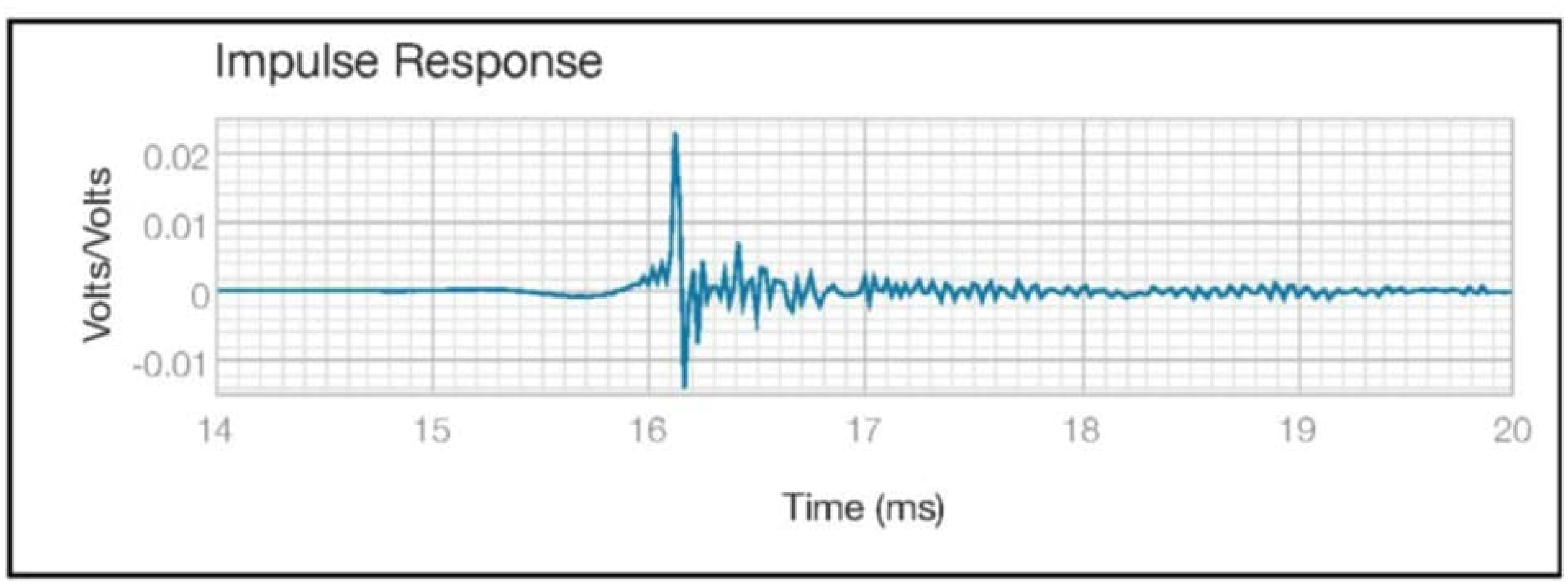


Figure 5: Speaker impulse response

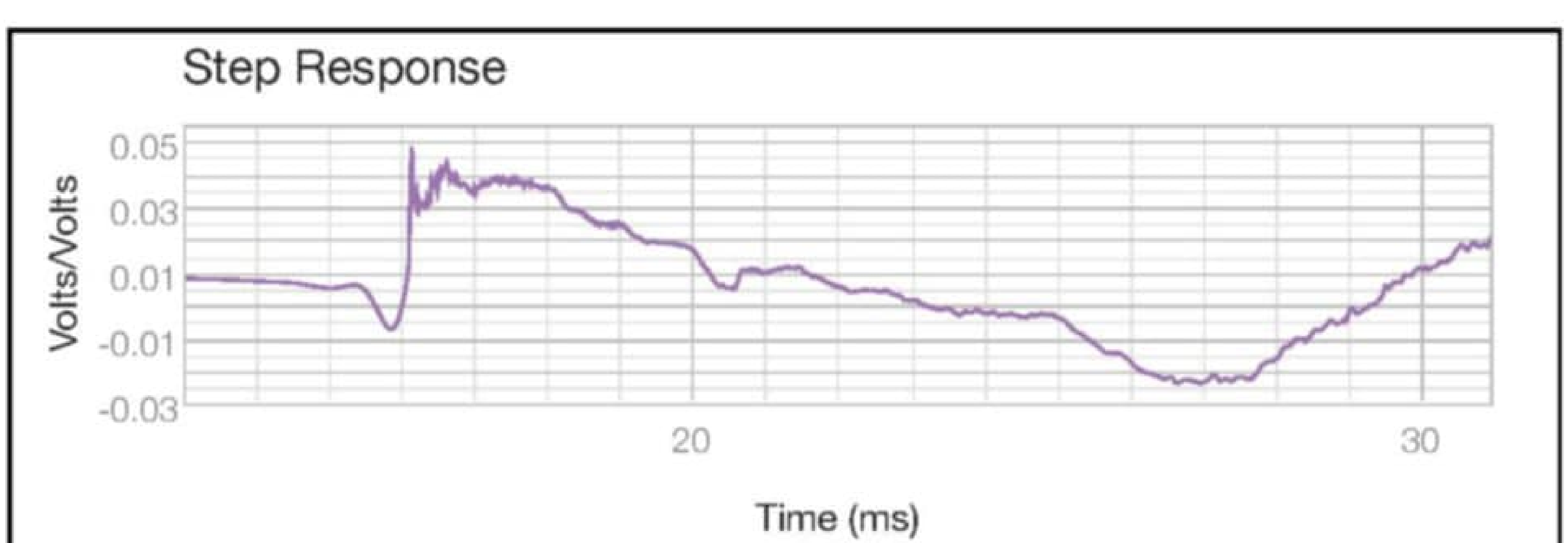


Figure 6: Speaker step response

Frequency response	-6dB at 19Hz to 19kHz
Sensitivity	100dB with 2.83V (1W) at 1 meter
Directivity	±2dB across multiple listener positions from 200Hz to 15kHz; ±45° constant directivity
Phase response	±30° from 80Hz to 15kHz
Power handling	500W
Nominal impedance	4Ω

Table 1: Key speaker specifications

Most dipole/open baffle designs are mounted on a flat board, so the dipole cancellation effect kicks in below about 80Hz. That means by 20Hz you've lost 12dB (-6dB/octave). I added "wings" on this design to eliminate about half of this cancellation.

Figure 7 shows the design sketch that I gave to my carpenter, Seth Cothron. He followed this closely; your own nuances will depend on the wood stock you select. Seth made wooden grille hoops slightly smaller diameter than the 16.375" diameter openings. With grille cloth stretched over them, they fit snugly. If I were to build these again, I would angle the front panel back 5° instead of the 10° you see in the sketch.

Then the bottom two woofers have "cancellation prevention" by virtue of the U-Frame cabinet sides, which extends the middle woofer down to about 55Hz before rolloff begins, and 35Hz for the bottom woofer before rolloff begins. This adds much bass extension while maintaining the overall dipole sound distribution pattern. The side wings then function as a short transmission line, with resonances between 150Hz to 250Hz. By crossing over at 110Hz, I sidestep consequent peaks in the upper bass. The low-end resonant frequency of the system is about 26Hz, with a Qt of 0.8.

The top woofer operates in full dipole mode across its entire range. This combination of three woofers, the "U-frame" wings for the subwoofer and the 110Hz crossover yield a constant directivity composite with 20Hz to 20kHz bandwidth.

The Crossover Circuit

The crossover circuit diagram is shown in **Figure 8**. I thought about tri-amping these, which you certainly can do, but I felt it was considerably simpler to use a two-way MiniDSP 2×4HD (fantastic product) and bi-amp them, with a very simple 6dB/octave PASSIVE series crossover between the subwoofers and the mid-woofer.

I used a series crossover instead of the more common parallel topology, because a series crossover has much cleaner interactions with impedance peaks of the woofers.

This however will not create a true 6dB crossover, because the mid woofer is acoustically rolling off at 24dB per octave below 80Hz. Thus, the phase

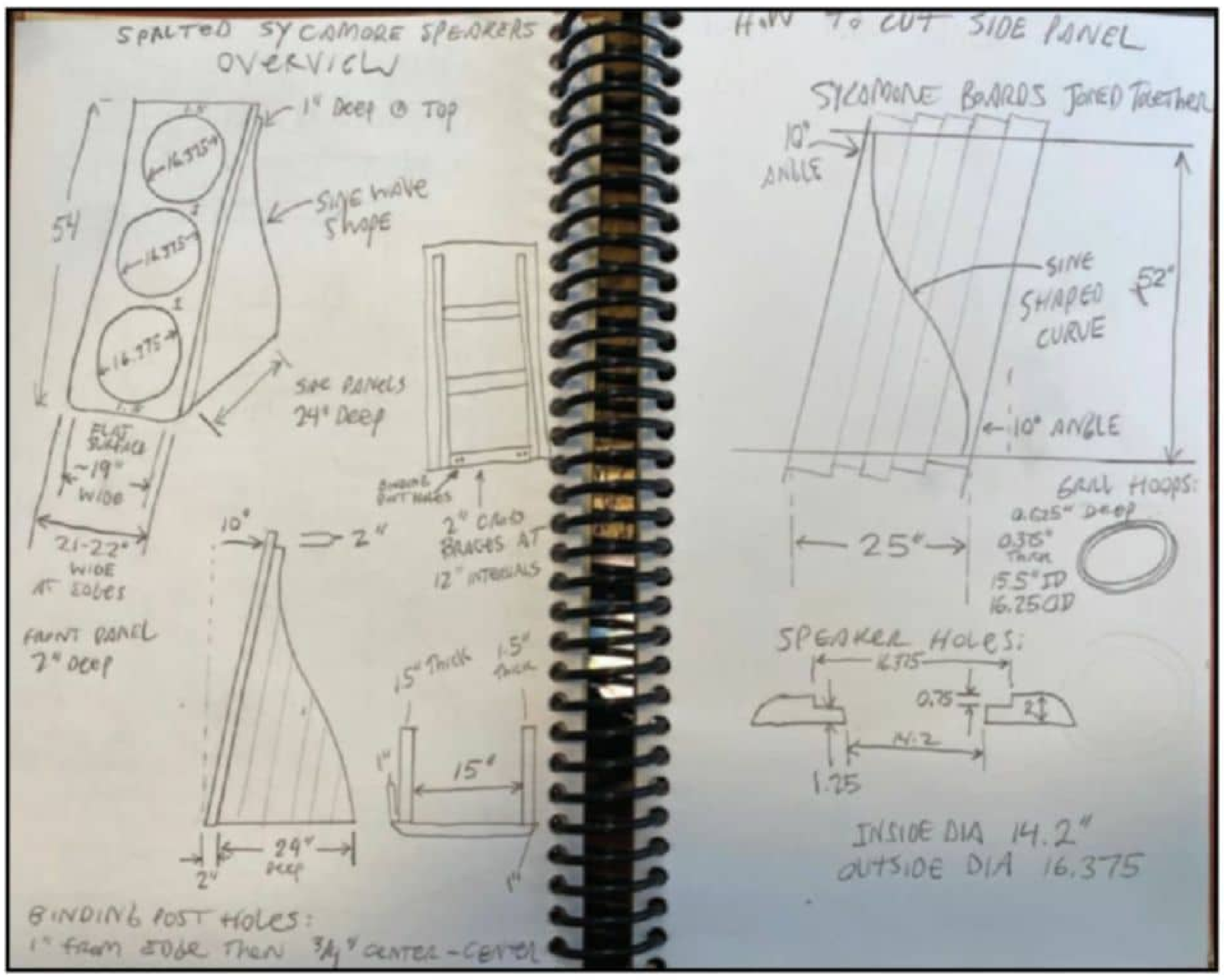


Figure 7: This is the cabinet sketch of the speaker that I gave to my carpenter.

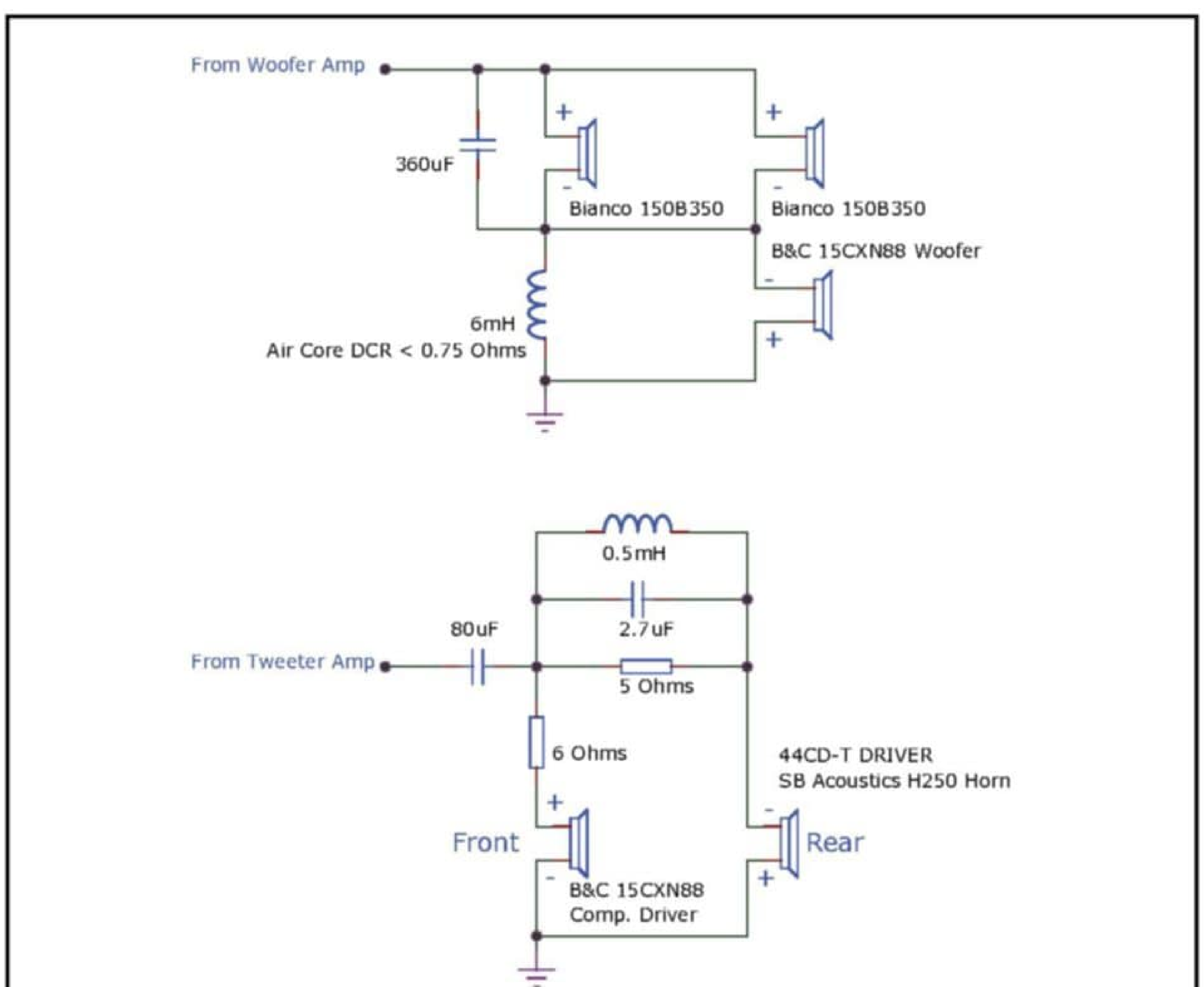


Figure 8: Passive crossover circuit diagram

About the Author

Perry Marshall is an author, speaker, consultant, engineer, and scientist. He's known for his books *Evolution 2.0* and *80/20 Sales and Marketing*, and his $10 million technology prize for the origin of the genetic code. His website, which links to other speaker designs, is www.perrymarshall.info. He lives with his wife and six kids (four standard-issue and two adopted) in Chicago, IL.

of the mid woofer must be reversed to prevent cancellation, which you can see on the schematic (**Figure 9**).

In order to restore linear phase, I used Eclipse Finite Impulse Response (FIR) filter software to insert a phase inverting "max phase" filter at 110Hz. This is loaded into the MiniDSP FIR function. I also used the FIR filters to linearize the phase of the B&C Speakers 15CXN88 woofers at the 1100Hz upper crossover frequency and flatten the phase of the B&C 15CXN88 horn tweeters at the 1100Hz upper crossover.

Horns almost never have the ruler-flat response of great dome tweeters. Domes are very "obedient," while horns require adult supervision. I have come to prize professional coaxial horns with constant directivity. You just must tease the best performance out of them. Constant directivity horns have a downward tilt in the treble, and you have to compensate for the idiosyncrasies in your crossover design. Horns demand more complex crossovers.

There is a payoff. With domes it is not possible to control directivity without a waveguide, and domes produce high distortion even at modest sound pressure levels. Horns in the hands of a skilled practitioner do not sound like the PA systems of the 1980s with their harsh spitty signature and coloration. Properly designed horns sound marvelous and have much lower distortion than domes. When you tame horns, best done with DSP, you enjoy exemplary dynamic range, high efficiency, and constant directivity.

Combining IIR and FIR Filters to Get the Optimum Crossover

Infinite impulse response (IIR) filters are the default DSP feature in the MiniDSP units. FIR requires special software. IIR is "minimum phase," which means that anything you do to alter the shape of a frequency response curve also has a non-negotiable phase penalty. FIR allows you to control time and frequency separately, but it's hard to pull off.

In this design, I did all of the crossover and major response shaping with IIR filters and then added FIR to solve minor frequency response and phase issues. The tweeters employ 80µF capacitors to prevent damage. They roll off below 300Hz, so the capacitors do not effectively factor into the crossover design. The woofer 6dB series crossover has a very high quality, low DCR inductor (<0.6Ω).

The B&C Speakers Woofer and Horn

The B&C Speakers 15CXN88 coaxial is a 15" paper cone woofer plus a 3" titanium compression constant directivity tweeter in coaxial configuration. At the 1100Hz crossover frequency, the polar

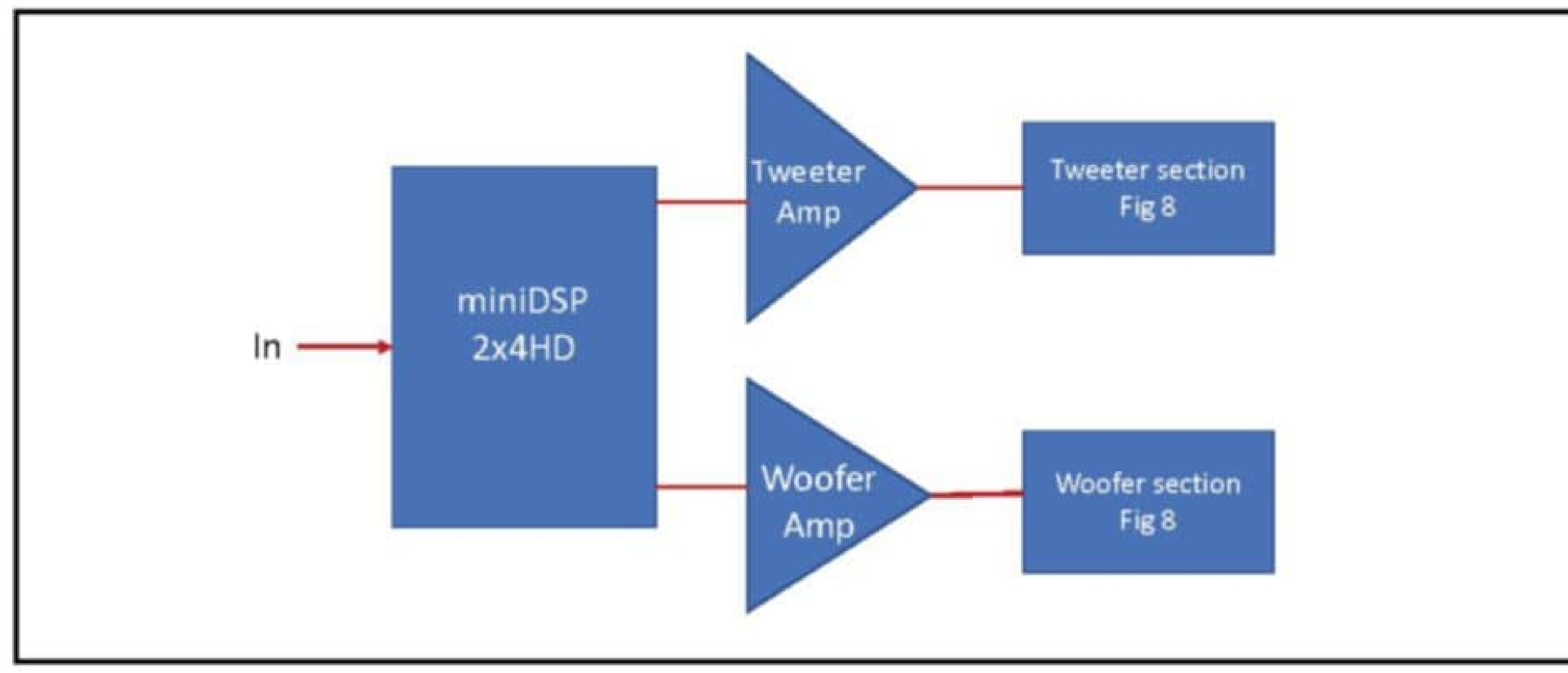


Figure 9: Active two-way crossover setup

pattern of woofer and tweeter match exactly and seamlessly. My measurements turned out better than the published factory specs. The woofer and tweeter SPLs are shown in **Figure 10**.

I EQ'd the tweeters with the MiniDSP and the Eclipse FIR software to obtain flat response both on and off axis. I chose the SB Audience Bianco 15OB350 woofers for their 11mm Xmax, high Q of 0.7, and 40Hz resonance, as they are designed for open baffles. In this design, they have a final resonance of about 26Hz. With 10dB of EQ boost they easily reach down to 20Hz.

In Open Baffle designs I've always felt that having an OB woofer firing on both sides, but tweeter firing on only one side, never sounds quite right. So, I added an SB Acoustics H250 horn and 44CD-T driver on the back, as shown in **Photo 3**. I added a small passive notch filter to flatten the response around 4kHz, which you can see in the schematic (Figure 8). The response at the rear is shown in **Figure 11**.

MiniDSP Files

The MiniDSP configuration files that I selected can be downloaded from the website: https://tinyurl.com/bitchesbrewdsp (Note: These will ONLY work on the MiniDSP 2×4HD and they are based on measurements of my drivers in my room. Use at your own discretion; all modifications are up to you.)

With 100dB sensitivity for the woofers and 106dB for the tweeters, one of the most important things is selecting amps with low noise! I use Adcom GFA2535s with the input levels turned down about halfway, which brings the noise down to below audibility.

These speakers particularly excel with percussion, strings, plucked instruments, horns, and aggressive bass lines, including electronic music. There is no music style where these do not excel, from Bruckner and Vivaldi to Yello, Hiromi, and Porcupine Tree. The resolution is incredible. You hear everything and the stereo image is huge. It fills the room and has a palpable sense of space.

Possible Future Improvements

Several modifications to this design could deliver even greater performance. For instance, choosing triamp instead of biamp, thus eliminating all passive. A good DSP option for triamping is the MiniDSP Flex

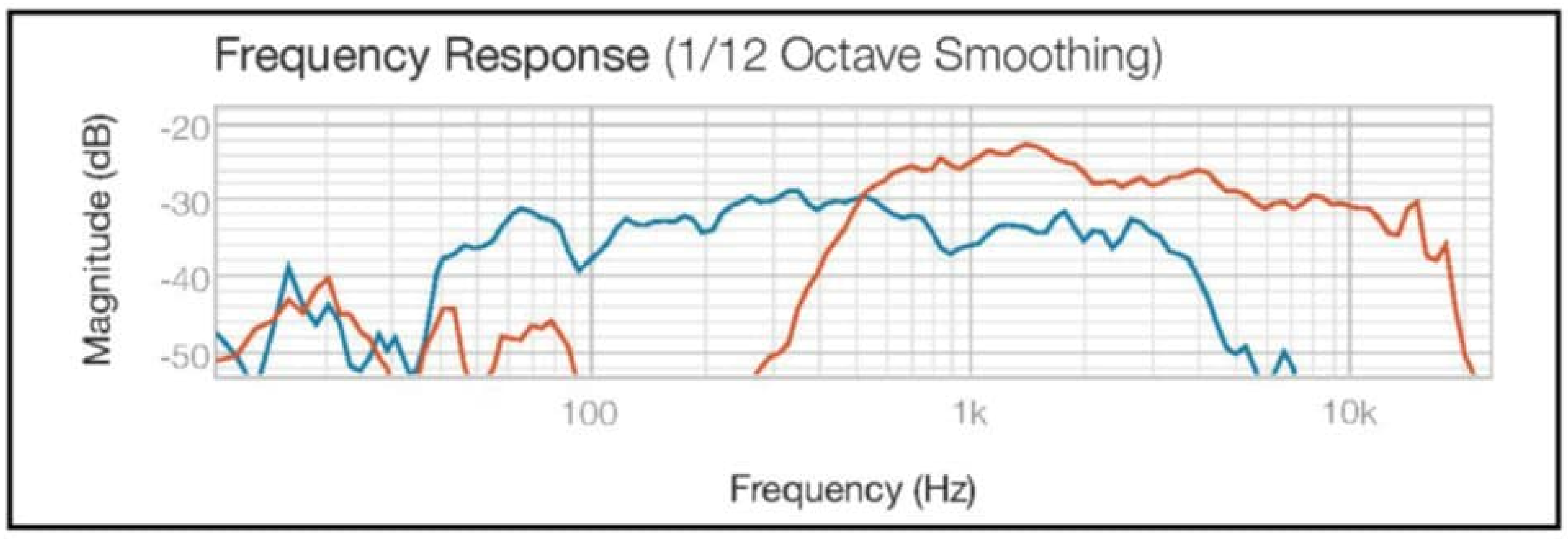


Figure 10: B&C Speakers 15CXN88 woofer and tweeter SPLs

Photo 3: The additional rear-firing 44CD-T tweeter is shown here.

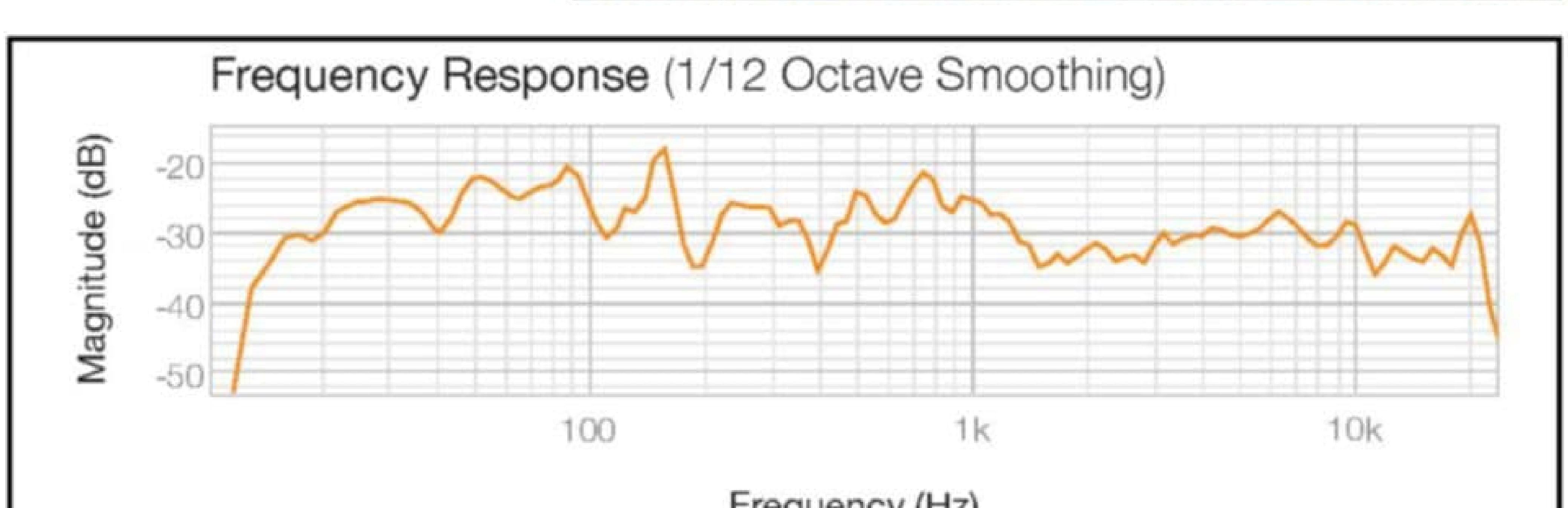


Figure 11: Response from rear-firing tweeter and woofer

Eight. I would still include the 80µF series protection capacitors for the tweeters.

The SB Audience woofers (**Photo 4**) have 11mm Xmax. A few woofers have more. I would only select high efficiency (95dB+) professional woofers for this design. I would not recommend conventional home audio 85dB efficiency high-Xmax subs, which are plentiful. This is because efficiency confers speed, realism and dynamics that low efficiency drivers inevitably sacrifice. This narrows your options to a very small number of high-end, high-efficiency, high-Xmax pro woofers from companies such as Faital and B&C Speakers. Most of those woofers have a low Qts. This is fine because you can always compensate with a DSP EQ boost. (Open baffle woofers traditionally are high-Q but that is unnecessary if you employ DSP. I would never attempt this design with a passive crossover.)

Of course, you could use 18″ woofers instead of 15″ woofers. But the 3× 15″ design option is elegant and there are no 18″ coaxials to match. You can also add L-pads to adjust the output of the rear-firing tweeter if you like.

Instead of buying the MiniDSP 2×4HD, you could use the DDRC-24, which has Dirac room and phase correction. Those who are not comfortable building FIR filters will find this option easier.

The Titanium horns have a barely noticeable trace of “Titanium harshness.” I used a Beryllium Radian 5208C in a different design and its top two octaves are slightly more transparent. There are a few candidate drivers I have not tried that might improve on the B&C Speakers 15″ coaxial. Radian Audio makes a 15″ coaxial with a Beryllium diaphragm; they also have a coaxial version with ribbon transducers. I am not aware of a coax with a Textreme™ carbon fiber diaphragm, but that would likely be a state-of-the-art, cost-no-object option.

What Do They Sound Like?

My brother-in-law Alan Pieratt, who’s been an avid audiophile for 40 years, stopped by and listened. He owns 18″ Velodyne subwoofers. He said to me, “Most subs have great specs but somehow sound like well-controlled thumps. Your open baffle woofers have such an open and natural sounding timbre, it’s the best bass I’ve heard in the business.”

When you play the song “My Bass” by Brian Bromberg, a gorgeous multitrack bass solo, the lowest pulses have a depth and transparency that ported and sealed speakers do not replicate. The mid bass and midrange exude a huge room-filling sense of air; high-end transients deliver speed and alacrity. There is never any sense that the speakers are straining to keep up.

"Flight of the Cosmic Hippo" by Bela Fleck shows off the sweet spot of these speakers, plucked strings: Fleck's banjo sounds like it's under a microscope, every incisive detail magnified to sound 10' wide. The low-bass solo in the middle of the song shakes the room and your chest without any compromise in musicality.

Some people claim the hardest instrument to reproduce is the human voice. I say it's the snare drum. Snare drums possess both explosive power and exquisite delicacy. On good jazz recordings you should be able to discern microdetail of quiet "ghost notes" on the snare drum including the shell and hardware; on loud whacks the speaker should explode with full energy without flinching. A slam on a floor tom should punch you in the gut without the slightest apology.

On Gavin Harrison's "Hatesong," you hear every articulation of the snare including the wires and the drumheads. The brass shines through with extreme resolution and microdynamics. The Bitches Brews also excel with choral music. Massed voices in "Silent Night" (Stille Nacht) by Oscars Motettkor shimmer with palpable, organic, human believability and goosebumps.

I look forward to hearing from other adventurers who build this design.

Photo 4: The upper driver is the coaxial B&C Speakers 15CN88, and the SB Audience 15OB350 woofers below have 11mm Xmax.

Building a Minimally Baffled Dipole Loudspeaker

Part 2 — Measuring the Drivers and Developing the Crossover

This article combines the theory, design, and construction of an open baffle loudspeaker, including crossover design, voicing equalization, and a practical example built by the author. In Part 1, the author reviewed the related theory and the approach to practical design and construction of the loudspeaker. In Part 2, he measures the drivers, develops the crossover system, and describes his own design.

"Baffle-less-ness," the author's four-way system open design, won first place in the 2022 Speaker Design Competition's Open Unlimited category.

By
Charlie Laub

Crossovers for open baffle loudspeakers often require more corrections than boxed loudspeakers and this is even more so for a nude dipole system. The crossover must flatten the responses, primarily of the midrange and woofer, EQ out any peaks in the response (typically in the midrange), implement crossover filters that define the band limits for each driver, and match the output levels of the drivers. The crossover is designed as the combination of a flat on-axis frequency response plus a voicing EQ curve. An example loudspeaker design that follows this approach is described.

Clean, high-resolution acoustic measurements are essential for creating an accurate model of the loudspeaker in the crossover design software package. Unfortunately, it is not always feasible for the loudspeaker builder to make such measurements directly, especially at low frequency. Out of practicality some measurements that include at least the effect of one room boundary are made. These must be interpreted and adjusted accordingly. In all cases it is strongly suggested that measurements are made with a fixed time reference to capture accurate phase information.

Acoustic Center Balancing

Before making a set of measurements there is a preliminary step in which the midrange and tweeter are physically aligned on the wireframe scaffold. Once these drivers are physically moved to the appropriate position their acoustic output will be as identical as possible as seen from the front and rear. This is a goal for the dipole loudspeaker and is why drivers with identical forward and rear response within their passband are utilized. In a dipole speaker only a physical adjustment of the relative driver positions can achieve the desired effect. In contrast, only the forward radiation from the drivers of a boxed loudspeaker reaches the listener and misalignment of driver acoustic centers can be removed using a digital delay or compensated for using an all-pass or other filter. But this does not work for a dipole. When you delay the tweeter signal, for example, as seen from the front it is akin to "moving the driver backward." But when observed from the rear, the driver would apparently be moved forward by delay. This means that, if the physical acoustic center is too far forward, using delay will have the apparent effect of moving it toward the rear as seen from the front, but from the rear the effect is to move it even farther forward, which would only make the offset problem worse. A better way to think about the delay is that an impulse sent to the speaker would be moved farther back toward the amplifier in the speaker wire and has the same effect on both the front and rear outputs. In a dipole system the acoustic centers of two drivers must be physically manipulated—this is not "aligning" the acoustic centers per se, only making the acoustic center offset between any two drivers the same as seen from both front and rear.

In general, if a planar tweeter and a moving coil "cone" midrange are used the appropriate tweeter position will lie near the plane of the midrange mounting flange. For those who wish to check the alignment via measurement, a technique for measuring the relative acoustic offset of two drivers has been described previously by Jeff Bagby and David L. Ralf [9,10]. In general, a series of measurements is performed in front of the loudspeaker and repeated at the rear. The interference technique is then employed to determine the offsets using software. When both the front and rear acoustic center offsets have been determined, the builder then physically moves one driver (typically the midrange) forward or backward so that the offset as seen from the front and the offset as seen from rear become equal. Note that this will not necessarily result in either offset being zero. If the front and rear offsets are identical, the crossover will have an identical effect at both the front and rear, which is the desired result. Keep in mind that, at 2kHz, an error in the alignment of one centimeter will result in approximately 17° of phase error.

Once the tweeter and midrange physical alignment has been performed, the next step is to obtain a set of measurements that are sufficiently "clean" to permit accurate crossover design. In this context, clean implies free of significant peaks and dips due to reflection from room boundaries. Often the technique that is employed is to window the time domain response to exclude the reflections and to move the loudspeaker as far away from boundaries as possible during the measurements. The loudspeaker design is a floorstander and the tweeter and midrange will be about 1m above the floor level. Attempting to directly measure 1m above the ground, at a typical listening distance of 2m to 3m and to a sufficiently high resolution and low frequency, would result in contamination of the measurement by ground reflection in the midrange band. For the midrange and tweeter, we can instead elevate these drivers by another meter or more, and measure at a close distance. This will largely eliminate the ground reflection problems for these drivers. On the other hand, our design relies on the ground reflection to augment the woofer panel output. As a result, the measurement on the woofer panel will be carried out on the listening axis at a distance and will include the floor reflection. These two different measurement conditions will subsequently be adjusted to a consistent basis before designing the crossover. (See the Supplementary Materials section on the *audioXpress* website for my own DIY measurement method.)

Crossover Development

Once these adjustments have been made to the measurement data, all measurements have been brought to the same location in space and the crossover design process can begin using crossover modeling software. I will now describe my approach to this task using a DSP (IIR filter) based framework.

Crossover development is carried out in two steps: flattening the response for each driver and applying the crossover filters. I typically choose to work on the midrange response first. This driver requires more correction than other drivers because its passband spans the dipole peak. The initial goal is to flatten the frequency response in a region extending well above and below the intended crossover points for this band. This has the additional benefit of flattening the driver's phase response within the intended passband. At this stage, the amount of power lift is not of concern since this will be largely eliminated when the crossover filters are applied if they are of sufficiently high order (e.g., fourth order). The first response area that should be worked on is the region around the dipole peak of the midrange. A combination of a peaking EQ band and a shelf filter work well, although it is a matter of some trial and error on the part of the designer to find the best solution. Additional shaping may be needed to flatten the response sufficiently overall.

Note that no crossover filters have been applied so far, only filters that flatten the response. For the woofer panel, the entire passband lies below the dipole peak and flattening its response is a bit more straightforward. First or low-Q second order shelving filters are effective in this regard. As with the midrange, the goal is to flatten and extend the response of the woofer panel, especially below the intended crossover frequency to the subwoofer. The tweeter typically requires the fewest corrections as the response is often relatively flat in comparison. After the driver responses have been corrected for dipole cancellation and flattened their levels should be made similar via gain adjustment.

At this point the design can progress much like with a boxed loudspeaker, with the goal of achieving a flat system response. Crossover filters should be

Photo 2 : This is the prototype loudspeaker.

applied at the target crossover frequencies (e.g., around 350Hz and 2kHz), as well as at the desired subwoofer to woofer crossover point. For the low frequency end of the woofer panel, Linkwitz [11] described a useful approach in which the second-order high-pass near-resonance driver response is transformed via the "Linkwitz transform" biquad equalizer into one stage of the high-pass crossover. For example, if the designer wishes to use a LR4 type crossover at 80Hz then the driver's Fs and Q can be transformed to F=80Hz, Q=0.707 to form one of the second-order filter stages that make up the LR4, with the other second-order stage implemented with an electronic filter also having F=80Hz, Q=0.707, the combination creating the LR4 response. Excess relative phase angle between a pair of drivers at their crossover point may need to be addressed so that the responses will sum correctly. The designer must decide how to continue to flatten and shape the overall system response. The over-arching goal at this stage is a completely flat system response with the drivers operating in-phase at the crossover points.

Voicing the Dipole Loudspeaker

The final stage of crossover development is often called "voicing." The voicing EQ correction is applied to correct the loudspeaker's tonal balance. A flat on axis response from a dipole system will often sound too bright. Research reviewed by Floyd Toole [12] and performed by Sean Olive [13] suggests that a down-trending power response over most of the audio band is preferred by both trained and untrained listeners and my own listening tests confirm this. Toole and Olive have published their findings as plots of the steady-state room curves for a variety of highly rated loudspeakers. The in-room steady-state response is similar to the power response of the loudspeaker, which may be estimated by inverting the loudspeaker's directivity index. One source of data regarding the directivity index of loudspeakers can be found online from published Klippel analyses [14]. For many well-regarded loudspeakers the same trend is observed: the directivity index begins to rise above 0dB between 100Hz and 300Hz, reaching 10dB or more by 10kHz. Since the loudspeaker's power response can be estimated from the inverse of this curve, the power response is undergoing a 10dB decrease over this frequency range. This power response trend influences how the loudspeaker sounds in the listening space since the off-axis acoustic energy that is radiated by the loudspeaker reflects from one or more room boundaries and returns to the listening location.

In contrast to a boxed loudspeaker, a nude dipole loudspeaker as designed in the way described in this article will have a flat on-axis frequency response and a flat power response because a dipole has nearly constant directivity. This characteristic makes it easy to tailor the power response and tonal balance by implementing a suitable voicing EQ curve to the crossover input, upstream from the crossover filters. I experimented over a long period of time, using more and less amounts of the down-tilting EQ and changing the range of frequencies in which the voicing EQ was applied. I found that a spectral tilt of approximately 4dB to 5dB per decade provided the most accurate and satisfying tonal balance. This is similar to, but steeper than, a "pinkening filter" used to convert white noise to pink noise. The down-tilt is used between 100Hz and 10kHz, typically amounting to 8dB to 10dB of correction in total. At the highest frequencies, the rising directivity of the tweeter will continue the trend without requiring EQ [15, Figure 4.20].

One way to implement this type of correction in DSP is via a filter bank consisting of multiple first order shelf filters, each applying only 1dB to 2dB of correction. Design of similar analog active pinkening filters is discussed by Douglas Self in Section 11.14 of his book [16], "equalizers with

About the Author

Charlie Laub holds a bachelor's degree from the University of Massachusetts and a Ph.D. from the University of Connecticut, both in Chemical Engineering. He has been building and learning about loudspeakers since he was 9 years old, and in his retirement enjoys applying his analytical skills and love of music to DIY audio. He has been experimenting with and building dipole and open baffle loudspeakers for more than 10 years and has developed several public-domain software tools related to acoustic measurement processing and crossover design such as the FRD Blender and Minimum Phase Extractor, and the ACD Crossover Design suite. He is a strong proponent of DSP processing in software on lightweight Linux platforms such as the Raspberry Pi and other mini PCs, and he has written several audio processing plug-ins for this purpose. At present, Charlie and his wife Birgit reside in Michigan but will be relocating to Bavaria, Germany in 2024.

non-6dB Slopes." For a passive crossover network, the spectral tilt should be targeted during the initial crossover development process instead of as a separate voicing EQ in addition to a flat response. At this point, the crossover development is complete and the system is ready for audition.

An Example of a Wireframe Dipole Loudspeaker Concept

I will briefly describe my own implementation (**Photo 2**), bringing the previous considerations into practice. A complete description and reference tables for my design are available in the Supplementary Material online.

I initially screened the drivers using measurements performed on- and off-axis to establish the range over which they were suitable for a dipole application (see for example Figure 4 and Figure 7 from Part 1 of this article, *audioXpress,* September 2023). This is most critical for the midrange, which must be used both above and below the dipole peak, followed by the tweeter. Because the woofers are used primarily at relatively low frequency only the driver's distortion performance, as measured by others [17], and its Thiele-Small parameters were considered as selection criteria.

The drivers used in my loudspeaker include:

- Aurum Cantus AST2560 tweeter with the face plate removed
- SB Acoustics SB17MFC-4
- Four Peerless 830869 HDS woofers in a 20″ × 20″ baffle

These were paired with a set of closed box subwoofers, consisting of multiple separate 12″ woofers, each in a sealed, 45-liter enclosure, placed against the front wall behind the dipole loudspeakers.

The crossover design was performed using the ACD design tools written by the author [18]. The DSP crossover was implemented using software based IIR filters on a computer running Linux and connected to a Behringer UMC1820 audio interface. Software included the LADSPA plug-in ACDf written by the author [19], with routing and processing tasks handled by a Gstreamer pipeline [20] run from the command line [21]. A list of ACDf filter types is provided in the Tables 1-5 document in Supplementary Materials.

As mentioned earlier in the text, I like to work on the midrange band first. The SB17MFC is one of the easiest midrange drivers to use in this application because of a lack of response anomalies in the dipole peak area and its relatively low distortion. I have found that a correction consisting of an EQ band and a second-order low Q shelving filter is a good starting point to flatten out the response. The goal is to make the response flat across as broad a band as possible. This typically means up to about where the cone breakup begins, and at least one octave below the intended crossover point. As an example, the midrange raw response is shown in **Figure 9** along with the flattened response resulting from these initial corrections. In this case I used a

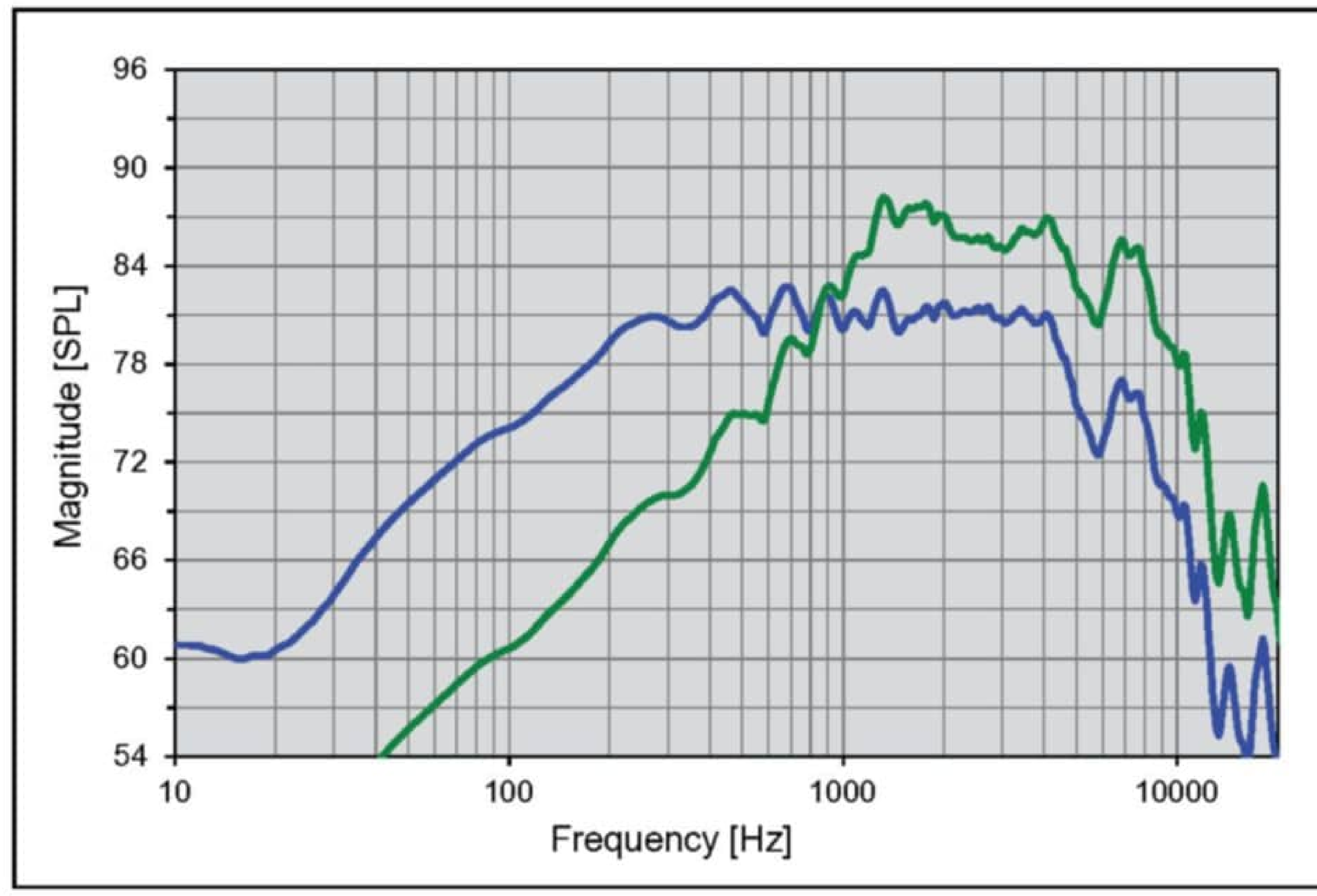


Figure 9: Midrange responses—raw (green) and after applying filters to flatten the response (blue)

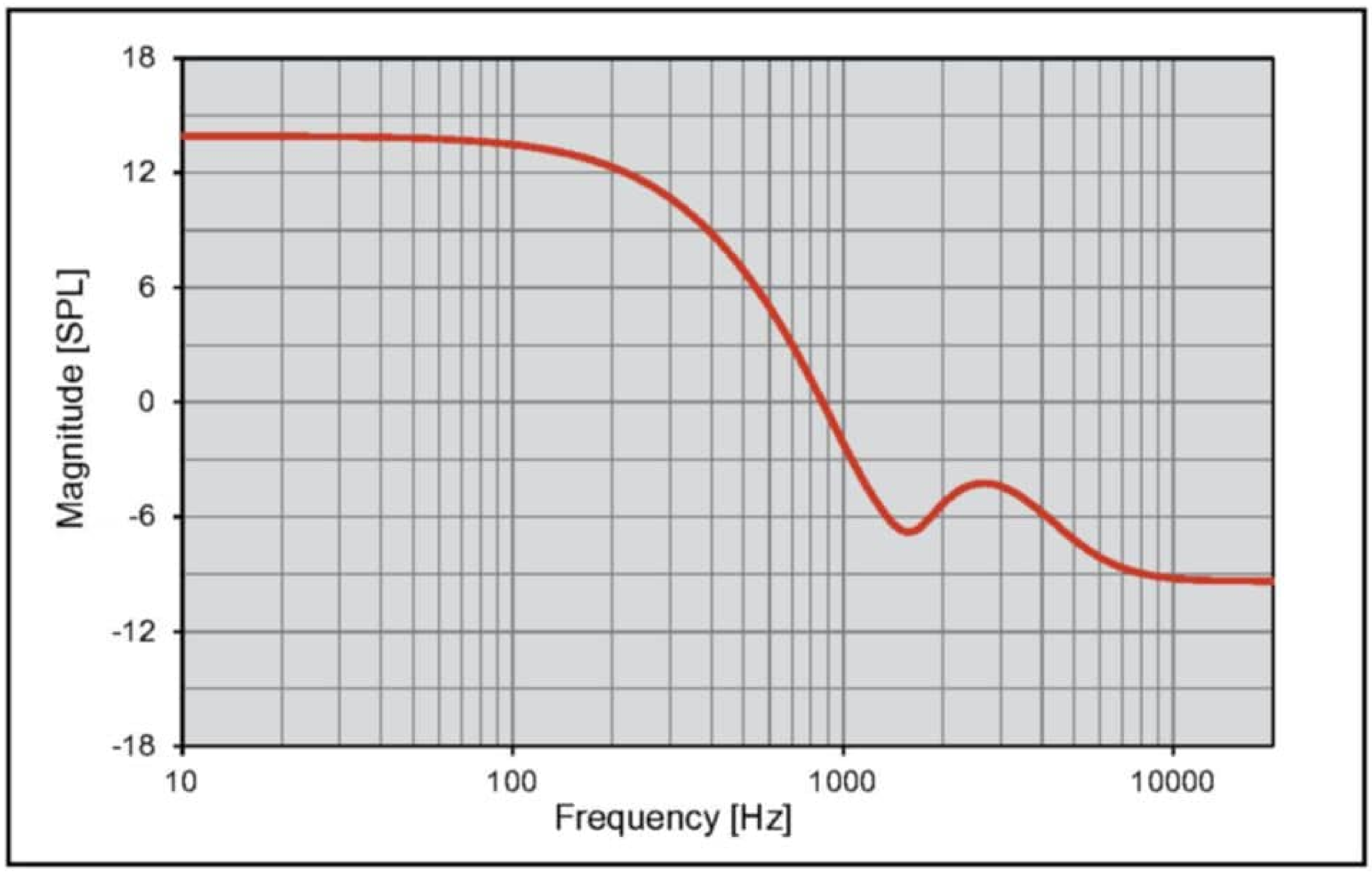


Figure 10: Response of the filters used to flatten the raw midrange response

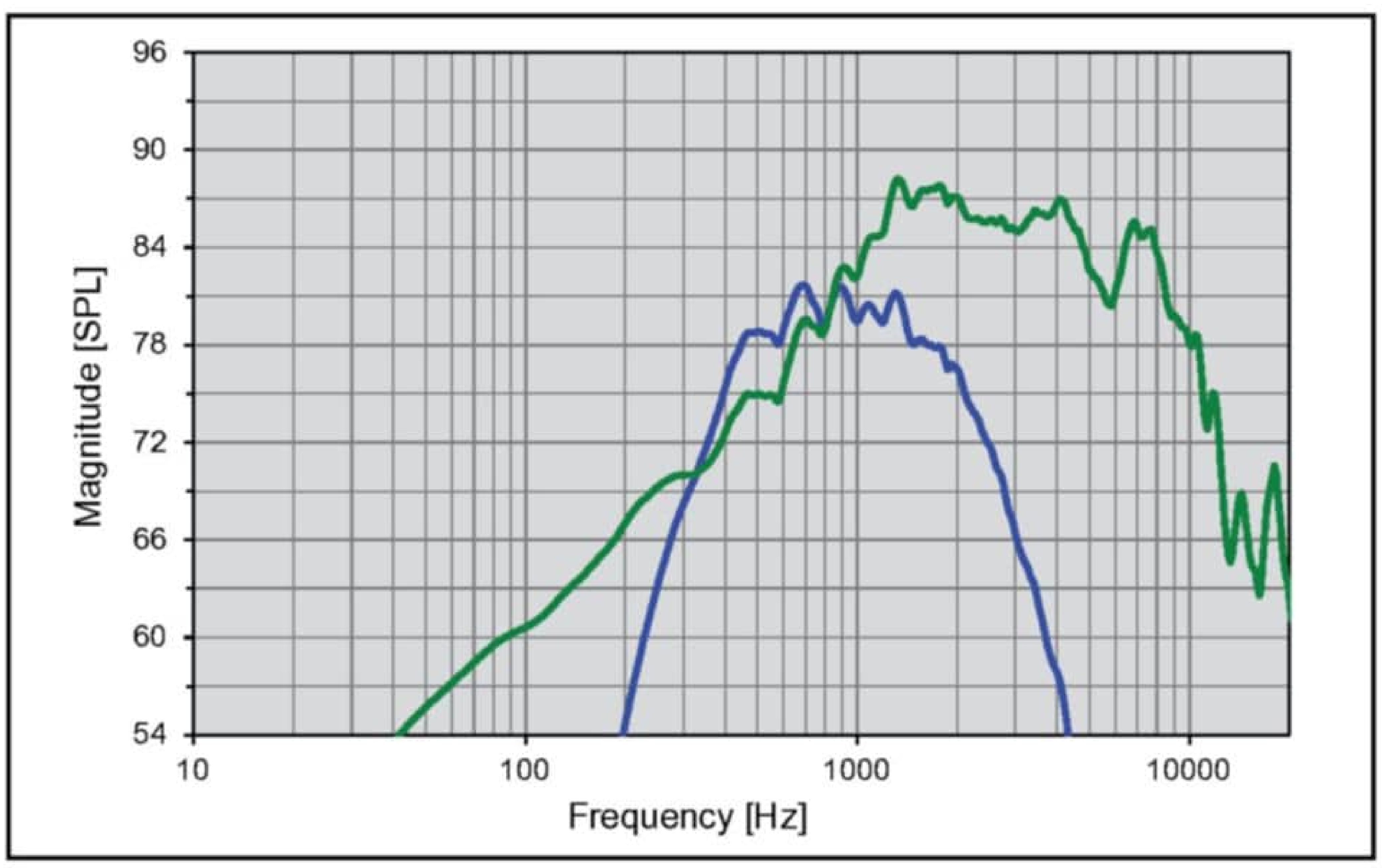


Figure 11: Midrange responses—raw (green) and with crossover filters applied (blue).

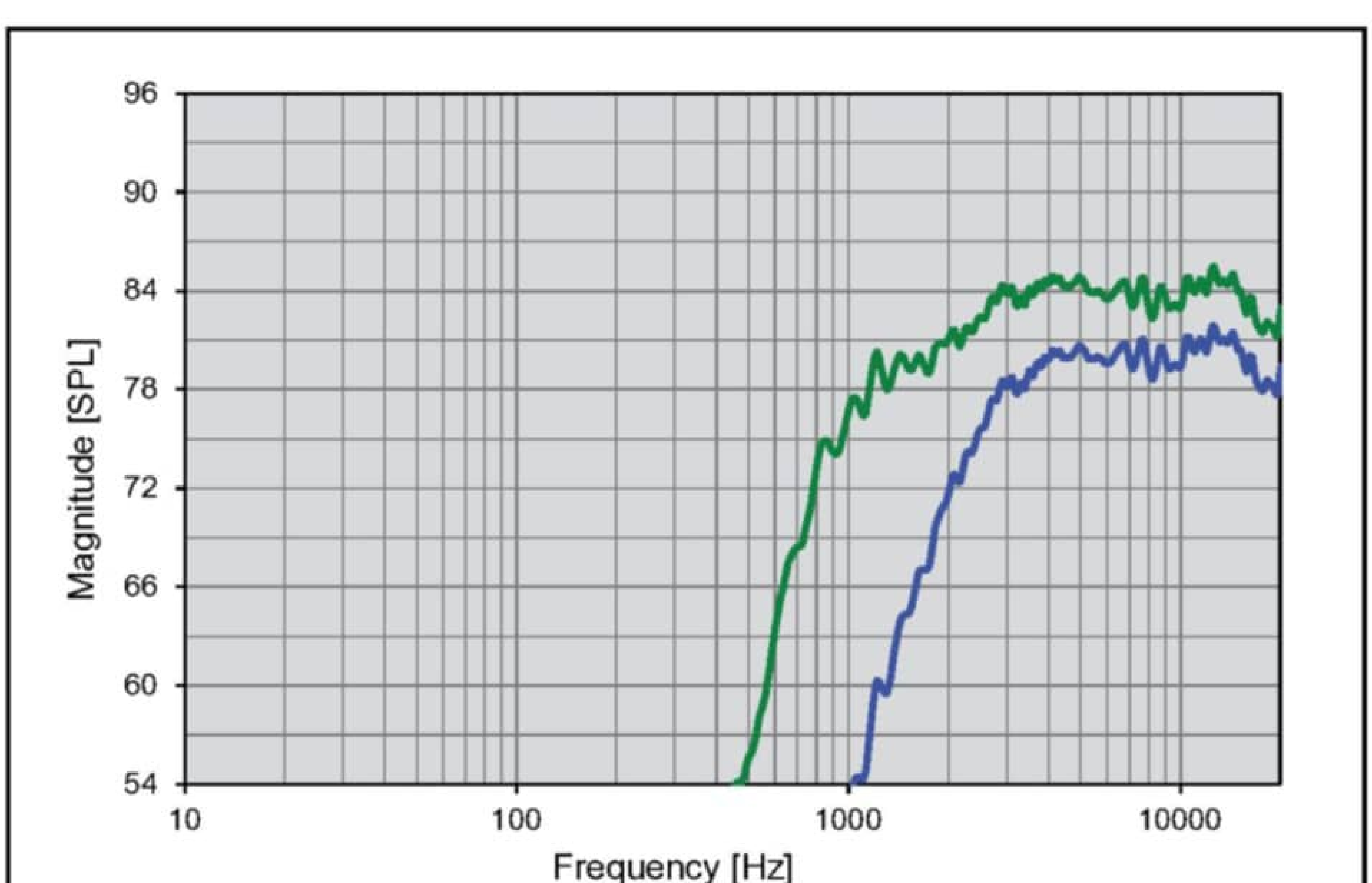


Figure 12: Tweeter responses—raw (green) and with crossover filters applied (blue)

biquadratic filter (type 28) to implement the shelf, since this permits different Q values for the start and end of the transition, and a PEQ band (type 26) having Q=2 at 1.6kHz to address the dipole peak. One additional other filter was used—a second-order shelving filter (type 25) further flattens the response above 2kHz. The response comprising these "passband flattening" filters is shown in **Figure 10**. Because fourth-order crossover filters will be used, the amount of low-frequency power lift will be strongly diminished below the eventual crossover point. With the response flattened, LR4 type crossover filters were added in the usual way. The resulting response is shown in **Figure 11**. The complete set of filters used for the midrange band are listed in Table 3 of the Tables 1-5 document in the Supplementary Materials.

The response from the minimally baffled tweeter is the easiest to work with in this loudspeaker. A shelving filter was used to correct a response drop below 3kHz. Thereafter, high-pass filtering complementary to that used in the midrange was created. This was implemented by the combination of a biquadratic filter (type 28) and a second-order high-pass filter (type 22). An all-pass delay was used to improve the phase angle match to the midrange at the crossover point. The raw, as measured tweeter response and the final response with crossover filters applied is shown in **Figure 12**. The filters used for this band are listed in Table 2 of the Tables 1-5 document in the Supplementary Materials.

Figure 13 shows the measured response of the woofer panel and the corrected response with crossover filters applied. The Peerless 830869 provides low distortion down to 80Hz and has an open basket. With four closely spaced drivers on a minimal baffle, the panel operates much like one larger driver at low and middle frequencies. Above 500Hz or so, there is an increasing amount of driver-driver interaction that causes nulls to form. In general, the panel has a dipole-like response, with a dipole peak around 300Hz. A first-order shelf (filter type 4) was used to compensate for the falling response below 300Hz. This was all that was needed to flatten out the response below the dipole peak. A second-order low-pass filter was used to define the upper end of the passband, targeting a 350Hz crossover point. This combines with the existing roll-off above the dipole peak to produce a higher effective slope that can match up to the fourth-order crossover filters used to define the low end of the midrange passband. To define the low end of the woofer passband, two filters were used. A biquadratic filter was used to transform the near-resonance response to Q=0.707 and F=80Hz. This forms one part of the LR4 crossover filter. The other part is a second-order high-pass filter, also with Q=0.707 at 80Hz. Because of the large amount of gain used in the first order shelving filter, there is an overall gain correction (filter type 0) of -16.25dB. The filters used for this band are listed in Table 4 of the Tables 1-5 document in the Supplementary Materials.

The system response (not including the subwoofer) before implementing the voicing EQ is shown in **Figure 14**, with the crossover filter responses for tweeter, midrange, and woofer panel shown in **Figure 15**. By choosing a flat response target, the task of getting the drivers well integrated and the levels balanced is relatively straightforward. This also decouples the task of integrating the drivers into a system from the task of voicing the loudspeaker. Because the measurement data is not extremely clean, there is some ambiguity to the exact levels of each driver, but in general the system response is approximately ±1dB between 150Hz and 10kHz.

The voicing EQ curve that was employed for this design is shown in **Figure 16** and the shelving filters are listed in the Supplementary Materials.

Listening Impressions

The loudspeaker was set up in the main listening area—a large, wide space with a vaulted ceiling. The front wall is a fireplace with a wide river rock face and a stone hearth. The system was powered via amplification of approximately 100W into 8Ω per channel, with identical amplifiers used on the woofer, midrange, and tweeter bands.

Well, how do they sound? I feel that they invoke a sense of a really large loudspeaker but without any of the associated coloration—it's a monkey-coffin speaker without the box resonance issues and with better imaging. The midbass especially sounds to me very clean and unrestrained. There is a large soundstage.

At the central listening location, the stereo image is similar to what one gets from a high-quality mini-monitor, which the baffle-less midrange and tweeter approximate. Overall, the loudspeaker provides a very satisfying musical experience, not just at the central listening location but in different parts of the home wherever I can perceive their output. This is likely because the power response has been shaped to be like the typical boxed loudspeaker while the dipole radiation character projects the same tonal footprint in all directions.

I hope that the reader now has some interest in building a minimally baffled dipole loudspeaker themself. What I have described in the prototype

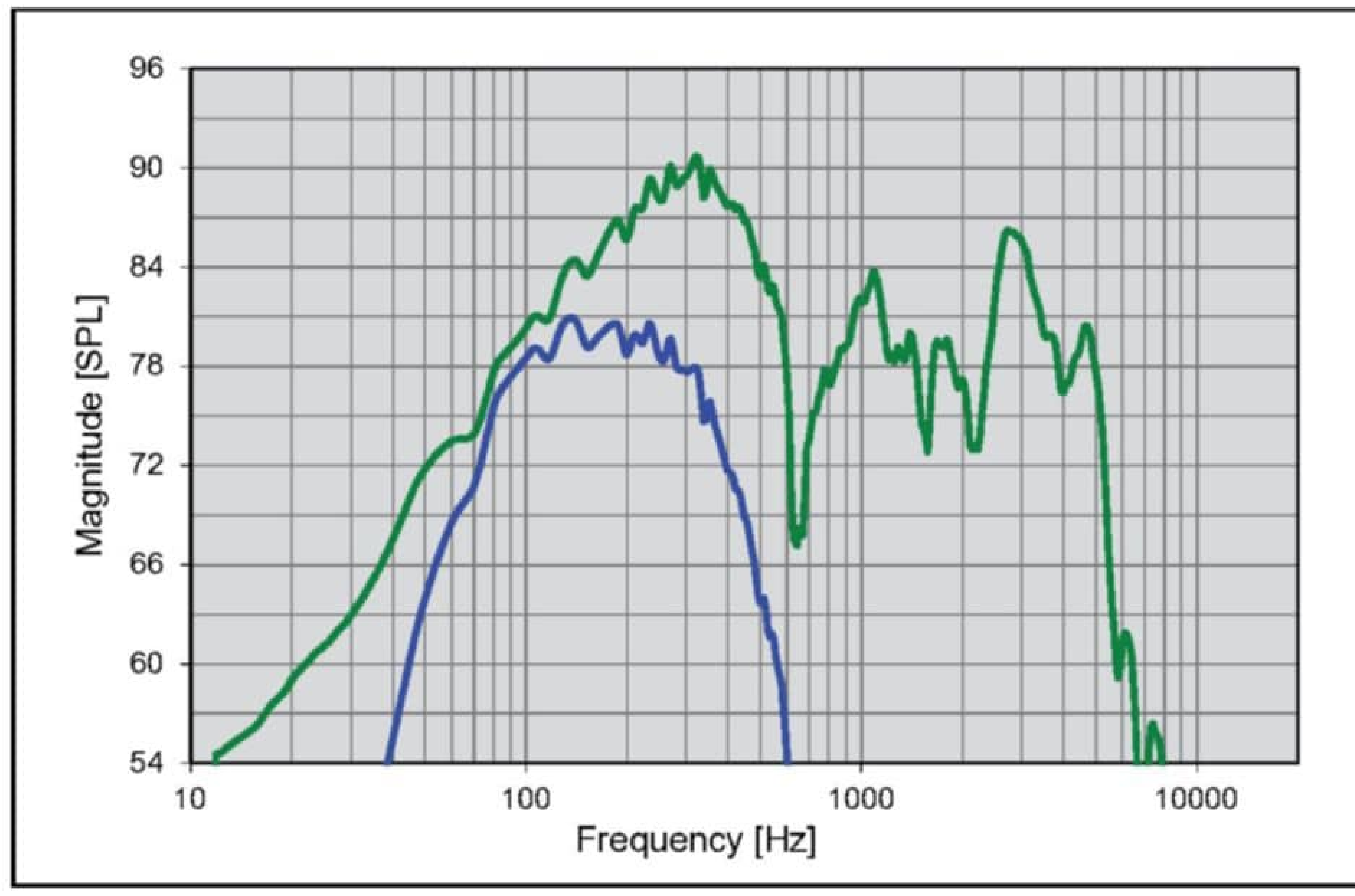


Figure 13: Woofer responses—raw (green) and with crossover filters applied (blue)

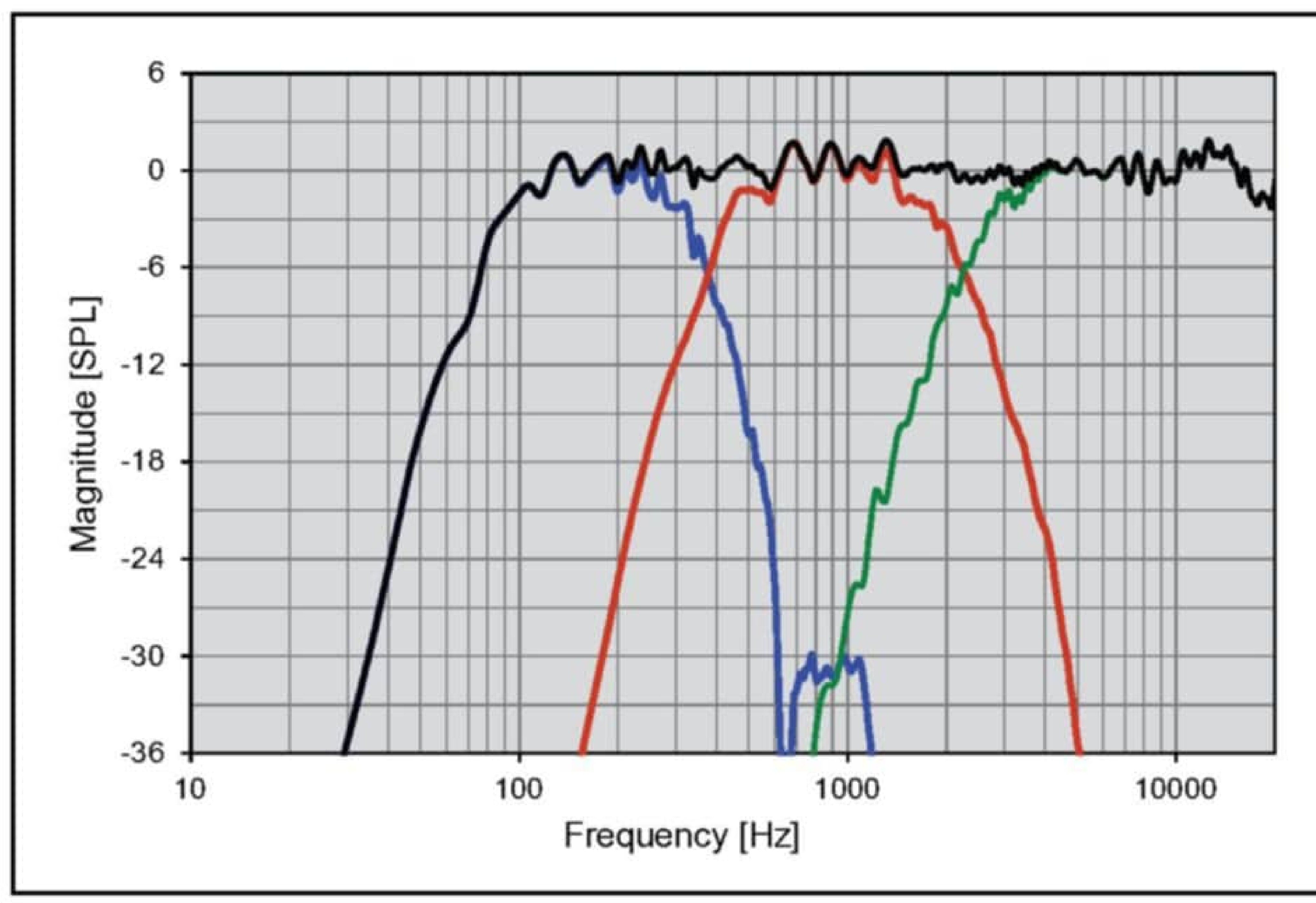


Figure 14: Responses after the crossover filters have been applied for the woofer (blue), midrange (red), tweeter (green), and the combined system response (black)

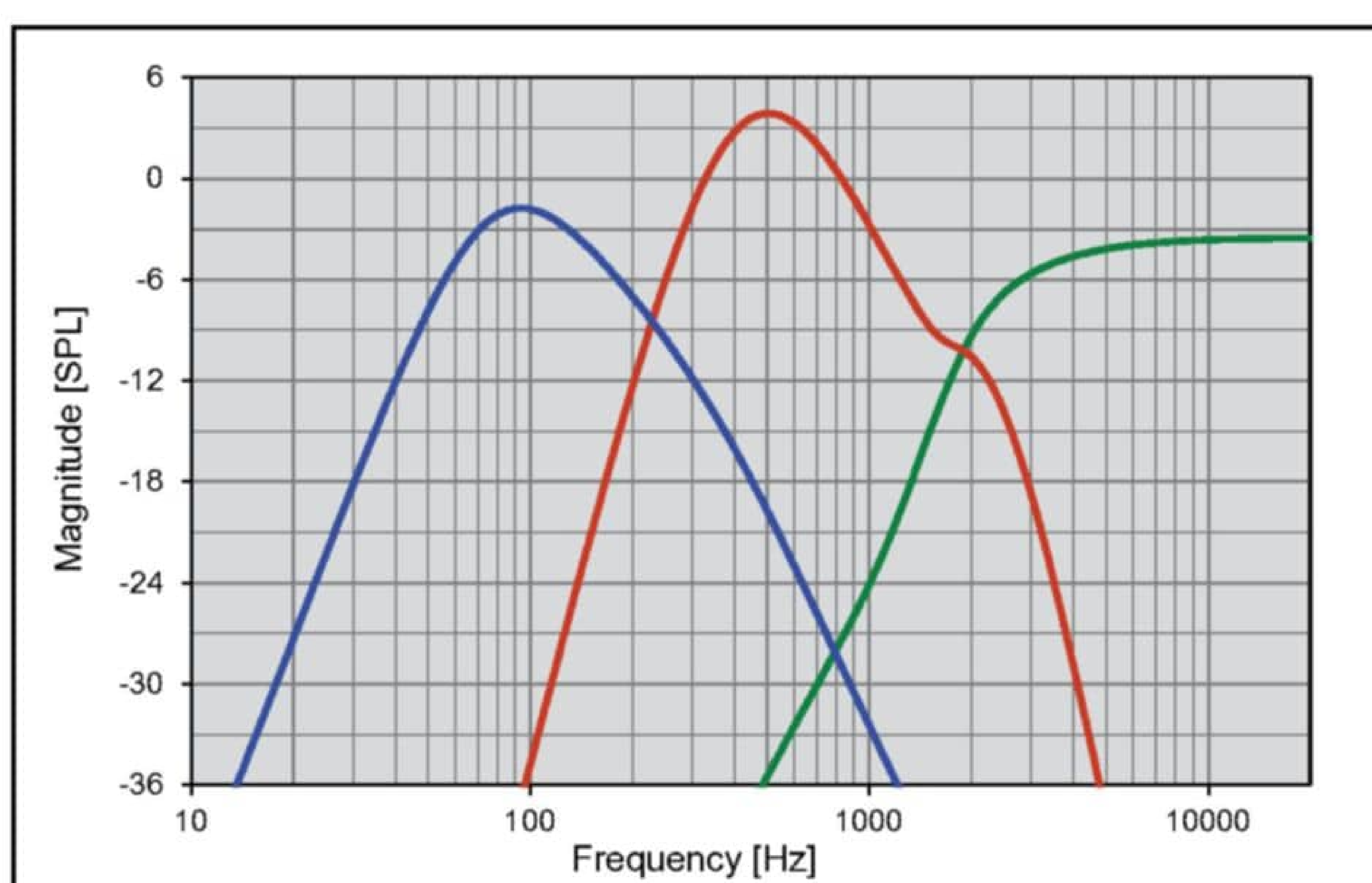


Figure 15: Responses of crossover filters for the woofer (blue), midrange (red), and tweeter (green)

loudspeaker example above is one possible embodiment. If the "minimal baffle" paradigm is followed there may be other ways to create the same type of loudspeaker. Some improvements could be undertaken as well, such as decoupling the woofers from the planar "woofer panel", since it is not the sound pressure in the air but the vibration the motor induces via the frame into the panel that leads to panel resonance. Another method, apart from the wireframe scaffold used in this embodiment, could provide support, decoupling, and a more aesthetically pleasing enclosure for the midrange and tweeter. An example of the minimal baffle dipole loudspeaker that I built can be found online in the Supplementary Material for this article (see the Project Files link included at the end of this article). I encourage builders to experiment and to make careful measurements when constructing their own version of this wonderful loudspeaker.

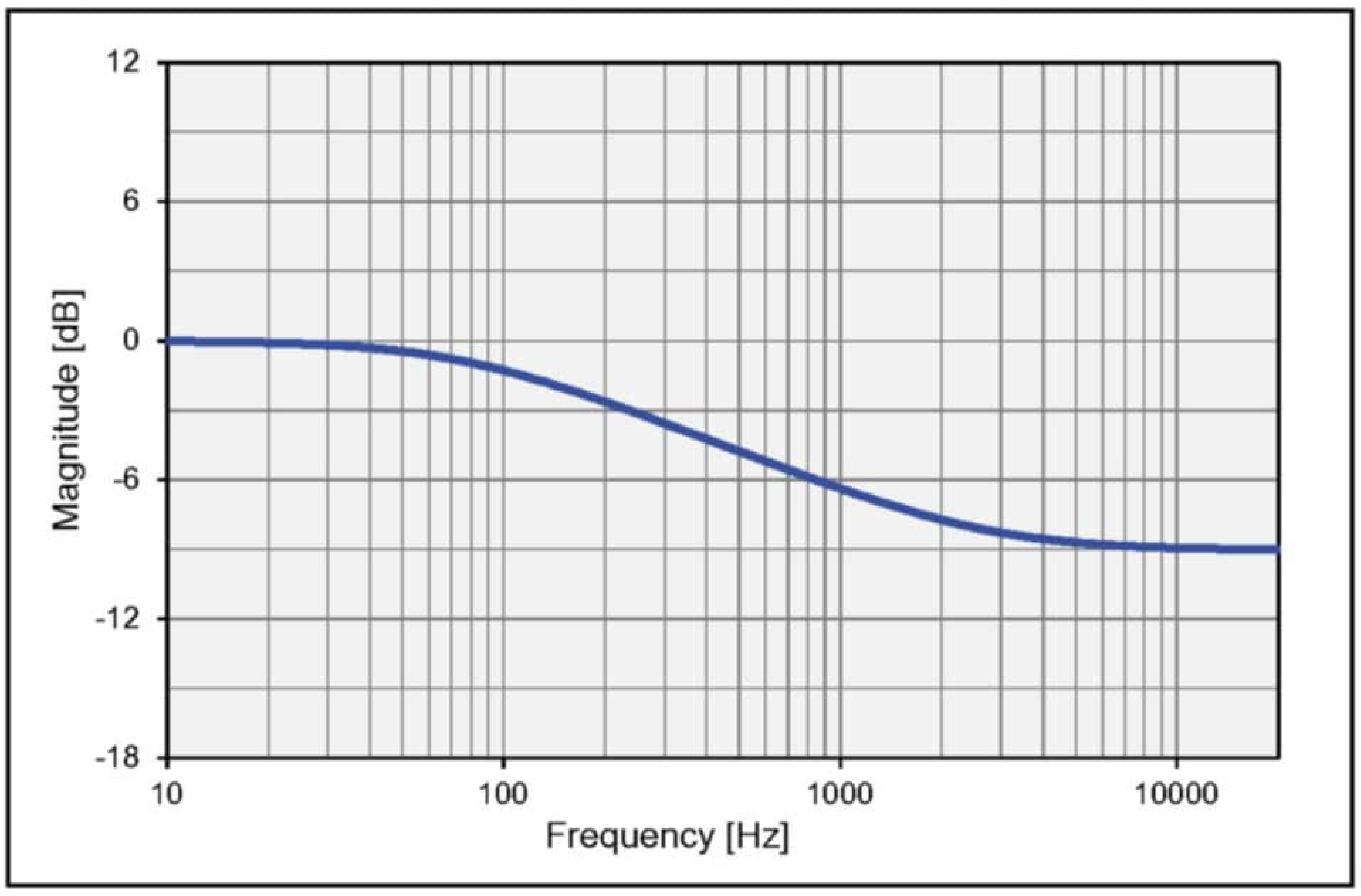


Figure 16: Response of the voicing EQ filter

Author Acknowledgements: I would like to extend my sincere thanks to Tim Mellow for sharing data arising from his investigation into baffled dipole and monopole sources. This made it possible to construct several useful figures in this article. Also, I would like to thank Andrew Jones for spurring the analysis of the sound pressure losses for a freely hanging driver.

Project Files

Additional material and files including Tables 1-5, which include a description and the required parameters for each filter type (Table 1); a list of filters applied to the tweeter band (Table 2); a list of filters applied to the midrange band (Table 3); a list of filters applied to the woofer band (Table 4); and a list of filters employed as Voicing EQ; as well as content regarding free hanging drivers, measurement methodology driver alignment, and voicing the dipole loudspeaker have been included as Supplementary Material for this article series. To download the content, visit: http://audioxpress.com/page/audioXpress-Supplementary-Material.html

References

[9] J. Bagby, "How to Use Passive Crossover Designer To Find The Relative Acoustic Offset," http://audio.claub.net/tutorials/Finding_Relative_Acoustic_Offset.pdf

[10] D. L. Ralph, "Finding Relative Acoustic Offsets Empirically," *Speaker Builder*, Issue One, 2000, www.speakerdesign.net/sbarticle.html

[11] S. Linkwitz, "12 dB/oct high-pass equalization" ("Linkwitz Transform," Biquad)," Linkwitz Lab, www.linkwitzlab.com/filters.htm#9

[12] F. Toole, "The Measurement and Calibration of Sound Reproducing Systems," *Journal of the Audio Engineering Society* (JAES), Volume 63, Issue 7/8, pp. 512-541, July 2015.

[13] S.E. Olive, T. Welti, and E. McMullin, "Listener Preferences for In-Room Loudspeaker and Headphone Target Responses," presented at the 135th Convention of the Audio Engineering Society, Convention paper 8994, October 2013.

[14] E. Hardison, "Various loudspeaker measurements," Erin's Audio Corner, http:// erinsaudiocorner.com

[15] L. L. Beranek, *Acoustics*, McGraw-Hill, New York, 1954.

[16] D. Self, *The Design of Active Crossovers*, Oxford, Focal Press, 2011.

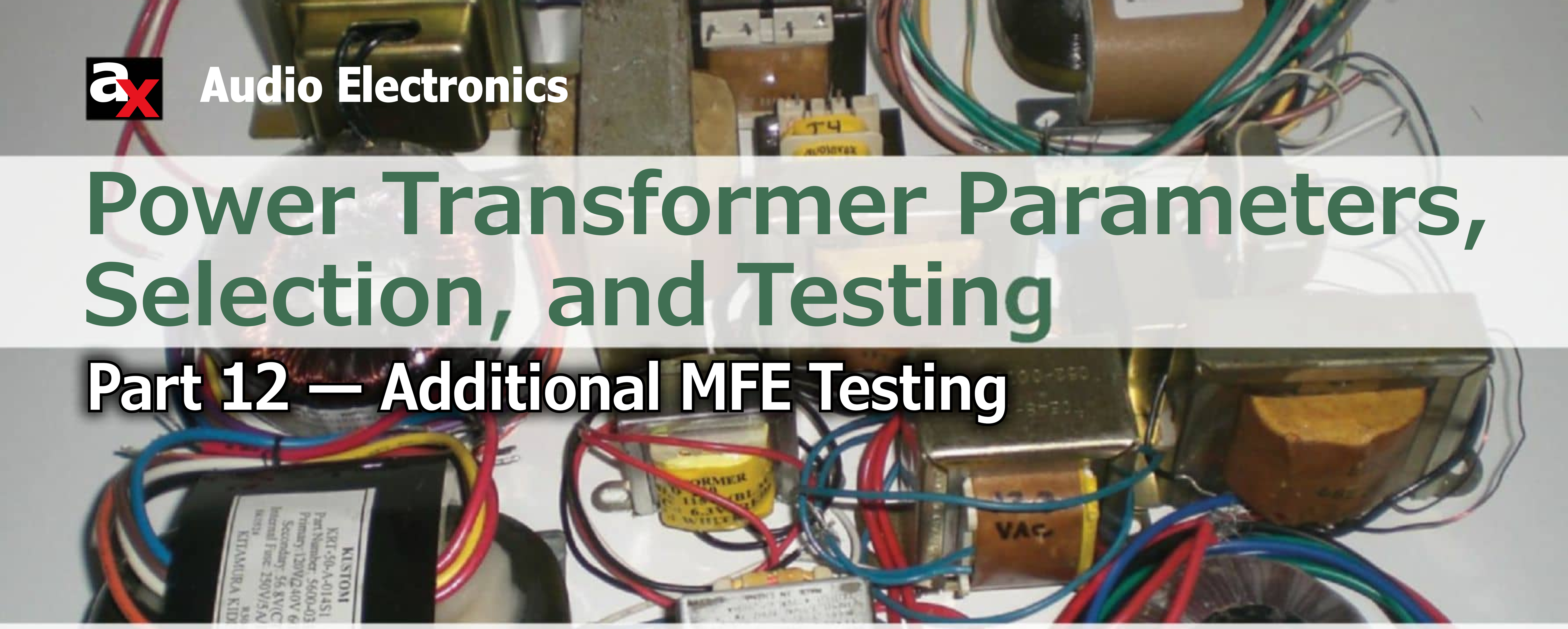
[17] "Measurements of the Peerless 830869", HiFiCompass, https://hificompass.com/en/speakers/measurements/peerless/peerless-830869

[18] C. Laub, "The Active Crossover Designer Tools," Charlie's Audio Pages, http://audio.claub.net/software/ACD/ACD.html

[19] C. Laub, "LADSPA Plugins," Charlie's Audio Pages, http://audio.claub.net/LADSPA-plugins.html

[20] GStreamer - Open Source Media Framework, https://gstreamer.freedesktop.org

[21] "Basic tutorial 10: GStreamer tools," GStreamer, https://gstreamer.freedesktop.org/documentation/tutorials/index.html?gi-language=c



Power Transformer Parameters, Selection, and Testing

Part 12 — Additional MFE Testing

In the previous articles we have detailed the history of transformer cores, their different construction methods, their materials, various testing methods and their losses, commercial power quality, the design of current transformers, additional transformer core types and their advantages and disadvantages, and conducted some testing. We finish up this series with an additional set of tests and some important considerations for audio designers.

By
Chuck Hansen

Earlier in this series of articles I mentioned the high inrush current for toroidal core transformers. I made a test setup to capture the first few cycles of inrush current for the Triad VPT48-520 just as the 115V AC primary voltage was applied at zero-crossing with the rated secondary 25VA load connected. Worst-case inrush current happens when power is applied to the primary winding just as the primary voltage crosses zero (see Figure 14 in Part 5, *audioXpress* March 2023).

To eliminate anything other than the transformer characteristic impedance, I preadjusted the remanence of the Triad core to zero by slowly increasing the primary voltage from zero to 120V AC at no load, then decreasing it slowly back to zero. I also did not use rectifiers and aluminum filter caps for any of my tests, which would introduce a lower initial secondary impedance.

I connected the fused 120V AC line for the test to the primaries of a big Technopower W5L Variac, which can produce up to 8.5A at 125V AC, (**Figure 50**) and the AnTek 0110 transformer (test data in Figure 48, *audioXpress* September 2023). The AnTek secondaries isolate the Pico scope input connector grounds, and the PC data cable that connects it to my computer, from the AC power ground, which may not be at the same potential.

Connecting the AnTek primaries in series keeps the core flux density well out of saturation to ensure accurate voltage readings. I also kept it continuously

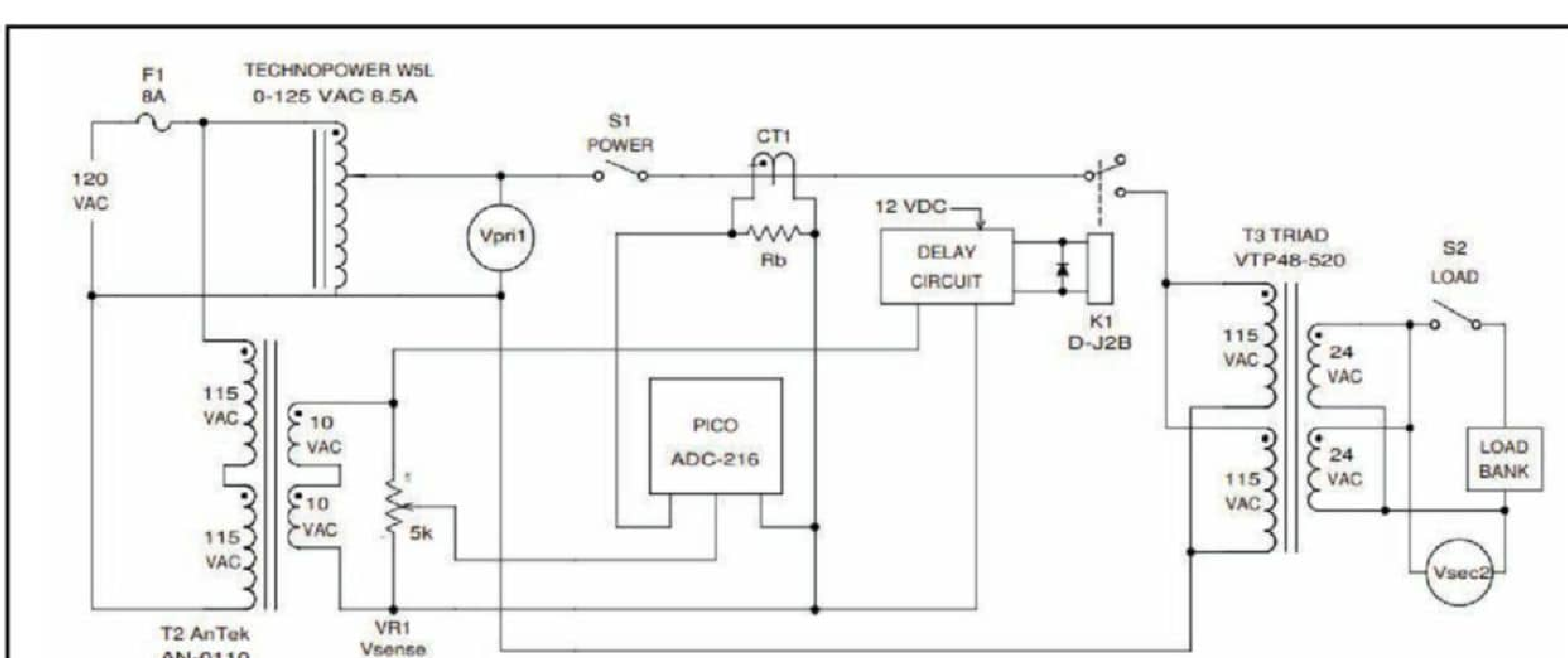


Figure 50: Inrush current tester schematic for the Triad VPT48-520 toroidal transformer

energized, so it doesn't see the initial line voltage application by the time delay relay. That eliminates its own single-cycle voltage surge, which would introduce an error in the voltage and current traces.

With switch S1 closed, I manually energized relay K1 and adjusted the Variac to the 115V AC rating of the Triad transformer primary windings in parallel at full load. I marked this setting on the Variac dial, and slowly reduced the ac voltage to zero to remove any remanence flux from the Triad core. Then I opened switch S1 and put the Variac dial back up to the Triad full load setting.

I designed an adjustable time delay circuit some time ago that detects the start of a negative-going voltage half-cycle then triggers K1, a MIL-R-6106 10 amp, DPDT relay, to connect the AC primary voltage to the Triad transformer just as the positive-going primary voltage half-cycle crosses zero. The relay itself has a nominal 4ms max closure time, which is accounted for with the delay adjustment. The delay circuit and relay are powered by a separate 12V DC AC adapter.

The nominal 10.43V AC from the series-connected AnTek secondaries goes to one input of the Pico ADC-216 virtual oscilloscope through a 5kΩ near potentiometer, to allow for adjustment of the display height for maximum peak-peak resolution without clipping above or below the display. The Pico oscilloscope has an input limit of ±20V peak (14.14 Vrms).

I monitored the current with CT1, a readily available Arduino current transformer. The Arduino CT has a 200Ω burden resistor at the coil output, and accuracy can be improved by adding another external resistor in parallel. The CT signal is connected to the other input of the Pico ADC-216.

The scope traces for the divided-down primary voltage (blue) and current (red) are shown in **Figure 51**, adjusted to keep the traces within the scope display. Converting the CT burden resistor voltage to primary amps, the single-cycle peak inrush current was 4.1 A, while the steady-state full-load peak current is 1.04A. This is why time delay fuses or circuit breakers are needed for low primary resistance toroidal core transformers (and probably R and O core transformers as well).

I noticed there is not much phase shift between current and volts shown in Figure 51. I tried it several times. I wonder if this is the second cycle, and the peak current is higher yet.

I did notice an interesting difference in the voltage and current waveforms. The waveforms are not horizontally symmetrical. The steeper side of the voltage waveform occurs as the voltage goes from positive peak to zero, while the steeper side of the current waveform occurs while the current goes from zero to peak. The voltage and current are in phase with each other, as they should be with resistive loads. I checked my CT polarity and the scope settings numerous times and can't explain this.

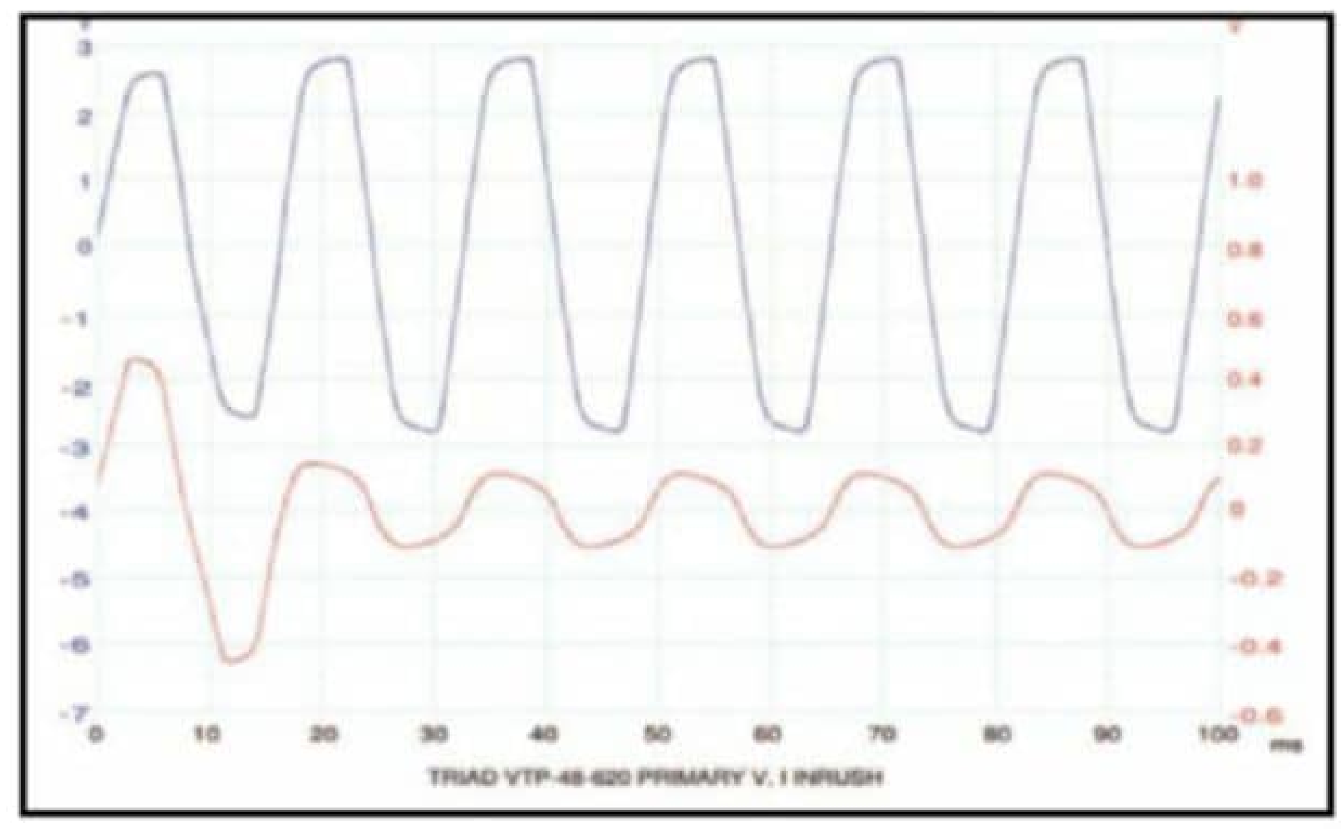


Figure 51: Worst-case inrush current for the Triad VPT48-520 toroidal core transformer

Avel D4050 Leakage Flux Density Test

The last transformer I tested was the Avel D4054, a big 330VA toroidal with two 115V AC 50Hz/60Hz primary windings and two 45V AC secondary windings. It is permanently installed on what I call a reverse-fed transformer test fixture. A very simplified schematic for the fixture is shown in **Figure 52**.

Its primary function was related to an investigation of losses using a pair of conventional E-I laminated transformers with their secondaries connected to obtain an isolated 115V AC or 120V AC at the second transformer primary winding.

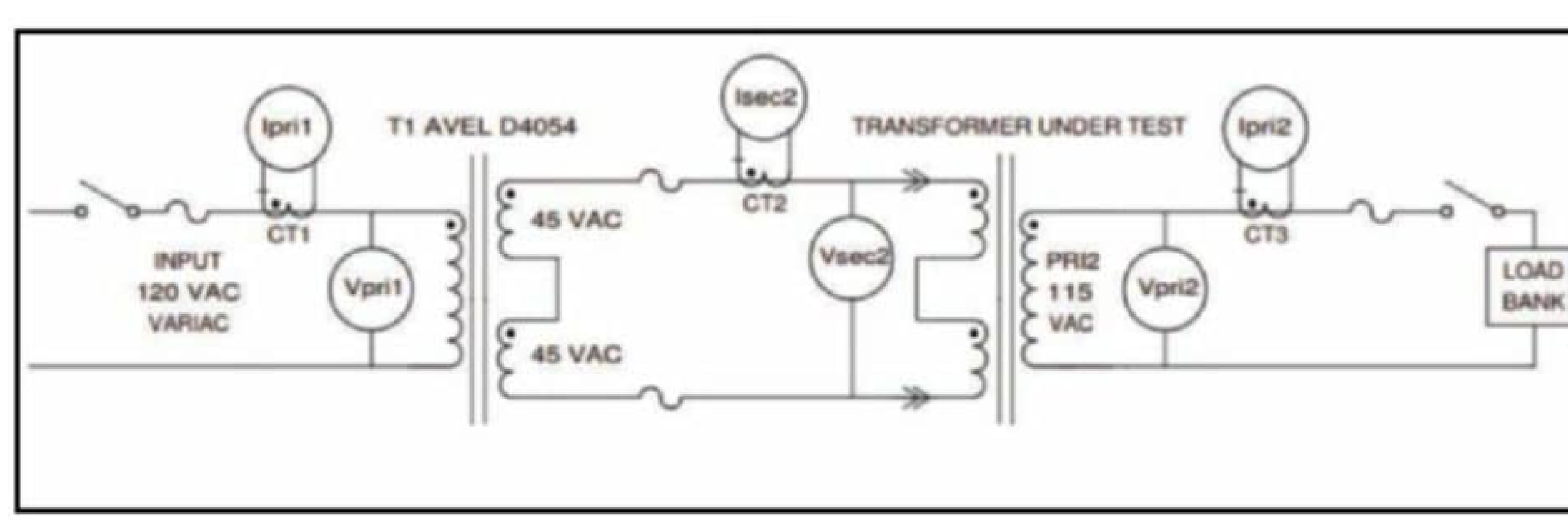


Figure 52: Reverse-fed transformer tester simplified schematic

About the Author

Chuck Hansen is an Electrical Engineer and holds five patents in his field of electric power engineering. He has worked in both the electric utility and aerospace electric power industries. He was assigned as the Instrument and Controls Startup Group Lead Engineer for two nuclear power generating units. He was the Electrical Systems and Controls Supervisor for a number of military programs as well as commercial and business jet electric power programs. He has written two books for Audio Amateur publications, and more than 260 magazine articles. He began building vacuum-tube audio equipment in college, and enjoys restoring and modifying audio equipment and test equipment. In his most recent roles as Senior Design Engineer and Lab Calibration Manager, he designed specialized test equipment particular to the aerospace electric power sector.

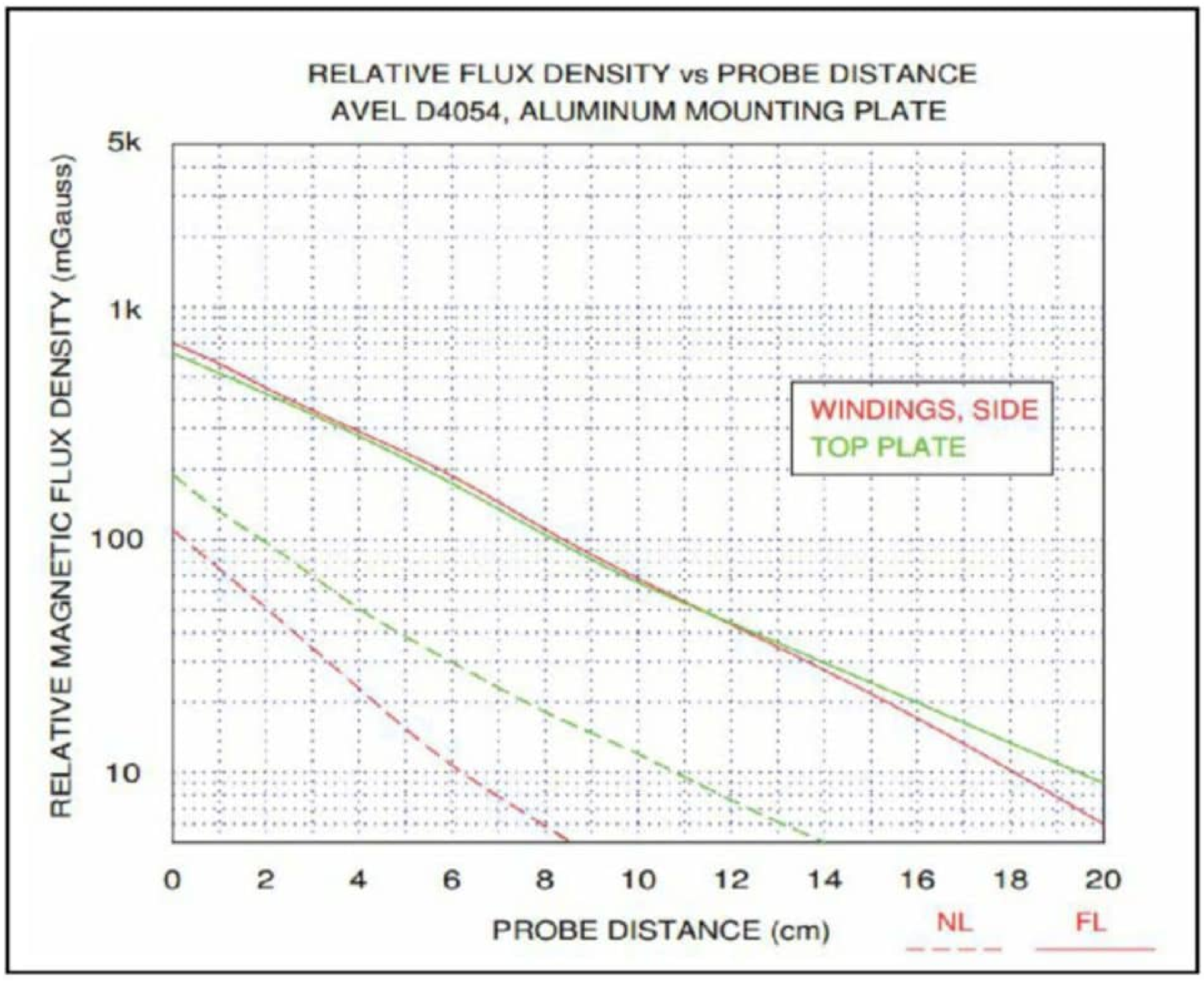


Figure 53: Leakage flux density vs. distance from MFE probe for the Avel D4054 toroidal transformer with the aluminum chassis

I also used it to measure the efficiency and voltage drop of the two reverse connected transformers.

This fixture also uses Arduino current transformers (CT-1 to CT-3) to measure the AC currents. Two rotary switches on the front panel select the three voltages and three currents of interest.

In place of the transformer under test shown in Figure 52, I just hard-wired the Avel secondary terminal block to the PRI2 primary terminal block. I also connected the primaries directly to my 120V AC test bench outlet rather than the Variac to ensure a stiff AC primary voltage and monitored the voltage throughout the tests.

I didn't have any combination of power resistors equivalent to 24.5Ω suitable for a 90V AC 330VA load, so I used 65W and 75W PAR3 and PAR4 tungsten filament lamps for my "load bank." The lamp load required to meet the 330W full load rating of the Avel D4054 secondary windings connected in series was a total of 550 W of 120V bulbs. I used five 65W bulbs and three 75W bulbs to obtain the 3.67A AC full load current.

Transformer Model	Lams	Fig.	VA	Primary VAC, Freq.	Secondary VAC, A	Wt (gm)	Mounting Plate	No Load Leakage Flux		Rated Full Load Leakage Flux		Regulation Factor*%
								Location	Pk Flux (mG)	Location	Pk Flux (mG)	
Hammond 166G36	E-1	34	18	115V AC	36 Vct	400	Aluminum	Frame Top	1,582	Windings	2,719	16.2%
	E-1	35		60Hz	0.5A		Carbon Steel	Frame Top	1,523	Windings	2,929	
RadioShack 273-1512B	E-1	40	50.4	120V AC	25.4 Vct	828	Aluminum	Frame Bottom	2,087	Frame Bottom	1,902	9.8%
	E-1	41		60Hz	2.0A		Carbon Steel	Frame Bottom	2,079	Frame Bottom	2,037	
Triad F-264U	E-1	42	120	115V AC	30 Vct	2,590	Aluminum	Lams Side	3,351	Lams Side	3,160	7.5%
	E-1	43		50Hz/60Hz	4.0A		Carbon Steel	Lams Side	3,182	Lams Side	3,540	
Kustom KRT-50-A-014S1	R	45	65	120/240V AC	58.6 Vct	1,118	Aluminum	Core/Windings	71.5	Pri Windings	388	7.2%
	R			60Hz	0.853A							
(eBay) RN-20	R	46	30	115/230V AC	2× 18V AC	528	Aluminum	Core/Windings	77.9	Pri Windings	470	31.6%
	R			60Hz	0.6A 2× 9V AC 0.45A							
AnTek AN-0110	Toroid	48	10	2× 115V AC	2× 10V AC	227	Carbon Steel	Top Plate	34.5	Windings side	329	17.2%
				50Hz/60Hz	0.5A							
Triad VPT48-520	Toroid	49	25	2× 115V AC	2× 24V AC	400	Carbon Steel	Top Plate	81	Windings side	926	11.0%
				50Hz/60Hz	0.52A							
Avel 04054	Toroid	52	330	2× 115V AC	2× 45V AC	3,300	Aluminum	Top Plate	186	Windings side	703	7.6%
				50Hz/60Hz	3.67A							

*Aluminum or Carbon Steel base does not affect Regulation Factor

Table 7: Peak flux densities for all eight tested transformers, at no load and full load

Figure 53 shows the relative leakage flux density vs. distance from the AC Magnetic Field Evaluator (MFE) probe for the Avel D4054 toroidal transformer mounted on its 16-gauge (0.05") aluminum chassis. The change in series-connected secondary voltage from no-load to full-load with a primary AC line voltage of 121.1V AC was from 95.2V AC to 88.5V AC. The regulation factor (ε) was 0.0757, or 7.57%. The peak FL leakage flux density was 703mG, measured just above the blue shrink sleeve where the four primary windings come out of the transformer (see Photo 33, *audioXpress* September 2023).

The peak leakage flux density at NL is 186mG at the center of the top plate (dashed green line). The peak leakage flux density at FL is 703mG, at the primary windings (solid red line).

The eight transformers I tested were of three different constructions, with widely different VA ratings. The peak full-load leakage flux densities for all of them at both NL load and FL are listed in **Table 7**. The highest leakage flux densities of all the transformers is 3,540mG for the E-I core Triad F-264U at full load. The two R core transformers performed well at both no load and full load, followed by the Triad and Avel toroidal transformers. The overall winner was the little 10 VA AnTek toroidal, which I also used in my Inrush Current Tester (Figure 50).

The calculated regulation factors for all the transformers are shown in the right-side column. I found that whether the mounting base was aluminum or carbon steel, it didn't affect the regulation factors.

For comparison, the usual grain-oriented silicon steel full load operating core flux density is 15kG to 18kG (1.5T to 1.8T). The ratio of the highest full-load leakage flux density to a core flux density of 1.5T for the Triad F-264U is 2.36 × 10-4. Very small indeed.

MFE Measurements for Audio/Video Equipment

Out of curiosity I checked the chassis of all the audio/video equipment I own with a ceramic permanent magnet, and they are all made of painted steel, except for my PS Audio Digital Link DAC which is made of aluminum. The front control panels are all either aluminum or plastic. I also tested one vacuum tube signal generator and two low-voltage AC plug-in adapter modules, one AC-to-DC and one AC-to-AC.

I used the MFE sensor to measure the highest leakage flux density at the outside of the chassis closest to the power transformer for most of my audio and video equipment, playing a tape or a CD of Percy Faith's orchestra playing "Jamaican Rhumba." I also measured the only vacuum tube equipment I still have, an HP 200AB audio oscillator.

First, I measured the A/V equipment that have volume controls, first with the volume turned all the way down, and then playing a bit louder than my normal listening level. Sources such as DVD and cassette deck players have a fixed output volume.

Table 8 shows the maximum leakage flux density test results for 12 audio/video and test equipment devices. A discussion of the details follows.

A/V and Test Equipment Leakage Flux Density	Probe Location	mGauss volume min.	mGauss volume loud
HP 200AB (vacuum tubes) Audio Oscillator	lower R side, rear (pwr xfmr)	673	673
Lafayette LR-5000 (ca. 1975) 35W/channel Quad Channel Receiver	R side, center (pwr xfmr)	93	101
NAD L53 50W/channel DVD Receiver	top, R side (power xfmr)	564	564
Bose TV Speaker Soundbar Speaker System	top, LF corner	177	177
Parasound HCA 1000A 125W/channel Stereo Power Amplifier	top center, front (toriod power xfmr)	93	143
NHT SA 2 Subwoofer Amplifier	top left, forward (power xfmr)	168	320
Adcom GFP-555 (modified) Preamplifier	top, left (power xfmr)	244	244
Rotel RCD-970BX CD Player	front bezel, drawer and display	51	N/A
Alesis Master Link ML-9600 Hi-Res Master Disk Recorder	L side, rear (power xfmr)	8	8
PS Audio Digital Link D/A Converter, 92, 196	R side, front (toroid power xfmr)	135	N/A
JVC TD-V662 Casette Deck	L side (power xfmr)	34	34 (headphones)
Music Hall MMF-2 Turntable	top by power switch (motor)	4	N/A

Table 8: Maximum leakage flux density test results for 12 audio/video and test equipment devices.

The HP 200AB sine wave oscillator has six vacuum tubes, and dates back to 1952. It has an output transformer that matches 600Ω loads. The peak leakage flux density of 673mG was measured at the lower rear, right side of the painted steel chassis, where the large power transformer is located. It can accept any power line frequency from 50Hz to 1kHz.

I bought the 35W/channel transistorized Lafayette LR 5000 four-channel receiver in 1975 when quadraphonic sound was hitting its stride. Many people found out it only took one or two playings of a quad vinyl disc to scrub away the rear-channel modulations, even with the special recommended phono cartridge. The peak leakage flux densities were 93mG at zero volume and 101mG playing louder than I would prefer. The power transformer is located near the right side of the steel chassis.

The NAD L53 DVD Receiver was from one of the Product Review projects Ed Dell asked Gary Galo and me to do for *audioXpress* in 2005-2006 [8,9]. The peak leakage flux density of 564mG was measured at the top, right side of the steel chassis. This is again where the power transformer is located.

The Bose TV Speaker soundbar unit sits in front of my wife Kathy's TV. The power cord is on the left side, and the peak leakage flux density of 177mG was measured at the top of the left front corner.

The Parasound HCA 1000A Stereo Power Amplifier was another Product Review, for the June 2001 issue of *audioXpress* [10]. The peak leakage flux densities of 93mG and 143mG were measured just above the center of the big toroidal power transformer, visible through the ventilation slots at the front of the amplifier. I had an enjoyable discussion of the design of the HCA 1000A with John Curl after the article being published.

The NHT SA 2 subwoofer amplifier was part of the NHT speaker system I bought, along with the SW2P subwoofer and a pair of SuperOne satellites. The peak leakage flux densities of 165mG and 320mG were measured at the top left of the steel chassis, just above the power transformer. G. R. Koonce and I wrote a design article for passive subwoofer crossovers in the Eight:1999 issue of *Speaker Builder* [11].

I modified this Adcom GFP-555 preamp for my particular audio system requirements, after being inspired by the modifications that Gary Galo made to his GFP-565 in *audioXpress* issues November 2003, December 2003, and January 2004.

My article about the preamp was published in the September 2006 issue of *audioXpress* [12]. The peak leakage flux density of 244mG was measured at the left side of the steel chassis, just above the

References

[8] C. Hansen, "NAD L53 DVD/CD Stereo Receiver," *audioXpress*, December 2005, pp. 38-53.

[9] G. Galo, "Listening Review, NAD L53 Receiver," *audioXpress*, March 2006, pp. 50-53.

[10] C. Hansen, D. MacArthur, and M. MacArthur, "Parasound HCA-1000A Power Amplifier," *audioXpress*, June 2001 pp. 58-61.

[11] G. R. Koonce and C. Hansen, "Mating Subs and Satellites via Passive Crossovers," *Speaker Builder*, Eight:99, pp. 38-51.

[12] C. Hansen, "Adcom GFP-555 Preamp Upgrade," *audioXpress*, September 2006, pp. 44-51.

[13] C. Hansen, "Upgrade Rotel's 970BX CD Player," *audioXpress*, February 2003, pp.26-31.

[14] G. Galo, "PS Audio Digital Link III," *audioXpress*, August 2009, pp. 36-40.

[15] C. Hansen, "PS Audio DL III DAC Measurements," *audioXpress*, August 2009, pp. 40-44.

[16] C. Hansen, *A Brief History of Bendix Red Bank Tubes*, pg. 21, reference 4, "Why 400 Cycles," Audio Amateur Press in 2004, ISBN 1-882580-50-8 (This was from Chief Rotary Engineer Fred Potter's notes in his History of Eatontown Division of Bendix Aviation Corp.)

Resources

AK Electrical Steel, www.aksteel.com

Arnold Engineering Soft Magnetics Application Guide – (out of publication)

Carpenter Technology, www.carpentertechnology.com

Hitachi Metglas Amorphous Metals, https://metglas.com

Magnetic Inc., *Design Manual for Tape Wound Cores,* www.mag-inc.com

Magnetic Metals Company Tape Wound Core Design Manual, www.magneticmetals.com

"Soft Magnetic Nanocomposites for High-Temperature Applications," Tech Briefs, November 2020, www.techbriefs.com

power transformer. I replaced the original E-I transformer with one that had a higher VA rating. It is enclosed within a steel shell, similar to the RadioShack transformer I tested for this article.

The Rotel RCD-970BX CD player was another modification article, for the February 2003 issue of *audioXpress* [13]. The peak leakage flux density of 51mG was measured at the front bezel, drawer and display while the unit was playing, and a slightly lower reading was measured at the left rear where the toroidal power transformer is located.

I used the Alesis Master Link ML-9600 High-Resolution Master Disk Recorder to compile digital audio test files that weren't available elsewhere. The peak leakage flux density of only 8.1mG was measured at the left rear side of the steel chassis, right adjacent to the power transformer.

The PS Audio Digital Link 96/192 D/A Converter review was another collaboration between Gary Galo and myself for the August 2009 issue of *audioXpress* [14,15].

The peak leakage flux density of 135mG was measured at the bottom center of the chassis, about three-quarters of the way toward the rear panel, where the toroidal power transformer is located. The top of the unit is made of aluminum sheet metal, and it overlaps the sides of the sheet steel bottom.

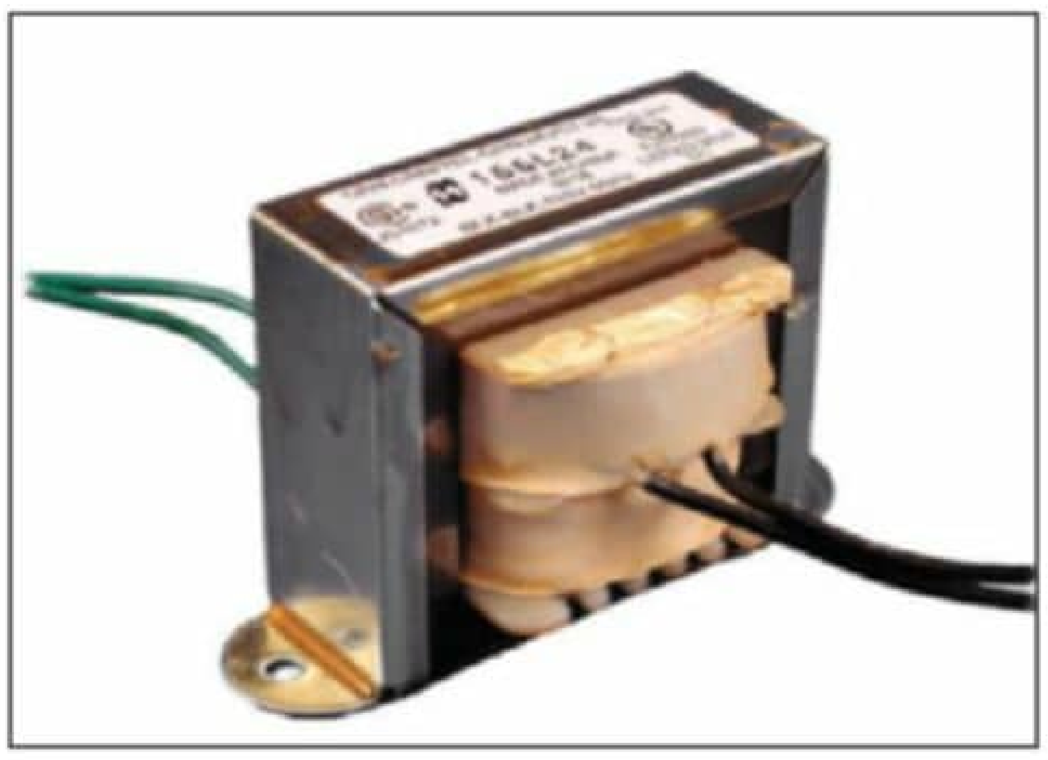
The JVC TD-V662 Cassette Deck had a peak leakage flux density of 34mG, measured at the left side adjacent to the power transformer.

Finally, the Music Hall MMF-2 Turntable had the lowest peak leakage flux density, 4.2mG, measured at the power switch on the lower left edge of the plinth. The motor is located just below it. The leakage flux density around the turntable and tonearm was too low to measure.

MFE Measurements for Plug-In AC Adapters

I also tested two 115V AC 60Hz plug-in AC adapters at no load and rated load. I made the measurements at the front of the plastic cases, directly in line with the step-down power transformer inside, and the rectifiers and filter cap for the AC-DC adapter. The energized AC outlet by itself produced no measurable leakage flux.

The AC adapter for my Pico ADC-216 digital scope is a StonTronics, Ltd. rated for 12V DC, 400mA. The NL flux density was 2,163mG and the FL flux density was 1,911mG.



The other AC adapter, for a Nintendo video game, is rated for 9V AC, 1.9A. The NL flux density was 1,784mG and the FL flux density was 1,843mG.

These small transformers have high leakage flux densities, considering I wasn't even in contact with the iron core. They sometimes have a fusible link in the primary winding to prevent a fire in case of a short circuit. I took one 15V DC, 400mA AC adapter apart that had a couple of screws holding the case halves together. It revealed a no-frills bare frame E-I split-bobbin transformer with the E laminations stacked on one end, the I laminations on the other end, and the coil in between. The mating edges of the I and E laminations were welded together. It uses the same construction method as the doorbell transformer I mentioned in Part 3, Photo 11 (*audioXpress* January 2023). The AC voltage was full-wave rectified by a 600V, 1.5A W06M full-wave rectifier and filtered with a Ricon 1,000µF, 35 V, +85°C aluminum cap.

I use these AC adapters when I want to keep the transformer outside my project chassis when I design sensitive circuits, such as my ×10 gain amplifier for this project. While the no-load leakage flux density was 1,658mG at the 15V AC 400mA adapter plastic housing, it was only 0.8mG at the end of the 5′ long cable from the plug-in transformer.

I made the ±15V DC for the gain amp from a 15V AC adapter using a simple voltage doubler rectifier-filter circuit. The filter capacitors must be twice the capacitance because the DC ripple frequency is 60Hz rather than 120Hz with a center-tapped transformer winding. I followed this raw DC with positive and negative IC voltage regulators with the specified values of tantalum and film capacitors.

I'm going to take another look at Ed Simon's March 2020 *audioXpress* article, "Powering Your Circuits," and Frans de Wit's article in the January 2022 issue of *audioXpress*, "Thinking about DC Power Supplies," for more ideas to reduce electromagnetic interference (EMI).

Capacitive Coupling

At this point I figured my article was complete, but then Gary Galo reminded me of an article in the April 1995 issue of Ed Dell's *The Audio Amateur* (TAA), specifically page 36. Rick Miller performed bandwidth measurements out to 200kHz on an Avel-Lindberg D3022 toroidal transformer (now Avel D4022), and a Magnetek (now Triad) FD7-36 split-bobbin E-I transformer. His results were plotted with his Audio Precision System One and showed his D3022 toroidal bandwidth wasn't down 3dB until just over 100kHz. By contrast, the 3dB point for the FD7-36 EI transformer was about 4kHz.

I have witnessed a lot of tests on aerospace electric power systems in our large EMI screen room, where RF test signals in some EMI test categories begin at 15kHz. The unit under test (UUT) is matched to the 50Ω RF test signal measurement equipment (receivers), RF antennas and RF current probes, using line impedance stabilization networks (LISN). For the power levels we generate, we also use a power combiner to reduce the fundamental signal level to the measurement equipment.

While I have no knowledge of the test setup details or the source voltage and impedance characteristics of Rick's test signal generator, Gary's question to me concerned the possibility that capacitive coupling between the transformer primary and secondary windings played a factor in the different bandwidths Rick found between the toroidal and E-I transformers.

So, that sent me back to the test bench to measure the primary to secondary capacitances of all eight of my test transformers with my Keysight U1733C XLCR meter set for 120Hz, to see if there was a discernable pattern.

I connected one primary lead to one meter input, and one secondary lead to the other input. If there were two primary windings that allow either 115V AC or 230V AC operation, I connected them in series (with one exception, the Avel D4054, which I will explain). If there were two or more secondaries, I also connected them in series.

The odd-ball of the eight transformers was the RN-20 R core (see Photo 31, *audioXpress*, September 2023). It has that center-tapped primary winding. If you connect it for 115V AC, be aware that the third red wire is going to have 230V AC on it. If connected for 230V AC, the yellow wire will have 115V AC on it.

The RN-20 has four independent secondaries, two rated 18V AC and two rated 9V AC. I had to add polarity marks to the four secondaries to ensure they would not be connected out of phase. And there is that screen (SCN) lead that probably required the SCN winding to be wound full length over both core halves, between the primary and secondary windings. I can see looking down one end of the RN-20, at least one of the secondary lead pair (white-white 18V AC) shows a third white wire at the other end under the outer insulation, connecting both core halves. I assume the other secondaries are wound this way as well.

One of the tests I tried on the RN-20 was to see if the capacitance between the primary and secondaries dropped when I connect SCN to the guard input of my XLCR meter. On to the test results.

The Hammond 166G36 E-I core transformer has a split bobbin, which will reduce the capacitance between the primary and secondary. I measured 36.0pF.

The RadioShack 273-1512B E-I core transformer also has a split bobbin. I measured 41.3pF.

The Triad F-264U E-I core transformer is layer wound with the primary closest to the core. I measured 308pF.

The Kustom KRT-50-A-014S1 R Core transformer has dual primary windings, which I connected in series. Then I measured the capacitance from each primary lead to both ends of the center-tapped secondary winding. I measured 96pF and 102pF.

The AnTek 0110 toroidal transformer has two primaries and two secondaries. I connected each winding set in series. I measured 161pF.

The Triad VTP48-520 toroidal transformer has two primaries and two secondaries. I connected each winding set in series. I measured 340pF.

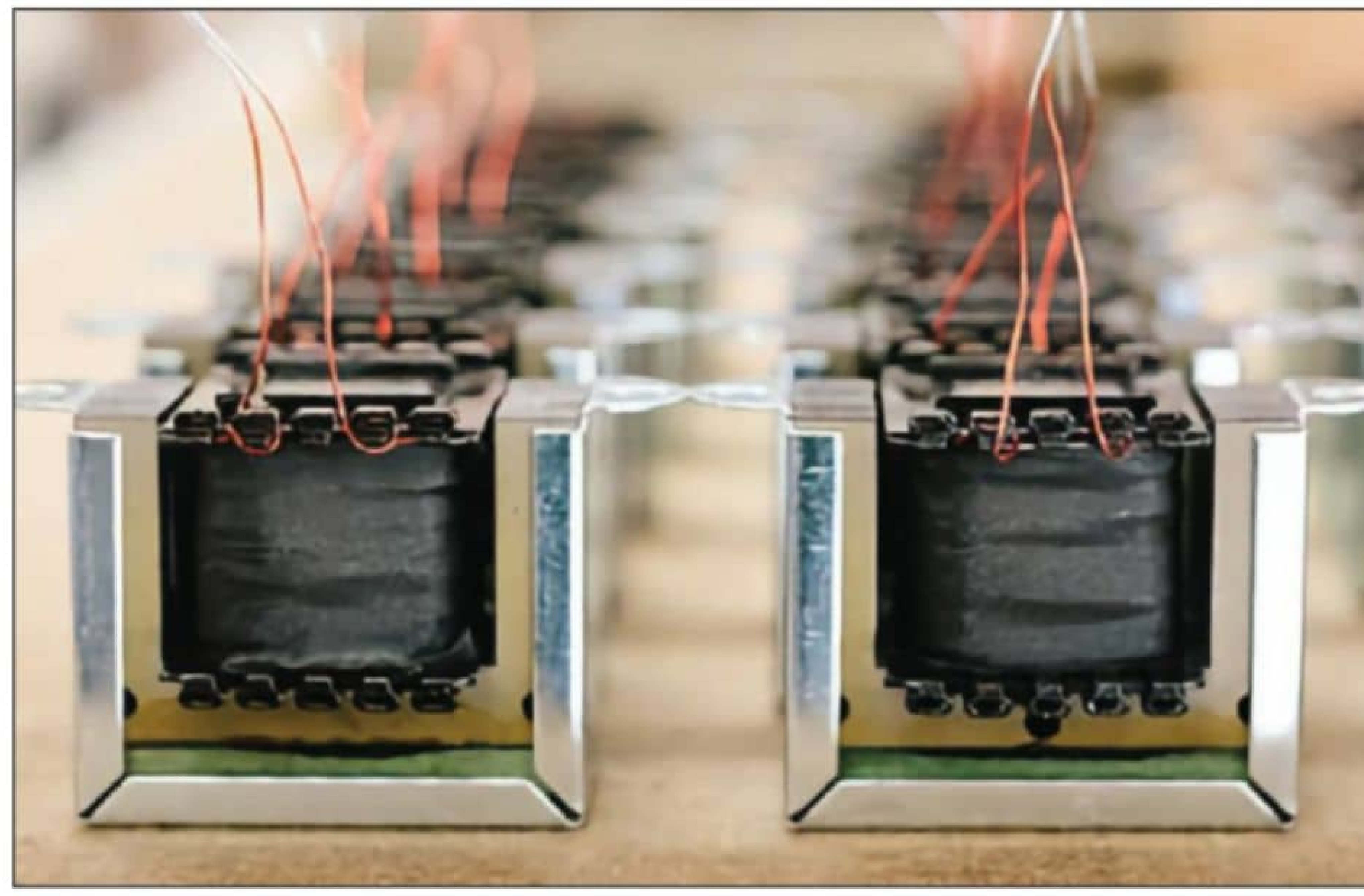
The big Avel D4054 has two primaries and two secondaries. As part of my transformer test fixture, the primaries were connected in parallel, and the secondaries were connected in series. While I could isolate the secondary winding leads by removing the two fuses (Figure 52), there was a lot of twisted-pair wiring under the chassis in series with the primaries that I did not want to disturb.

I connected the XLCR meter to one end of the secondaries, and to one end of the primaries at the chassis-mounted male AC connector. I measured capacitance at all three blades of the connector: AC hot, ground, and AC neutral. The ground is connected to the aluminum chassis of the test fixture.

The secondary to AC hot or AC neutral blades (opposite ends of the primary) measured 456pF. The secondary to ground blade measured 278pF. That makes some sense because the secondary was the outside winding of the D4054 which was mounted on the aluminum chassis.

I left the RN-20 R core transformer for last since I had to measure all combinations of connections between the tapped primary and the four secondaries. I used the black common primary wire for one XLCR connection. I used the following secondary connections for the other:

- Primary to each 18V secondary winding: gry-gry = 70pF and wht-wht = 72pF
- Primary to each 9V secondary winding: grn-grn = 63pF and brn-brn = 64pF
- Primary to both 18V in series: = 94pF
- Primary to both 9V in series: = 85pF
- Primary to all four secondaries in series: = 88pF



Clearly there is not a simple addition of capacitance going on here. The primary to all four secondaries (88pF) is lower than just the two 18V secondaries in series (94pF).

The final test was to measure the primary to all four RN-20 secondaries in series, but with the yel-grn screen lead connected to the XLCR meter guard terminal. It dropped from 88pF to 67pF. That's a 24% reduction.

The Hammond 166G36 and RadioShack 273-1512B E-I core transformers were both bobbin wound, and they had the lowest primary to secondary capacitance of all the tested transformers. That compares favorably with Rick Miller's bandwidth measurements. However, the layer wound Triad F-264U E-I transformer's 308pF was 8.6 and 7.6 times higher, respectively, than the Hammond and RS transformers.

The highest Kustom KRT-50-A-014S1 R core capacitance was 102pF, and the highest RN-20 R core capacitance was 94pF. R core transformers wound with the primaries on one bobbin and the secondaries on the other will have lower capacitance than if half the windings are wound on each bobbin. There may be some fringe flux cancellation between the bobbins during operation, but I don't think that matters for a capacitance measurement.

Finally, the AnTek 0110 toroid measured 161pF, the Triad VPT48-520 toroid measured 340pF and the Avel D4054 toroid measured 456pF. These transformers had the highest capacitance between windings. From Part 3, we know the MIL-PRF-27 Pico 88660 toroid primary-secondary capacitance is 473pF.

My conclusion is that it is not the core type that results in higher capacitance, but the way the transformer is wound. Layer wound E-I cores will have higher capacitance than split bobbin wound E-I cores, or transformers wound with the two windings on separate legs of a C core. We have occasionally wound toroidal core transformers without overlapping the primary and secondary windings to produce lower capacitance. Using copper screen windings between the primary and secondary also works.

Another Thing, Where did 400Hz Come From?

Bendix Chief Rotary Engineer Fred Potter explained where the aerospace 400 cycles (Hz) AC frequency came from in my book *A Brief History of Bendix Red Bank Tubes*, published by Audio Amateur Press in 2004 [16].

Colonel Ted Holiday, USAAF, commissioned General Electric to develop a 115V AC threephase AC generator for the Convair B-36 Peacemaker bomber program that began in 1943. It entered service in 1948. The electrical loads planned for the B-36 radars, the six remote-controlled 20mm gun turrets, and other loads far exceeded the capabilities of the 28V DC starter-generators then in use. The auxiliary gearboxes on the six 28-cylinder Pratt &

Transformer Wire Lead Color-Codes

Here we get into what is required by the national electrical code (NEC), what is required by the MIL-Specs and what is subject to the whim of the transformer designer.

For transformers whose primaries are connected to 120V AC 60 Hz power through attached self-leads, such as door-bell transformers or step-down transformers that are externally mounted on HVAC equipment, the primary wires must conform to the National Electric Code (NEC). The hot wire must be black, and the neutral wire must be white. If a ground wire is provided it must be green. The white wire must connect to the primary winding end closest to the core to minimize the capacitance from the power line to the transformer case and equipment chassis.

For transformer primaries connected to two-phase 240V AC 60Hz, a red wire must be added to represent the other hot phase wire. There are other NEC color requirements for three-phase transformer primaries, such as brown, orange, and yellow for the three 480V AC phases.

The secondary wires colors are not as stringent, unless they are found in control transformers in 120V AC, 208V AC, 240V AC, and/or 480V AC 60Hz circuits. Otherwise, transformer secondaries cannot use any of the reserved primary winding colors.

Information on international electric power wire color requirements is available at the Electronics Hub website (www.electronicshub.org/electrical-wiring-color-codes)

There are fewer wire color requirements for the internal wiring colors of aerospace electronics equipment. The interconnect wiring in aircraft is frequently all white with printed source and destination codes rather than color codes.

Back when I started at Bendix, we were using a white rubber insulated tinned solid copper wire for breadboards. It had one advantage in that you could easily strip the wire with your fingernails. The big disadvantage was that the rubber melted at a much lower temperature than PVC and, if two wires were too close together, they could short out during soldering or high ambient temperature testing. It also became brittle at low temperature. The aerospace temperature range for electronic equipment can be as wide as -55°C to +125°C.

PVC wire was an improvement over the rubber, but it still melted and shrunk (melt-back) if you weren't very careful during soldering. Any accidental contact with the solder iron along the insulation resulted in a black blob of smoking PVC.

The next improvement in hook-up wire was irradiated PVC, which is made from extruded PVC insulation that has been exposed to controlled levels of radiation. This improves abrasion and cut-through resistance and increases the maximum temperature from 80°C to 105°C.

For many decades we have used MIL-W-16878/4 Type E, 600 V-rated silver-plated stranded-copper Teflon insulated PTFE hook-up and lead wire in aerospace equipment. It has a maximum temperature rating of 200°C. Unlike PVC, Teflon wire will not melt-back during soldering. It also makes excellent lead wires for our transformers.

Teflon is also available in wire gauges from AWG-10 down to AWG-32, in the same basic 10 colors as the resistor color code. The MIL version uses solid colors or solid with a single contrasting color stripe. MIL ribbon wire is also available in Teflon.

Teflon wire is also available in commercial versions with up to four different color/sizes of striping on individual wires. The only down-side of Teflon wire is the extrusion process makes the wire very stiff and individual wires won't hold bends as well as PVC. Using it in harnesses is less of a problem since you can pre-form multiple wires into stiff bundles that better hold shape.

All copper wire needs plating or solder tinning to enhance solderability. Teflon wire is silver-plating because at the high temperature used for extruding the Teflon insulation, tin-plating or solder-tinning would melt the copper strands into a solid tinned wire. The Teflon extruding process is expensive, but Teflon will withstand any aerospace environment without degrading. It is readily available online, so I use it for all my own electronic projects.

Any part of the silver-plated conductor that is left extended beyond the insulation is subject to a black corrosion called silver sulfide (Ag_2S), the same tarnishing that occurs on kitchen silverware. It is caused by the reaction of the silver with any sulfur compounds in the air, in particular hydrogen sulfide (H_2S). The tarnished portion of the wire will not readily accept solder, so it needs to be trimmed off and discarded.

Whitney R-4360 “Wasp Major” radial engines turned at 6,000rpm, and the required AC load rating for each three-phase generator was 40kva.

The AC generator used slip rings and brushes for the field instead of the segmented commutator and brushes used on DC starter-generator armatures, allowing for a much longer brush life. GE used an eight-pole rotor design, so the frequency became 400 cycles.

Frequency in Hz, F=Poles × rpm/120

One Last Thing…

Consumer product companies often mention “aircraft-grade,” “military-grade,” or “MIL-Spec” grades of aluminum and steel chassis material. I’ve worked in aerospace for 48 years and have never heard of any of these terms officially used. In my opinion, the term is probably a bit of jargon to say the alloy has the same composition as one of those alloys commonly used in aircraft, even if not heat treated in the same manner.

All aluminum and steel grades have multiple specifications from many entities to cover all kinds of applications from consumer to aerospace. Aluminum is used in aerospace for everything from military airframes and control surface structures to the decorative trim around the seatback tray in an airliner.

Aluminum and steel alloys, like every other metal, are listed in the *MIL Handbook 5* (Metallic Materials Properties Development and Standardization, or MMPDS). Neither aerospace nor the military mandate any “better” alloy properties than those first described in the coding system for the classification of different types of steels developed in the 1940s by the industry organizations American Iron and Steel Institute (AISI), American Society for Testing and Materials (ASTM), and Society of Automotive Engineers (SAE). ASTM produces an aluminum selection guide called ASTM-B211.

Aluminum alloys are available with identification numbers 1*xxx* to 9*xxx*, depending on properties. The 2000 series are used for fatigue-critical applications. The 4000 series are used for investment and sand castings, or welded assemblies. Naval military and the National Electrical Manufacturers Association (NEMA) applications use the 5000 series, as do many ordinary commercial uses. The 6000 series aluminum alloys aren’t as strong as some others but have excellent corrosion resistance and welding ability. The 7000 series are the strongest alloys. Each series has multiple chemical compositions. Metallurgical engineers pick the correct alloys needed to meet their specific requirements.

For instance, aluminum-grade 5052 is used for general-purpose appliances and equipment chassis because it is easily formed without cracking when it is bent in a brake press. Holes can easily be made with chassis punches or punch presses without deforming the aluminum.

Just as important is the basic temper designation; F (as-fabricated), O (annealed), H (strain-hardened), W (solution heat-treated), and T (thermally treated). Other subsets of temper are added to the four-digit designation. For 5052 grades, H32 through H38 specify the combinations of operations used to produce the alloy. Alloy 6061 can be T0 dead-soft malleable, to full hardness T6 which is the most brittle, as well as other T combinations. Likewise, steel alloys are also given ASTM-SAE identification numbers 1xxx to 9xxx. 10xx grades of carbon steel are the most common, used for both commercial and military general-purpose equipment chassis and brackets.

Teflon insulated wire available in multiple colors to match electric codes.

In Conclusion

I hope you enjoyed the Power Transformer Parameter series as much as I enjoyed writing it for *audioXpress*. I could not have done it without the talented contributors listed here who put in the time to thoroughly review my manuscript. They made some great comments and I learned a lot about what others felt was important to add to the content. And I especially want to thank Shannon Becker for keeping things on track during the tight schedules.

Author’s Acknowledgments: I would like to thank the following people for the contributions to this article series: Vince Caroppo, generator and motor electromagnetic design engineer; Bob Cordell, audio author and frequent audioXpress *contributor; Gary Galo, audio author and frequent* audioXpress *contributor; Scott Jacobs, chief mechanical engineer; Santi Lugones, generator and transformer winding supervisor; John Schubel, electrical engineer with electric utility experience; Vineet Sharma, winding tests and test equipment calibration; and Ashok Subramanian, metallurgist.*

Editor’s Note: All audioXpress articles from 2001 to present can be found on the aX Cache, a USB drive available from www.cc-webshop.com.

The Western Electric 91E Amplifier

In January 2022, Western Electric started taking orders for its new 91E integrated amplifier, following the resumed production of the 300B triode tubes at the modern Rossville Works facility. In this article, we review the origins of this single-ended amplifier in the 91A and how it evolved to a totally new design for the 21st century, offering over 20W per channel—a first for the 300B.

By
Richard Honeycutt

In 1935, Western Electric (WE) introduced the Model 300A vacuum triode, whose intended application was described in the WE datasheet as "an audio-frequency amplifier in positions where power outputs of approximately ten watts are required at relatively low plate voltages." In 1936, WE released the 300B, which was identical to the 300A except for the location of the bayonet pin in the base, allowing for mating the tube with a different socket. WE's market for audio amplifiers of a few watts was cinemas.

The WE 91-A amplifier (**Photo 1**) and 91-B, both using 300-type triodes, were introduced in 1936 to serve that market. Since cinema speakers in the mid-1930s were typically horns, with an efficiency of ~25%, in a relatively acoustically dead small cinema, an 8W amplifier could produce an average sound level of about 81dB. In a more acoustically live room, this level would be more like 89dB. With target average sound levels for cinemas being about 80dB, this shows that a WE 91-A or 91-B would have done a good job in a small cinema. **Figure 1** shows a schematic of the Model 91 amps, with notations indicating the differences between the A and B versions. The changes consist of improvements in the power-supply filter and feedback circuits, making the 91-B quieter and adding a field-implementable modification to reduce the gain if desired.

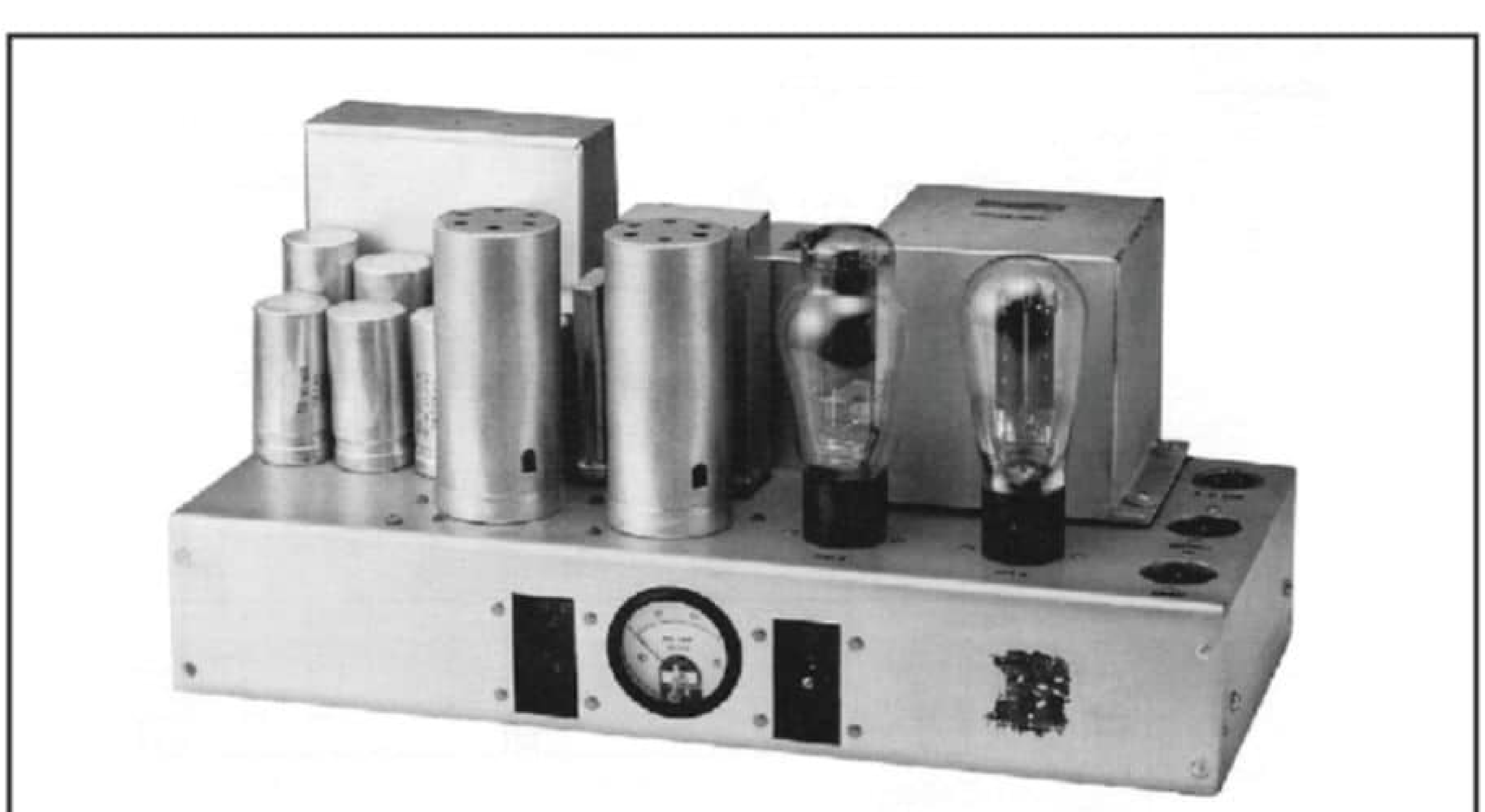


Photo 1: The introduction of the Western Electric Model 91A was a crucial step in the development of really good cinema sound.

For larger cinemas, the Model 92 push-pull amp with an output power of 20W debuted in the same year as the 91-A, providing about 4dB more power. It had a specified frequency response of 50Hz to 8,000Hz, within a tolerance of +1dB.

A New Model 91

Many audio purists prefer the single-ended (SE) triode output stage over push-pull stages or pentode stages. Thus the 91-A and -B have become among the most-sought classic amplifiers. Today, however, there are several inherent disadvantages in the original 91s, due partly to changes in the audio sources with which the amplifier is used, and partly to the rise in popularity of loudspeakers having lower efficiency and improved high-frequency response. The original 91s were designed for one purpose: cinema sound. Thus, they have one input, whereas modern listeners may want to play recordings on a phonograph, CD/DVD/BR player, FM tuner, digital music player, and perhaps even a reel-to reel or cassette tape player. Despite the arguments in the 1950s and 1960s over the preferability of stereo receivers versus separate preamps and power amps, today's listeners tend to want integrated amplifiers with plenty of inputs and a good output stage. A speaker system having a sensitivity of 85dB at 1W at 1 meter, powered by an 8W amplifier such as the 91-A, will produce a maximum peak SPL of 89dB at a listening distance of 2 meters in a typical living room. Average SPL must be 10dB to 20dB lower to avoid amplifier clipping on music that is not heavily compressed, such as good classical, jazz, or rock recordings. This can mean an average SPL near 69dB—only slightly above normal conversational level. If the amplifier power could be doubled, we would gain 3dB in maximum SPL.

Many cinema speaker systems in the 1930s used the WE tweeter designed by Bostwick. These horn tweeters extended the frequency response of a speaker system to 12kHz. Today, most decent speakers respond up to at least 15kHz. On one hand, this fact would imply that a good amplifier should have flat response at least that high. However, the authors of various research papers have concluded that flat response limits for true high-fidelity listening should extend to anywhere from 12kHz to 40kHz. The papers supporting the highest limits posited that facial cilia are responsible for human sound perception in the 20Hz to 40kHz range. The tests involved subjects' ability to perceive so-called "ultrasonic" sounds both with and without facial covering (think "ski mask").

Various means of assessing the audibility of high-frequency sound have been employed: audiologist-style perception of single-frequency tones, A-B tests involving music, and so forth. With the modern home theater experience including many sound effects whose impact is important to enjoyment of a movie, it can be argued that limiting the tests to music programs is too restrictive. However, agreement upon the importance of extended high-frequency response is not absolute. The upper frequency limit of human hearing changes with age, noise exposure history, and other factors. Tests done in the late 1960s with subjects of

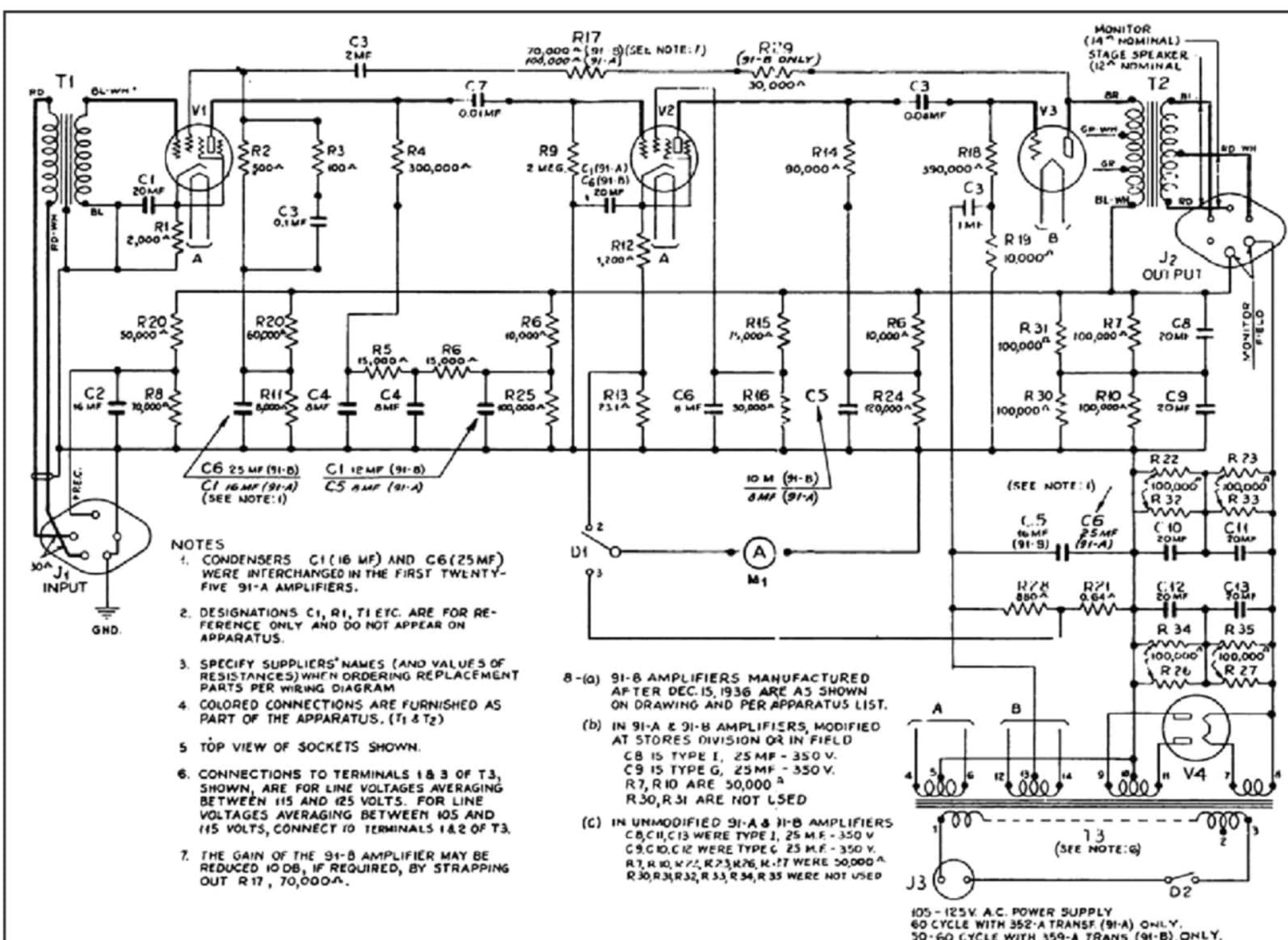


Figure 1: This schematic shows the circuitry of both the 91-A and the 91-B amplifiers.

various ages and levels of experience indicated that most people cannot tell if high frequencies are cut off above 12kHz, although for a few, the threshold was 14kHz. CDs produced according to Red Book standards have a 22.05kHz upper frequency limit. But still, there are those who advocate for sound reproducing systems with higher maximum frequency limits. But human perception aside, with the frequency limits of human hearing being generally understood to be 20Hz to 20kHz, there are certainly valid marketing reasons for a sound system to respond to that same range of frequencies.

The new Western Electric Co., under President and CEO Charles Whitener, set out to design the best possible 300B amplifier. The result was the 91E, with greatly increased output power, a stepped logarithmic attenuator, and an interchangeable output transformer (**Photo 2**). The specs list a flat frequency response from 15Hz to 32kHz +3dB (CD input) or 30Hz to 20kHz within +0.5dB of the RIAA curve (phono input), and phono, CD, tuner, two auxiliary, and Bluetooth inputs, as well as preamp and line outputs. Unlike the 91-A, the 91E is a stereo amplifier.

Twenty Watts from an Eight-Watt Tube?

The original WE datasheet lists the 300B as providing 8W maximum power output (7W at 5% THD) in a Class A circuit. The Model 92 amplifier achieved an output of 20W at 2% THD but did so by means of a push-pull circuit. To coax more power out of the tube while maintaining the Class A design, WE invented and patented the Steered Current Source (SCS) output stage design. The SCS concept begins with a parafeed concept, in which the plate of a triode used in a Class A output stage is not connected to the power supply through the primary of the output transformer as in conventional circuits, but rather is connected through a constant current source, and capacitor-coupled to the primary. Thus, the triode's direct idle current does not pass though the primary, enabling the use of an optimally designed transformer not requiring a gap in the core to prevent distortion due to the magnetization produced by the DC idle current. The current source is designed to create an effective negative resistance in parallel (AC-wise) with the 300B, allowing the circuit to produce twice the output power of the original 91-A amplifier, while keeping the output triode's operation within its safe operating area (SOA)—not exceeding maximum plate voltage, current, or power dissipation.

Photo 2: User-pluggable output transformers optimally accommodate different speaker impedances.

The actual maximum power output of the 91E is specified as 20 watts per channel (WPC) at 10% THD, 16 WPC at 5%THD, or 14 WPC at 3% THD, with either a 4Ω, 8Ω, or 16Ω load. Note that this represents a doubling of output power, giving a 3dB increase in maximum sound level at the same THD. Since the 91E uses no negative feedback, the onset of clipping is very gentle, and even at a 20W output (10% THD), the sound is remarkably clean. Rather than using a tapped output transformer, the 91E accommodates user-pluggable toroidal output transformers to provide extended frequency response. The 8Ω transformer is supplied with the amplifier, and the 4Ω or 16Ω one can be purchased as an accessory. Each is designed to perform optimally with its specified speaker impedance, having lower winding capacitance and thus better high-frequency response than a tapped output transformer.

When the 91E is powered up, a 30-second warmup cycle is initiated, during which the bias on the 300B in each channel is automatically set, eliminating concerns about balancing the performance between channels.

Unlike most audio amplifiers that use field-effect transistor (FET) based digitally controlled attenuators, the 91E uses a passive design with what would be called a logarithmic taper in a potentiometer. The stepped attenuator is built of high-quality resistors connected in a ladder configuration.

Moving-Magnet or Moving-Coil Phono Cartridge

Phono cartridges are available in both moving-magnet (MM) and moving-coil (MC) designs. Since the equivalent (Thevenin) AC electrical circuits of these differ, the required gain and the optimum

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Photo 3: The MC/MM switch and phono load jacks can be seen at the left of the 91e's back panel.

load impedance for the flattest frequency response differs in both resistance and capacitance. Since the load impedance for the cartridge is the input impedance of the amplifier's phono preamp, changing from one cartridge to another has normally involved at least purchasing a separate phono preamp. In contrast, the 91E incorporates a switchable-gain preamp and the capability to optimize the preamp's input impedance, as shown in **Photo 3**. The phono termination plugs are shown in **Photo 4** and are available as accessories that can be purchased in pairs (right/left) having values of 100Ω, 330Ω, or 1000Ω; or 100pF, 220pF, or 330pF.

Shhhh!

Noise in hollow-state amplifiers has been widely studied. It consists of four main components:

- Ionization of gas molecules in the tubes due to incomplete vacuum causes a popping noise first described as "sounding like lead shot hitting a tin roof." As vacuum pump technology has improved over the years, "shotgun noise" has become less of a problem. But carbon composition resistors also produce shotgun noise. The change to carbon film and metal film resistors has also helped cut down shotgun noise.
- Johnson noise results from the discrete nature of electron flow in any conductor. The Johnson noise voltage depends upon the resistance of the conductor, the temperature, and the bandwidth over which the noise is measured.
- Flicker noise results from impurities in a conducting channel, and its magnitude is inversely proportional to frequency.
- Power supply noise comprises electrically or electromagnetically induced noise entering the signal path, as well as hum and noise caused by poor grounding in the amplifier circuitry.

The first three types of noise can be minimized by careful attention to material purity and manufacturing process in the building of vacuum tubes.

The 91E specifications show a remarkably high signal-to-noise ratio (SNR) of 101dB(A) when the input is applied to the CD jacks, 83dB(A) for MM phono cartridges, and 73dB(A) for MC cartridges. This superior noise performance is achieved in part by the engineering and manufacturing processes used in making the ECC81 preamp and 300B output tubes. But the power-supply noise level is also minimized by optimum circuit layout and grounding practices. In addition, electromagnetically induced power-supply noise is reduced using a toroidal power transformer, which has much less magnetic leakage than a traditional EI-core transformer.

The Control Panel and Other Items

Unlike its more basic predecessor, the 91E impresses with a black, champagne, or nickel exterior, cylindrical glass shields to protect the user from touching the output tubes, pushbutton source selection, and a bright and easy to read color LCD, showing which source is selected and displaying the output level (**Photo 5**). An infrared remote control is included; it allows easy adjustment of input levels, balance, and display brightness.

For added convenience the 91E also offers several Bluetooth wireless audio codecs and profiles, including higher quality options such as AAC and aptX. Other included parts and accessories comprise a pin straightener for the 9-pin ECC81 preamp tubes, a power cord, a pre-installed 8Ω output transformer block, a Bluetooth antenna, and

Photo 4: Phono termination plugs can optimize the load impedance presented to the cartridge.

About the Author

Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built up a "junk box" by disassembling trade-in TVs from his uncle's furniture store. His dad bought him the Van Valkenburg, Neville, and Nooger Basic Electronics book set, so he learned about tube circuits as well as transistors. He started repairing electronic devices about 1965, earning his First-Class Commercial FCC license in 1969. He has worked part-time in broadcast radio engineering and audio electronics repair since 1968, in addition to full-time work in acoustical and audio system design, plus a 20-year stint teaching electronics at the college level.

a user manual, as well as two ECC8 preamp tubes and two 300B output tubes. An Ethernet connection and a USB port allow convenient software and firmware updates. Further details about the 91E can be found on Western Electric's website.

Listing to the 91E

The *audioXpress* testmaster Stuart Yaniger made some measurements on the 91E, which will be detailed in a separate review.

As most audiophiles will agree, listening to recorded music is a subjective experience, and even a panel of "golden-eared experts" will almost never concur on all aspects of a listening test. With that in mind, to assist the readers in evaluating how my listening experience with the WE 91E might compare with theirs, I'll begin with some information about the "human test instrument," myself. My years of hearing the full spectrum from 20Hz to 20kHz have passed. I lost the ability to hear the 15,750Hz horizontal deflection squeal in an analog TV in about 1988. I can now hear fairly well up to 10kHz.

My low-frequency limit—like that of most people—is difficult to nail down, but I can hear low-frequency tones from my subwoofer (whose measured response is only 3dB down at 20Hz) all the way down to where hearing passes into feeling. I do not suffer from tinnitus. I still frequently attend live music performances, so I have clear aural memories of the sounds of male and female human voices, choirs, grand pianos, pipe organs, most varieties of guitars, trap and concert drum sets, and most band and orchestral instruments. When evaluating a sound-reproducing system, these memories are my touchstone for comparison.

Amplified music is of dubious value for evaluation, since in many cases, the "original sound" existed only in the studio and at only

Photo 5: The WE 91E Integrated Amplifier is shown in black and champagne finishes. Up to six source devices can be connected, and a Bluetooth input is also available for listening convenience.

Photo 6: This is a similar configuration system for subjective evaluation of the new 91E amplifier to the one I had in my listening session. This image is from the Western Electric demo room at the 2023 High End show in Munich.

one point in time, thus giving no standard for comparison. I make no pretense of being able to compare the sound of a system with that of another system, because in the absence of ABX testing, such comparisons have been shown to be extremely subjective. (See the Hollow State articles in the August through December 2012 issues of *audioXpress* for further discussion.)

I auditioned the 91E through a pair of concept speakers using the WE 777 midrange driver (www.westernelectric.com/777), combined with the AirBlade high-frequency transducer from British manufacturer Arya Audio Labs in a three-way speaker prototype that WE intends to bring to the market in 2024 (**Photo 6**).

The furnishings and stored items in the room provided sufficient acoustical absorption so that the reverberation time was within the range expected for a good listening room. The speakers were located about 9′ in front of me and were separated by about 8′. The background noise level was about 35dBA, as measured using the NIOSH noise app. Average sound level during playback was 92.8dB(A). Given the efficiency of the speakers, the amp was only producing about 1W on average, probably peaking about 10W.

My first impression when the amp was initially powered up with a phono input selected was of silence: it was almost impossible to hear any hum or hiss. Throughout the audition, remote controlled input levels were left at the default setting, except for about a 3dB reduction after a snare drum rim shot had caused brief clipping. The clipping was very clean: no "tomato hitting a wall" splat such as would have occurred if the clipping caused momentary instability such as Barkhausen oscillation.

Listening Tests

In this section I have identified the music that I used for the audition and included my comments.

- Mozart Piano Concerto in D Major, K. 107 (MHS record 973396X)—Piano and orchestra sounds are familiar to serious music listeners.
- Wills, "The Vikings" (HDCD RR-905CD)—The orchestra and the 32′ (maybe 64′) organ pedal bass clearly convey the power of this masterpiece.
- Chesnokov, "Salvation Is Created" (HDCD RR-905CD)—The Turtle Creek Chorale's performance provides a well-recorded, superb choir sound.
- Blood, Sweat, and Tears, "And When I Die" (CBS studio album titles Blood, Sweat, and Tears)—David Clayton-Thomas' voice and every instrument supply a reference signal that is very familiar to me.
- Diana Krall, "S'wonderful"— Krall's voice gives the opportunity of evaluating how well a sound system handles the female voice.
- James Taylor, "Sun on the Moon" (James Taylor Live CD)—The strong vocals, Taylor's and those of the mixed-gender backup singers, and aggressive percussion, along with Taylor's teams' usual excellent mixing and production allow one to accurately critique a sound system's rendition of a familiar male voice.
- Linda Ronstadt, "You're no Good" (Capitol record Heart Like a Wheel)—Ronstadt's inimitable style and Peter Asher's direction/production provide another fine example of an excellent recording of the female voice with a band.

Now you know something about me, how the system was set up, the music I chose, and why. But how did the amplifier sound? I prefer to avoid adjectives, as they can be misleading: "robust bass" can mean a well-balanced bass response, a "disco bump" at 150Hz, or a poorly damped output affecting the sound at lower frequencies. "Crisp highs" can mean a well-balanced high-frequency response, a peak in the 8kHz range, or abundant harmonic distortion products above 5kHz. "A full, broad sound stage" is meaningless when discussing amplifiers, as the only way an amp can affect the sound stage is if it has poor channel separation, and it has been decades since any decent audio amplifier had that problem. Suffice it to say that I did not identify any audible artifacts that got in the way of the music. The 91E was as close to a straight wire with gain as I've ever heard.

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