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By Oliver A. Masciarotte and Kent Peterson

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RSX—A Large, Modern DIY Speaker
By Chris Biese

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The APR-17 Aperiodic Loudspeaker (Part 2)
By George Ntanavaras

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By Michael Steffes

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'Cause Everything Changes

My work with *audioXpress* magazine started seven years ago, and I still remember that one of the first lessons I learned from readers' messages and letters was that some readers were still complaining about things that happened 13 years before me. No matter the continuing efforts of the original publisher, tube enthusiasts were still expressing anger and dismay for the fact that they were "forced" to share a publication that also discussed solid-state audio circuits. Likewise, the *Speaker Builder* fans where expressing concern that their "unique" interests would fade away, submersed in a sea of meaningless magazine pages full of schematics and audio electronics.

No matter how much explaining Edward T. Dell Jr.—the publication's founder—offered, showing that in practice his quarterly titles—*Audio Amateur/Audio Electronics*, *Glass Audio*, and *Speaker Builder*—had merged into a larger, monthly publication, effectively serving more of everyone's interests. Once a niche, there's always something of that club feeling, and the DIY speaker building community still today laments the "disappearance" of *Speaker Builder*.

Unfortunately, Dell passed away in February 2013, and I never had the chance to discuss my plans for the publication with him. And how I intended to expand *audioXpress* even more from DIY into true R&D and more audio applications—which effectively was the only way to regain a much needed focus that had been somewhat lost since Dell retired in 2008.

For Dell, *Speaker Builder* was a part of *audioXpress*. For me, it remains an essential part of what *audioXpress* is today, and like our sister title *Voice Coil*—addressing the more specific needs of the loudspeaker industry professionals—I feel we have reinforced it and responded to the needs of the speaker design community.

The traditional Speaker Focus issue of *audioXpress* every September was already in place before my time in the magazine, but I don't remember any issues in these last seven years that didn't feature speaker-related articles. More recently, given the industry's level of activity in speaker components, with the dramatic expansion into microspeakers, MEMS speakers, and even completely new driver technologies, I decided to reinforce our editorial calendar with a dedicated Speaker Trends—Drivers and Transducers edition every January. Yes, *audioXpress* embraced audio applications that weren't yet recognized or even addressed by technical audio publications at all (e.g., portable speakers, smart speakers, or mobile audio). We extended the focus from living room stereo to installation and studio monitors to automotive audio speakers.

It's just a fresh way to look at speaker applications, from an innovation and product development perspective—which is what *audioXpress* magazine is about now.

But yes, *Speaker Builder* and speaker DIY is very much alive in *audioXpress*, as we show this month with three completely different projects. One of those projects, Chris Biese's RSX speakers, took first place in the Dayton Audio category at the annual Midwest Audiofest (MWAF) Speaker Design Competition, promoted by Parts Express. *audioXpress* is truly proud to offer its creator the opportunity to share the build details with our readership.

Now, as speaker enthusiasts ourselves, we are aware that our readers and many thousands more speaker builders all share their passion on a daily basis in one of the countless Internet forums and social media groups. Our authors are there, we are there. So, I'm proud that in 2013, when I joined *audioXpress*, I decided to evolve the publication's format, in order to find a way to stay useful to our original DIY community and gain an expanded readership that was then—and still is today—looking for an expanded information source on design and development of all sorts of audio products.

In this Speaker Builder | Speaker Focus edition, we include a diversity of topics that I hope answer those needs—from how to do loudspeaker measurements at home, to using modeling, simulation, and even virtual reality tools in product development and provide a better understanding acoustic quality control at the end of the production line, among other equally diverse topics.

Increasingly, I have noticed that many speaker manufacturers have approached *audioXpress* because they recognize the publication's unique role in serving their side of the industry, while I noticed that hundreds of new threads in online discussion groups have been opened with a link to an *audioXpress* article.

J. Martins
Editor-in-Chief

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COLUMNISTS

Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, a monthly publication. Although he has been involved with publishing throughout his career, he still works as an engineering consultant for a number of loudspeaker manufacturers.

Dr. Richard Honeycutt fell in love with acoustics when 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.

Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is a graduate from New York University, with post graduate work in signal processing, and he holds multiple patents licensed or assigned. For the past 35 years, Mike has worked on countless R&D projects for large and small companies. Mike specializes in materials and fabrication techniques to enhance audio performance.

Michael Steffes has worked in high-speed signal path IC development, spanning IC design roles for the BurrBrown high-speed amplifier products, and product launch/application manager roles through Texas Instruments/Intersil. He continues to focus on high-performance ADC interface solutions, DSL/PLC line driver design and optimization, and signal path design optimization tools. In a career spanning five different major IC vendors, he has participated in the introduction of 150 industry leading amplifier products (setting many of the now standard practices in high speed amplifier characterization curves), publishing more than 100 contributed articles and application notes from 1989 onward.

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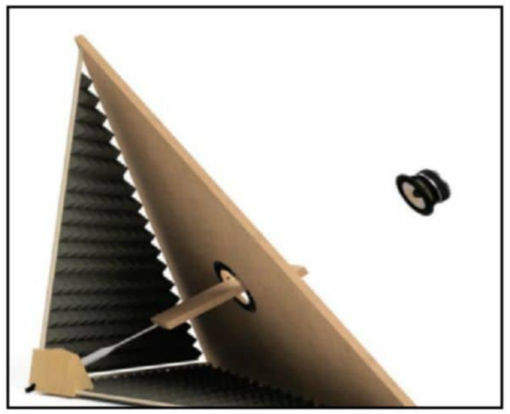

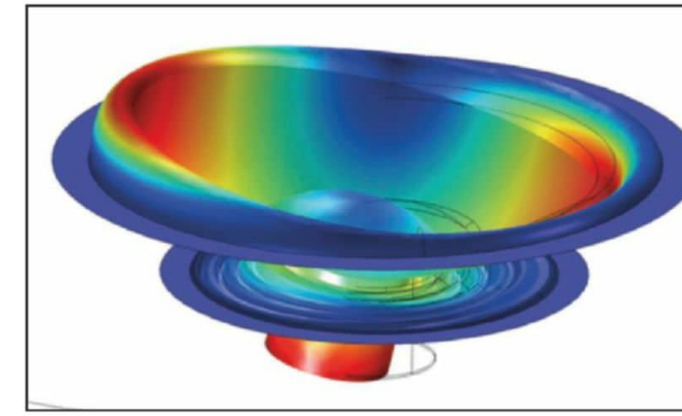

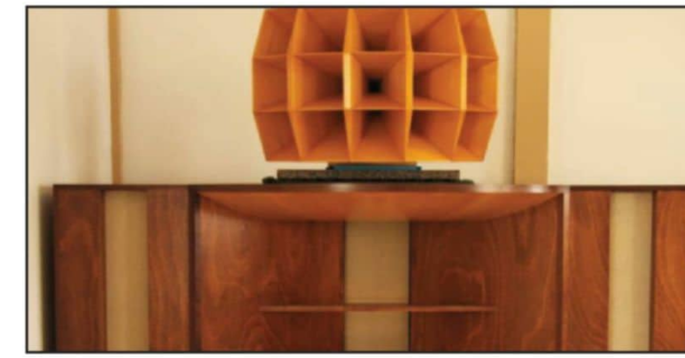

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The Magnepan LRS

Back to the Future

For those of you familiar with Magnepan, its Little Ribbon Speaker (LRS) is a full-range quasi-ribbon speaker that was created to whet your audio appetite. Intentionally designed to extract the most from high-end amplifiers and electronics, Magnepan says its LRS is "an easy introduction to their brand and house sound." Oliver Masciarotte provides his opinion of the speakers while Kent Peterson measures its performance.

Photo 1: Magnepan's Little Ribbon Speaker (LRS) is a slim and stylish alternative to entry-level MDF boxes.

By
Oliver A. Masciarotte and Kent Peterson
(United States)

Most audiophiles know of Magnepan and its iconic Magneplanar planar magnetic loudspeakers. Unfortunately, fewer people have actually heard these usually room-divider–sized panel transducers. If they did, they'd appreciate how much fidelity one can buy for so few shekels. Take, for example, the new LRS model (see **Photo 1**).

Like their progenitors, these two–ways are thin—a mere inch in thickness. Somewhat novel, at least by Maggie standards, is their size. They're diminutive little panels, with solid-wood trim available in three finishes and three choices of fabric covering. At 4′ tall by 14″ inches wide, they'd fit into pretty much any space and virtually disappear when viewed from the side. They are also "handed," supplied as a left/right pair. In my space, I ended up configuring them with tweeters on the outside. Because they're dipoles, you can't hang them on the wall. Magnepan makes the MMG for that. The LRS, or Little Ribbon Speaker, requires a good bit of breathing room. My pair ended up 42″ from the rear wall, and some 7′ from my listening position.

The Setup

They arrive in a single 45 lbs. carton, with their bent iron, L–shaped feet detached (see **Photo 2**). Eight stainless screws later, two for each foot, and Bob's your uncle. Well, almost...I had recently swapped out my old Level 3 ANTICABLES for a significantly improved set of 9-gauge Level 3.1s. At the time I ordered them, I thought I was so clever to specify bananas at the amp end and large spades at the speaker ends. Upon eyeing the terminals provided by Magnepan, I was a bit disgusted to find neither three ways nor flat screws, but a strange and unholy something else altogether—a sort of small receptacle with angled set screw, made of iron no less. Since the LRS are two–way speakers, they even come with an iron jumper (!) to a second set of sockets for bi–amping. These terminals appear to be a throwback—way, way back, to a time when one would typically grab a length of zip cord, strip the ends, and carefully stuff them into the socket. Torqueing down on the set screw with a supplied Allen key would keep the wire in place. I was not about to chop off the spades on my newly minted cables, so I inserted one blade of the spade into the socket and carefully held it in place. To prevent shorts of the "free" spade blades against the painted metal plate on which the sockets are mounted, I cut a piece of paperboard to act as insulators (see **Photo 3**). Not ideal by any stretch of the imagination, and kludgy at best.

Next up was the signal chain: Mytek Brooklyn's USB DAC balanced outs directly into the Cambridge Audio Edge W (see Resources). I didn't even try my low-power AMPX solid-state Class A since the Magnepan website recommends "…direct-coupled, Class A/B designs with high current capability… (particularly) Class A/B amplifier designs that come close to doubling power at 4 ohms." How about 200 W of Class XA as delivered by the Edge W into the LRS' 4 Ω load? In a word: excellent. Plenty of grunt and clarity to match the quasi-ribbons, which are known for their transparency. Out of the box, they sounded noticeably harsh, but very quickly broke in. Within a half hour, they were well behaved and, after a single overnight brown noise workout, they were ready to charm.

Photo 2: The legs are a simple affair, with an iron loop to provide a slight forward tilt adjustment. The loop was prone to slipping on my wood floor.

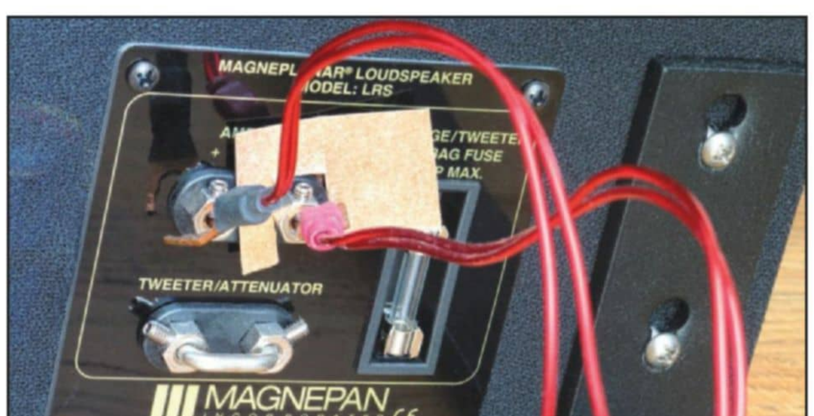


Photo 3: The input terminals are closely spaced bare metal, hence the cut paperboard. In this picture, the "tweeter attenuator" terminals and the factory iron jumper are clearly visible along with the fuse protection.

I know what you're thinking…"A $4,000 amp feeding a budget speaker? Harrumph!" Okay, it is a bit of a price mismatch, but I wanted to really hear what the speakers are capable of. For budding audiophiles, or for a second system, a modern Class D would nicely fit the bill. NuPrime's $745 STA-9 is a less than $1,000 example, and Parasound's $1,195 NewClassic 200 Integrated brings a whole lot of extra functionality to the party. Both are rated into a 4 Ω load. So, for less than $2,000, you'd only need a source component or two for a complete and highly capable system.

If you want to get all hi-fi highfalutin on me, you might want to go with one of several Purifi-based Class D monoblocks from Nord Acoustics or Apollon Audio. I had none of these on hand during this test, but I did have my inexpensive custom $300 50 W into 4 Ω Class D gathering dust, so I swapped the balanced ins and speaker leads and, guess what? Though the soundstage narrowed and clarity went by the wayside, the essence of the music was still there along with plenty of playback volume. Moral of the story? When in doubt, drive the LRS with modern pulse width modulation (PWM) technology unless you already have a quite hefty Class A/B amp lying around.

Ribbon Speakers

Let's rewind a bit and talk about true ribbons versus quasi-ribbons…First off, Maggies are passive planar magnetics, with corrugated aluminum ribbons glued to a single-layer polyester film carrier. The film is stretched taut between dual magnetic fields created by arrays of flexible permanent ferrite magnets closely related to those rubberized magnets you may have on your fridge. The magnets are affixed to perforated sheet steel, improving the rigidity and field strength. So, the metal foil stands in for a voice coil, and the film acts as a firm mechanical support in the same way as the former, spider, and surround do in a conventional driver.

In a true ribbon, the metal foil is unattached, vibrating freely in the magnetic flux. Planar magnetics from other manufacturers, typically seen these days in headphone designs, are fabricated with printed labyrinthine copper "voice coils" on a film carrier. When I first moved to the Twin Cities, I wrote an illustrated piece on Magnepan speakers detailing their construction. Suffice it to say that Magneplanar loudspeakers, regardless of the model, were the first planar magnetics of their kind and are still manufactured by hand with the aid of a few jigs. Inside the LRS is a low mid/woofer set of ribbons, and a set of narrower mid/tweeter ribbons, side by side on the Mylar. They are fed by a commodity first-order crossover.

Initial Impressions

The input terminals aren't the only aspect of the LRS' construction that I find vexing. In the packing box, you will find a set of tweeter fuses, spares for the ones already installed on the back. Remember those? Horrid little things, not conducive to full fidelity. But these a delicate speaker, in that a careless move on someone's part could damage the ribbon or, more likely, weaken the glue from overheating. But wait, there's more! The LRS also comes with two sets of power resistors. The 1 Ω and 2 Ω, 5% wire wounds are for padding down the tweeter. Let me state that there is nothing bright about these speakers. In fact, they start rolling at 400 Hz and just keep dropping at about 10 dB per octave. So, by the time you get 20 kHz, they are down by 30 dB! I know, that makes them seem as though they have no top end but, subjectively, that ain't the case. They are dark but certainly one doesn't need to drop the high end further.

So why the pads? Take a look at Kent Peterson's plots. The LRS exhibits a mid-range and a high-frequency beaminess, a prominent vertical poke in the eardrum that's all too apparent if you place them incorrectly. Hence, the tweeter pads (see **Photo 4**). Do yourself a favor—forego the questionable pads and position the speakers such that your head is below the beam. With that, you'll find the overall timbre to be subjectively neutral, if decidedly midrange-forward. Those L-shaped feet I mentioned earlier have rectangular loops of metal at their end that allow you to easily choose one of two settings for the speaker's tilt. I ended up using the tilted setting, finding it to be correct for my placement and listening position.

Overall Impressions

Being the size they are, an LRS will embrace you with a whole lot of love, but not a ton of bottom octave—50 Hz is about all you can expect. They are as bass-shy as a typical set of large bookshelves but, with or without a bit of EQ, they offer more than enough bass for everyone save the booty-obsessed. Their super-light drive elements make for an exceptionally smooth sounding and cohesive delivery. Being open-baffle dipoles, their low end couples naturally to the room, presenting an open, diffuse soundstage with solid imaging, and a top-to-bottom cohesiveness seldom heard with budget box two-ways. In my living/listening room, I routinely heard phantom images well outside the physical location of the speakers. This has not been the case with other speakers, even my open baffle Trio15 references. By the way, if you really need

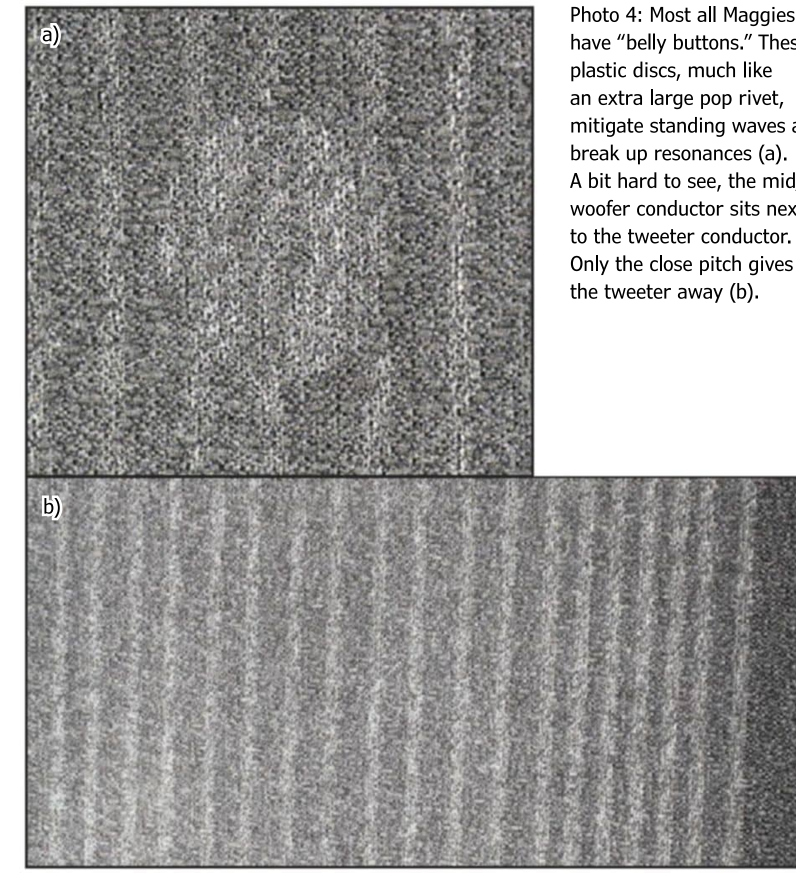


Photo 4: Most all Maggies have "belly buttons." These plastic discs, much like an extra large pop rivet, mitigate standing waves and break up resonances (a). A bit hard to see, the mid/woofer conductor sits next to the tweeter conductor. Only the close pitch gives the tweeter away (b).

About the Authors

O. A. Masciarotte has spent more than 30 years immersed in the tech space, working on facilitation, optimization, marketing, and product development for clients worldwide. As an author and speaker, he enjoys informing folks about technological best practices. More information is available at seneschal.net and othermunday.com.

Kent Peterson is sneaking up on 30 years in the Pro Audio and Acoustics field. Having designed hundreds of audio systems for professional install and approximately 100 miles of highway noise walls, Peterson now manages Warkwyn (www.warkwyn.com), where they help companies design better audio systems, represent Klippel and Microflown, while he simultaneously spreads sawdust on employee "accidents."

Resources

O. Masciarotte, "Cambridge Audio Edge W Power Amplifier: Amplification to Live With," *audioXpress*, January 2020.

O. Masciarotte, "Magnepan Incorporated—A Tiny Giant," HIFIZINE, September 2013, www.hifizine.com/2013/09/magnepan-inc-a-tiny-giant.

more bottom, employ a dipole woofer/subwoofer. It will integrate best with the open-baffle LRS.

Budget? So far, I haven't mentioned the cost of these little wonders. Would you believe $650? These are truly entry level in price but, performance-wise, there are only a handful of worthy alternatives. ELAC's Uni-fi line and MartinLogan's Motion line are front of mind, and those little boxes are lovely for the money but very different beasts. Like every other product I review, I first heard the LRS, then decided to write them up. Frankly, the Little Ribbon doesn't measure well, as Kent Peterson's companion metrics attest. If all you go by are specs, then these little droids aren't the ones you're looking for. If, however, you are careful about placement, adjusting their vertical angle relative to your listening position, you'll be rewarded with musical reproduction far in excess of expectations.

One advantage of all Magnepan products that's not intuitively obvious is their amenability to upgrades. Whether it's their crossovers, hardware, or their internal wiring and fuses, swapping out factory components addresses that tweaker need to experiment. Despite its faults of omission, no deep bass and darkish top end, Magnepan's LRS is a crazy good loudspeaker for the price. It really begs for a much more costly amplifier than you would typically pair with something this inexpensive. These little guys really do well when driven with confidence by a powerful amp. By saving a little on your speakers, you have more leeway to purchase a better quality amp and still land on a satisfying, low-cost system. Another advantage of any Maggie is the DIY upgrade community that surrounds the brand. You can drastically improve performance with a bit of pocket change and some very low-impact sweat equity.

Wendell Diller, Mister Sales at Magnepan, characterizes the LRS as an "appetizer," and low-cost entry into the world of Maggies. "The beauty is in the simplicity," he says, which is one of the reasons why all Magnepan products pack a huge amount of value into a cost-effective package. Maybe you're in the market for a second system. Possibly you're just starting out in this hobby or you know someone who is. If any of those scenarios apply, then you owe it to yourself to evaluate a pair and that's easy to do.

In the sidebar to this article, Kent Peterson shares the measurements he obtained for the LRS speaker using the Klippel Near-Field Scanning system.

EQ Is Your Friend

I listened to the LRS with and without EQ. Although a beefy amp is essential, equalization is highly recommended. I've mentioned it in the past and would like to stress that the audiophile aversion to equalization is a holdover from the "Golden Age of High Fidelity," when resonant frequencies and group delay were baked into "tone controls" and, with few exceptions, there was nothing hi–fi about the associated analog circuitry.

I hate to break the news, but times have changed. My two favorite cross–platform player apps, Amarra and Audirvana, both have EQ capabilities—one built in and the other hosting third-party products. For this review, I first pressed Amarra's in-built, minimum phase EQ into service to good effect. A tasteful low-end shelf, and a touch of high-frequency shelf to illuminate the LRS' darkness and mucho más divertida, kids!

As with the amplifier choices, I wanted to test the limits of equalization. For that, I switched over to Audirvana, which hosts my favorite surgical EQ. The $589 linear phase thEQorange is made by my company, MAAT, and is arguably the most transparent software EQ available (see **Photo 1**). So, I simply had to try it. The result, in short, was wonderful. With careful equalization, the LRS achieves a whole new level of performance, easily competing with loudspeakers costing three or more times their price. You don't need to go to such extremes but I do recommend, if you have a file-based playback chain capable of hosting better quality equalization (DMG Audio, FabFilter, MAAT, PSPaudioware, and Sonic Studio) that you give EQ a try. You just might change your mind about tone controls.

My fellow engineer and old friend Jayson Tomlin was over for a listen, and was quite surprised by the cleanliness and non-fatiguing definition offered by the equalized LRS. Listening to a hi-res audio (HRA) guitar track on Qobuz, he commented, "I can about tell where the knob is on the chorus pedal for that acoustic." In other words, subtle detail without etched harshness.

Photo 1: This EQ combines an upward low-frequency tilt with a midband scoop and a steep low-frequncy roll-off to limit excursion. Band #12 provides a 3 dB lift at higher frequencies to correct for the dark aspect I mentioned.

The Measurements By Kent Peterson

I measured Magnepan's LRS at Warkwyn's facility using the Klippel Near-Field Scanning (NFS) system (see **Photo 1**). The NFS delivers a 360° acoustical radiation balloon allowing for an examination of the radiation pattern, on/off-axis frequency response at any angle or distance, and any directionally important information that might help when determining where to place this dipole into your listening environment.

Unfortunately and right at the time of measurement, COVID-19 restrictions and work-from-home mandates were put into place, which cut short our complete evaluation of the LRS. Sadly, that meant I couldn't visit Oliver Masciarotte's cozy home for a listening session. Nevertheless, we were able to make substantial acoustical measurements on the LRS.

For the measurements of the LRS, we used 3.5 Vrms, a calibrated ACO Pacific free-field microphone with the 4048 pre-amp and a 7052E capsule was used and all on-axis data is referenced at 1 m/2.82v for sensitivity, and 3 m for some evaluations in the listening field. Measurement points around the speaker totaled 2500 and were processed with a resolution of 18 points per octave and from 20 Hz to 20 kHz. The length of the stimulus was 0.40 seconds with an averaging of 4.

As with the previously reviewed Pure Audio Project Trio15 Heil (*audioXpress*, September 2019), before examining the frequency response curves, we remind readers that the LRS is a dipole speaker configuration. With that come some pretty jagged response curves as a result of the cabinet-less design. Another factor contributing to the complex interference and summation across many frequencies is the acoustic "shortcut" or diffraction around the cabinet.

Dipolar loudspeakers function well when paired with reflective surfaces in your listening environment and sometimes cold, scientific anechoic measurements do not tell the entire story. To address that in this analysis, we have added CEA-2034 listening window curves, which better define what a user might experience in a typical listening environment. More on that later, let's dive into the raw data as it is measured anechoically.

Horizontal On/Off Axis Frequency Measurements

Examining the frequency curve on-axis shown in **Figure 1**, the LRS exhibits two prominent peaks at 55 Hz and 90 Hz, a pronounced low-mid bump between 300 Hz and 500 Hz, and a gradual roll-off to 8 kHz where the high frequency rolls off by 10 dB and into the noise floor at about 16 kHz. Summation and interference are presented as jagged peaks and valleys throughout the spectrum and are to be expected due to the dipole nature of the cabinet. Of interest are the high-frequency peaks between 10 kHz and 16 kHz in both the 15° and 30° off-axis measurements, the 8 dB increase between 1.5 and 6 kHz at 60° off-axis, and the deep notches present between 650 Hz to 1.8 kHz, 15° and 30° off-axis.

However, examining only positive degree measurements do not tell the entire story as the LRS has an asymmetrically positioned ribbon tweeter. In our measurement condition, the tweeter was positioned on the right side of the cabinet- or in the positive phi angle plane. Giving fair due, an examination of the negative phi off-axis measurement is shown in **Figure 2**.

A trip around the left side of the cabinet and at 0° theta completely fills in the holes in the mid-band present on the right side and between 500 Hz and 1.5 kHz. Even further off-axis the mid-band is accentuated. Interestingly there is a boost in the high frequency off-axis and at -60° albeit slight.

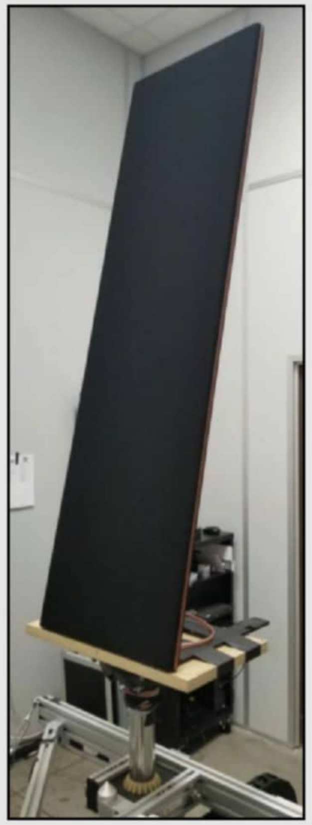


Photo 1: Magnepan's LRS is shown in its default angle for measurement using the Klippel Near-Field Scanning (NFS) system at Warkwyn's facility.

Horizontal and Vertical Contour Maps

Examining these single-frequency response curves can be a good indication of where to place this entry-level dipole, but I prefer to look at an unwrapped radiation balloon to understand the completeness of acoustical radiation at all frequencies. However, before we look at those plots, it is worthwhile to understand where the near-field of these speakers' transition into the far-field. Given

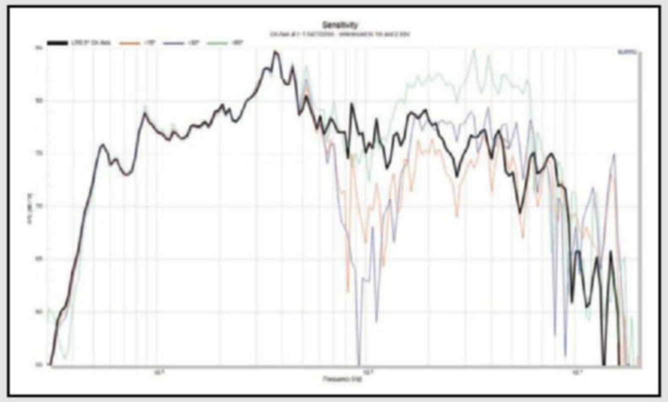


Figure 1: Horizontal sensitivity on-axis, 15°, 30° and 60° (no smoothing) in the positive axis direction

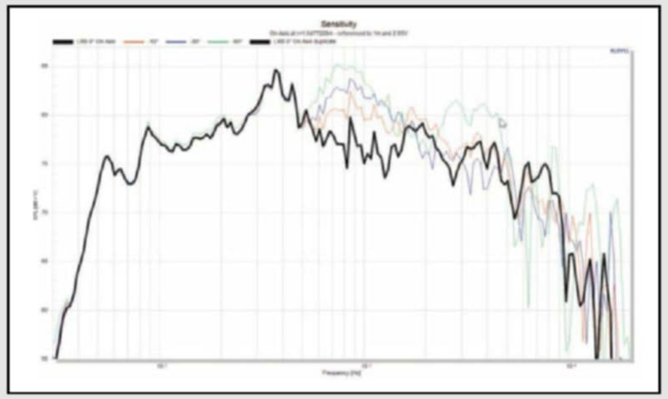


Figure 2: Horizontal sensitivity on-axis, -15°, -30° and -60° (no smoothing) in the other, negative direction

the size of the LRS (122 cm tall, 37 cm wide) and their complex radiation pattern, we wouldn't want to sit too close. Thankfully, the Klippel NFS helps us with an estimation of where the far-field should begin. I have provided an examination of that transition distance.

Figure 3 details sound power (Y) over distance (X, in meters) as the spherical harmonics increase (N) in ascending order. The near-field is characterized by a high amount of reactive power represented by high order Hankel functions. As seen in the graph, the near-field effects decrease over distance as it transitions into the far-field. After the transition, apparent sound power begins to be consistent as velocity and sound pressure become in phase with each other. This point of consistency is where the far-field begins.

For this speaker, that distance is approximately 1.5 m. Note that the use of Hankel functions in the computation could be controversial in this instance as Hankel functions describe the in/out cylindrical expansion of the radiation balloon and are attributable to more standard point source systems. If we are to take the LRS as a true plane wave or line source generator, then the far-field could be construed to be further away due to the $1/r^2$ law being gamed. That being said, and to avoid controversy, let's double the distance to the far-field to a 3 m distance.

All that setup allows us to present the unwrapped horizontal and vertical contour mapping in the far-field with a high degree of confidence at 3 m. The horizontal contour map is shown in **Figure 4**.

In conversations with Masciarotte, he suggested that I might be able to help the reader better understand these contours during this write up. I stare at these tie-died results on a daily basis, but they may be foreign to some. In the horizontal map, I have added the top of a human head. This is a simple representation of the listening position in relation to the mapping. The entire plot is an equidistant measurement, with the vertical axis (Y) representing horizontal degrees relative to the speaker's central axis. At 0° (black center line through the mapping), the listener is right on center of the speaker, with the corresponding frequency response along the black line. The plot's horizontal axis (X) represents frequency, traditionally plotted with low values at left and higher toward the right and from 20 Hz to 20 kHz. The amplitude is represented as a heat map, with a decibel scale shown to the right of the head symbol.

Remember that the unit under test was one half of a mirrored pair, with each speaker having an asymmetrically placed tweeter side-by-side in a horizontal arrangement. Notice the left–right asymmetry caused by that speaker's offset ribbon tweeter (tweeter on right side of cabinet) and illustrated here by the mismatch between the plot above the 0° axis with that below the axis.

By moving the virtual head up to +20° in the plot, you are essentially moving the virtual listener around the right of the cabinet while maintaining an equal elevation and distance from the speaker. Move the head into the negative, below the 0° axis, and you would be moving around the left side of the cabinet. At the extreme top and the bottom of the map, you are effectively looking at the rear of the cabinet (180°), and of course at 90° you are looking at the side of the cabinet. Those horizontal, dark blue areas at ±90° represent the telltale total drop out of pressure below 2 kHz, indicative of a figure-8 dipole pattern.

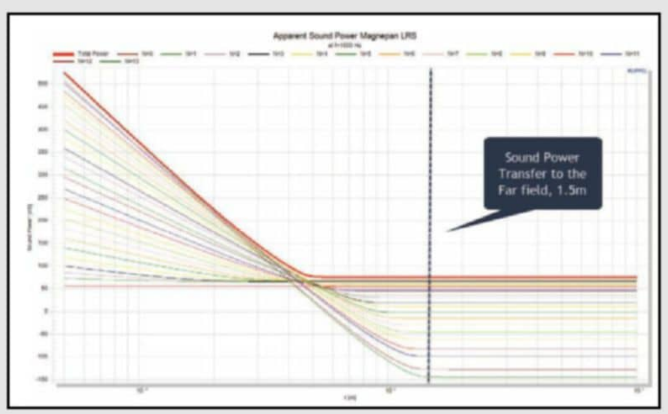


Figure 3: Apparent sound power transitioning from near- to far-field

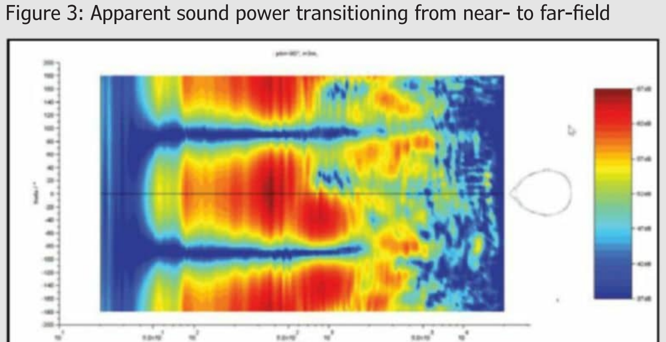


Figure 4: Unwrapped horizontal contour balloon, 3 m

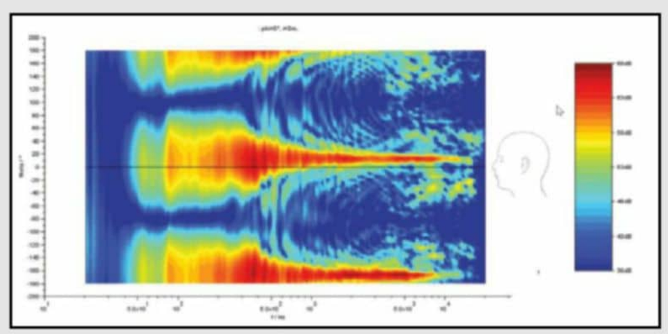


Figure 5: Unwrapped vertical contour balloon, 3 m

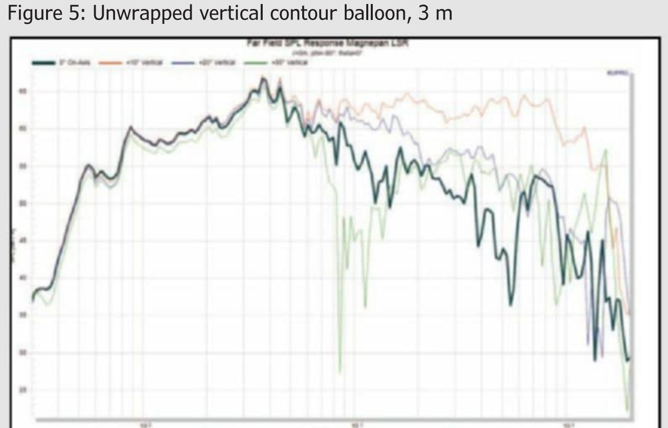


Figure 6: Vertical frequency response at 10°, 20° and 30°, 3 m

Close examination of the horizontal contour map reinforces single-curve measurements on/off axis and at ±15°, 30°, and 60°. From 0° to 30° in the + direction (up" in the map), there is a complete loss of amplitude from 600 Hz through to about 3 kHz. Interestingly, we can see that there is a recovery in pressure level from about 1.5 kHz to 7 kHz at 40° through 80°, this showed up in our 60° curve measurement in Figure 1, and is indicative of the ribbon location.

The usefulness of the unwrapped horizontal contour map shows us this increase in pressure level is good through about 80°, even approaching 90° higher in frequency.

However moving around the left side of the cabinet, negative degrees in the plot, offers a different set of challenges, namely low amplitude from 1.6 kHz and up through 40°. This had me repeatedly going back and looking at the curves in Figure 1 and Figure 2 for verification and yes—there is generally a smoother roll off of the high frequency to the left as opposed to the right though the upper ranges are down up to 10 dB from the right side of the cabinet. The hole encountered at +20° centered at 1 kHz is now present at -20°. And interestingly, the LRS seems to emit more sound in the mid and high frequencies at 90° and at almost 140° behind the cabinet. Contour maps can be your friend, especially when placement of your speaker near reflecting surfaces is so important with dipole systems.

Examination of the vertical contour map offers a good look at your vertical listening window (see **Figure 5**). As in the horizontal contour map, in the vertical contour map I've placed a human head at 0°; he's deep listening. As he "moves up," away from 0°, he's going up and around the cabinet. Moving in the negative direction (down), he is going toward the bottom of the cabinet. At ±90°, he's either listening to the top of the cabinet or the bottom. I wouldn't recommend flying these in a PA installation, but I've seen weirder.

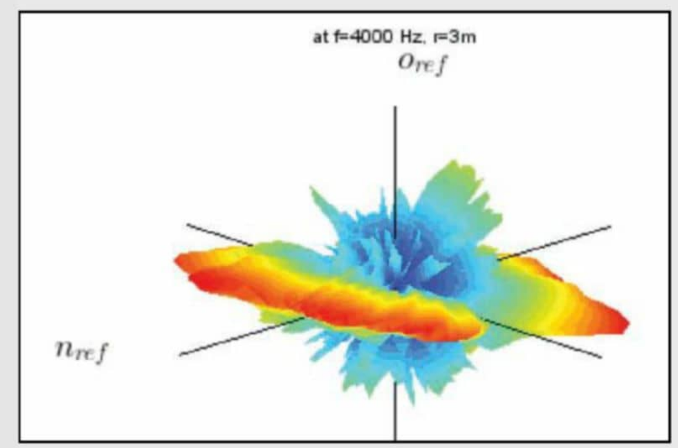


Figure 7: Isometric radiation balloon at 4 kHz, Nref = 0°

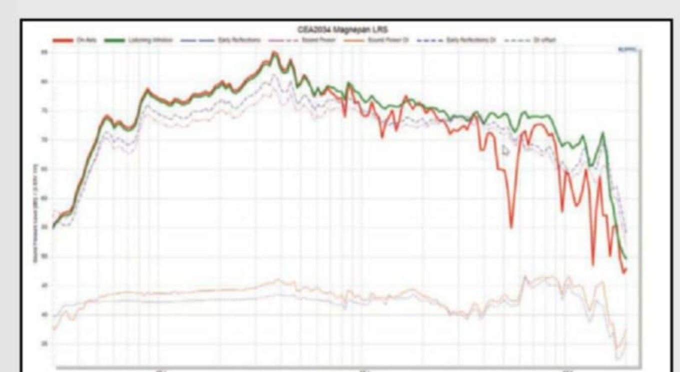


Figure 8: ANSI/CEA2034 graph with indicated listening window

Evident in the contour map is the beaming nature of a planar/electrostatic speaker. Contrary to the horizontal mapping, frequency response looks good albeit in a very tight listening window and angled about 10° up. This, in itself, tells you that analyzing frequency response in the horizontal plane at 0° theta alone might not provide you with all of the information you might need in assessing speakers. It also infers how the speakers are positioned in the default factory stand and tilted relative to the listener's head and floor.

Going back to the measurement platform, we can look at the frequency response at +10°, 20°, and 30° in the vertical plane (see **Figure 6**). Now we're getting somewhere! A +10° shift in the vertical produces a very respectable frequency response curve albeit as tight as can be and already rolling off around 20°. Could this be the difference between sitting and standing at 3 m? **Figure 7** shows the 10° beaming in a 3D radiation balloon.

Of note, there is equal and opposite angular lobing in the rear and at -10°. It is a dipole. Consider though, how the rear reflection possibilities might support the high frequency. Sort of a precedence effect/reverse Haas kicker/reflection board could be placed behind the cabinet on the floor and within a 10 ms reflection window? OK, maybe I'm stretching.

The moral of the story? Don't just rely on the 0° theta horizontal axis measurements to understand a speaker's capabilities, especially when the mid-band or high-frequency components are aimed in a particular vertical orientation.

ANSI/CEA 2034 Analysis and Comments

As mentioned in the introduction, the Magnepan LRS speakers were just between measurements and listening when the world caught a serious virus. Most of us were sent to work from home and that put a stop to a total assessment (including impedance measurements). However, CEA 2034 offers a way to examine the frequency response while the device is in a typical listening space. It does this by integrating and averaging a number of frequency response curves, taking into account near-field reflections. It also provides a listening window, which is a spatial average of the 9 magnitude responses in the ±10° and ±30° angular ranges (see **Figure 8**).

So, when all of the critical, cold and objective analysis is done, at the end of the day and in a "typical" listening environment, the frequency response of the entry-level Magnepan LRS isn't all that bad. Sure, it looks mid-frequency dominant, and begins to roll the high frequency off around 500 Hz, but a little high-shelf equalization can correct for this. Is there a ton of low end? No, but that's typical of electrostatics, and you're probably going to hide a sub in the corner and put a doily on it.

My inclination would be to make the speakers stand perpendicular to the floor so that the high-frequency beaming aimed 10° up would be parallel to the floor and right at my comfortably seated ears. Overall, I think the LRSs are pretty sexy for their very low price point. After these last two reviews of dipoles, my curiosity is heightened, if not just for the uniqueness of the packages.

From Acoustics to Electroacoustics

The Roots of Early Live-Sound Speakers

In this article, Richard Honeycutt discusses the changes in live sound acoustical devices as they have evolved using techonological innovations.

Photo 1: Greek theatrical masks functioned as megaphones.

By
Richard Honeycutt
(United States)

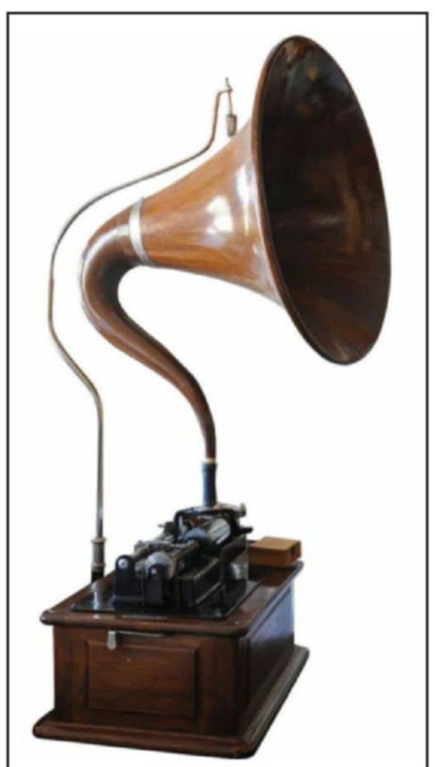


Photo 2: The horn of this Edison cylinder player provided acoustical amplification.
(Image courtesy of Ruby Lane)

The ancient Greeks realized that actors needed technological assistance in projecting their voices to the audience. They used the highest tech known to them: theater masks with what we recognize as short horns located at the actor's mouth (see **Photo 1**). Thousands of years later—in 1910— the phonograph was invented, and more elaborate acoustical devices (still horns) were still the state of the art for acoustical amplification (see **Photo 2**).

In 1926, Joseph P. Maxfield and Henry C. Harrison of Bell Telephone Laboratories developed the first full-range acoustic phonograph—the Victor Orthophonic—based upon the scientific analysis of the acoustical horn published in 1919 by A. G. Webster (see **Figure 1**). The throat of the horn was fed by a 2-mil-thick aluminum diaphragm activated by the phonograph needle. The diaphragm boasted a tangential compliance much like the ones used in modern compression drivers.

The Start of Electroacoustics

The acoustic age and the electroacoustic age overlapped: the first electrical loudspeaker was demonstrated in 1900 by Leon Gaumont and the Gaumont Film Co. The Chronomégaphone was a device that provided synchronized sound playback from a disc recording with the visual image from a projector. It used a horn fed by a modulated-airstream driver. A cross-section of such a driver is shown in **Figure 2**.

A simpler driver with greatly improved performance, invented by Peter Jensen, was used to project concert sound to a crowd of 50,000 in Golden Gate Park in San Francisco, CA, on Christmas Eve in 1915. This driver looked and worked much as modern compression drivers do, and the device—often called a "loudspeaking telephone" (see **Photo 3**)—was used for more than a decade to provide live sound for large crowds.

Due to limitations in the practical size of horns, all the early live-sound events suffered from poor response below about 500 Hz. When Edward Kellogg and Chester Rice invented the cone loudspeaker in 1925, live sound entered a new world of frequency response and naturalness of sound. The Kellogg-Rice speakers could be used to drive large horns with usable response below 100 Hz. Of course, size and weight limited these devices to fixed applications, and they were first popularized by cinemas.

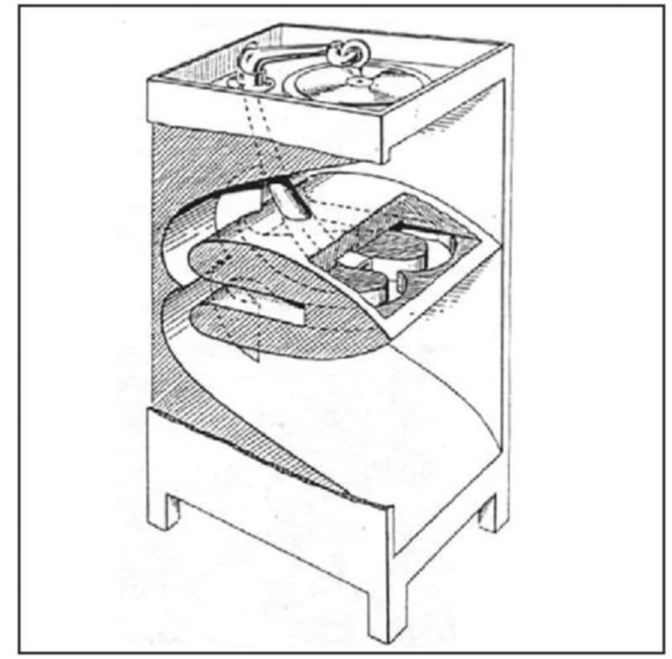


Figure 1: This Victor Orthophonic acoustic phonograph extended the frequency range of phonographs from 2 octaves to 5 octaves.
(Image courtesy of the Audio Engineering Society)

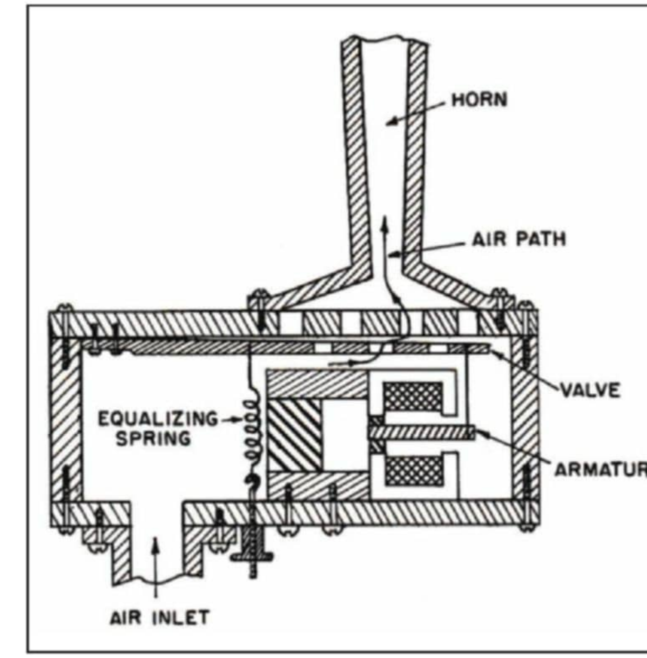


Figure 2: This throttled-airstream driver used an electromagnetically driven valve to modulate the flow of air from a compressor.

One of the first truly fine theater horns was the "Shearer" horn, a folded design using a large cone driver in what is now often called a "W-box." This horn was designed in the 1930s through co-operation among MGM, RCA, and Western Electric, and it was named for Douglas Shearer, head of MGM's sound department. **Photo 4** shows a Shearer horn. Note the "wings" that enlarge the mouth, smoothing bass response.

The First Purpose-Built Live Sound Speaker Systems

An unusual outdoor event took place in 1939 that involved four separate custom-built speaker systems. It was not a "live" sound event in the sense that the term is used today, but there were 206,000 people in attendance on opening day, and more than 4 million people attended the 1939 New York World's Fair.

Of the four custom-built speaker systems, the most fascinating one was part of the sound, light, and water fountain show at the Lagoon of Nations. The program consisted of recorded music and music performed live on a Hammond organ. So it was not a concert but a multi-media event. The speaker system for this show was designed by Rudy Bozak of Cinaudagraph (later, founder of the Bozak speaker company). It consisted of four fireproof sound projection chambers, each containing two low-frequency and four high-frequency horns.

The high frequencies were handled by Western Electric 594A drivers feeding Western Electric 26A multicellular horns. The low-frequency units were double 28-Hz horns, each driven by a gigantic cone speaker designed for the project. Different sources describe these cone speakers as being 27″ speakers built into 30″ frames, or 24″ speakers built into 27″ frames.

Photo 5 shows one of the speakers loaded onto a pallet. The field coil of each speaker weighed 450 lbs., and provided a gap flux of 21,000 gauss (2.1 tesla), corresponding to the gap flux of a top-quality modern mid/high-frequency compression driver, and

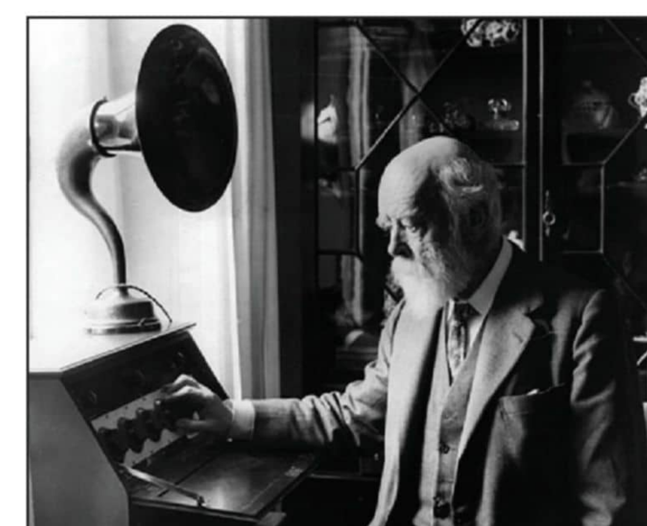


Photo 3: Sir Oliver Lodge is shown listening to a Western Electric model CW929 loudspeaking telephone. (Image courtesy of the Audio Engineering Society)

Photo 4: This is a copy of the original Shearer horn: two W boxes with 15″ drivers topped by a multicellular high-frequency horn with RCA compression drivers. (Photo courtesy of Jim Hunter, Manager of Engineering, Operations, and Resident Historian of the Klipsch Museum of Audio History)

exceeding the gap flux of a high-efficiency modern woofer by a factor of about 1.5. With its 6″ voice coil, each speaker could handle an input power of 125 W continuously. This speaker system served more than 100,000 people at one time, providing the first known example of a stereo system with 360° perspective.

The Altec A7 Voice of the Theatre (VOTT)—an all-horn speaker system designed in 1945— became one of the standard live-sound speakers for a couple of decades (see **Photo 6**), although it was designed for cinema use, as were other members of the VOTT family. It combined a front-loaded horn fed by a 15″ cone driver, a ported rear enclosure for the bass driver, and either a 511 (500 Hz) or 811 (800 Hz) horn. This combination used the ported rear enclosure to produce bass, eliminating the need for a large, heavy bass horn, thus making the A7's relatively portable.

Altec built both the low- and high-frequency drivers for robust power handling, high efficiency, and accurate sound. The rear enclosure was tuned similar to what modern designers might call a "boom box," having a bump in the low-frequency output relative to the mid-frequency output. However, the A7's low-frequency bump was designed to help compensate for the lower efficiency provided by the ported enclosure compared to the front-loaded horn.

Photo 5: This huge speaker was one of eight used to drive the low-frequency double horns at the Lagoon of Nations system. (Photo courtesy of Jim Hunter, Manager of Engineering, Operations, and Resident Historian of the Klipsch Museum of Audio History)

Photo 6: The A-7-500 and A7-8 Voice of the Theatre speakers were the first more-or-less portable live-sound speakers adapted from cinema designs.

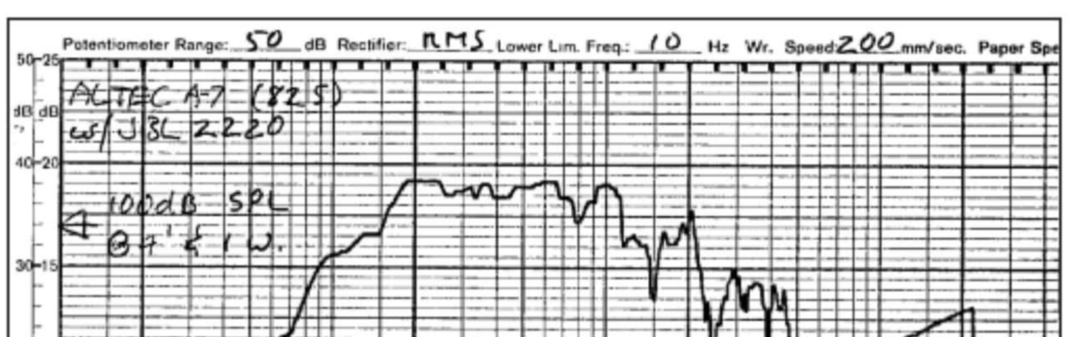


Figure 3: The frequency response of an Altec A7's bass unit with a JBL 2220 driver shows low-frequency drop-off below 200 Hz.

Figure 3 shows the frequency response of an A7's bass unit. This response curve was measured in an anechoic chamber by G. R. Thurmond using a calibrated measurement microphone, a swept sine-wave generator, and a graphic level recorder. As the notations indicate, the 50-dB left vertical axis should be used. The darker vertical grid lines indicate decade multiples of 20 Hz. Thus we see that, despite the boost achieved by the tuning of the rear enclosure, the response at 100 Hz is about 7 dB below that at 200 Hz, and low-frequency response drops more rapidly below 100 Hz.

The ports give a phase inversion to the back wave of the driver, so that it radiates in-phase with the front wave emitted by the horn. The low-frequency drop-off is an unavoidable result of the lower efficiency of the ported rear system compared to the front-loaded horn (~12% versus 25%). Altec A7s were among the speakers used at the Woodstock festival.

About 1960, RCA introduced a large cinema horn similar to the A7 in that it used a combination of a front-loaded horn and a ported rear enclosure. The MI9462 was paired with a short-throw and a long-throw radial horn mounted atop the bass horn, but as shown in **Photo 7**, it was often used with an Altec multicellular mid/high-frequency horn.

In 1963, Paul Klipsch introduced a speaker that was essentially a miniature Shearer horn. Using a W-box design with a single 15″ bass driver, no "wings," and Klipsch's trademark "rubber throat"-modified folding pattern, the La Scala became a significant presence at large outdoor shows. The La Scala (see **Photo 8**), a three-way system that included an internally mounted Klipsch-designed midrange horn and an Electro-Voice T35 horn tweeter, had a remarkably flat frequency response.

The early 1960s also saw the introduction of the JBL "scoops," which were direct-radiator speakers with horn-

Photo 7: This RCA MI9462 bass horn is shown with an Altec 1503 multicellular high-frequency horn.

Photo 8: The Klipsch La Scala was essentially a miniature Shearer horn.

Photo 9: The JBL "double scoop" used two 15" drivers in an enclosure similar to that of the single scoop, but provided somewhat expanded bass response and significantly greater acoustical output.

loaded back waves (see **Photo 9**). The scoops were designed as bass bins only. Users would pair them with mid/high-frequency systems of their choice. Despite published response curves, the measured response shows a less-than-perfectly-flat response including a sharp roll-off below 100 Hz, a hump (dubbed by some sound engineers as a "disco bump") between 100 Hz and 200 Hz, and a sharp dip just above 200 Hz, resulting from the cancellation of the front wave by the out-of-phase rear wave.

Many listeners at live events consider the disco bump to constitute good bass response. Because of the human hearing system's ability to fill in missing bass if its harmonics are present, only very critical listeners can distinguish between a disco bump and truly extended bass response. LP record manufacturers took advantage of this same phenomenon by attenuating the bass below 50 Hz (sometimes below 80 to 100 Hz), in order to reduce groove width and permit more minutes of music per side of a record.

Today's Live Sound Systems

Although A7s and La Scalas were dominant in live-sound applications in the 1950s and 1960s, the situation today is somewhat different. Sound Engineering legend Bill Hanley of Woodstock fame said that it was important to stack the bass speaker cabinets vertically in order to "compress the sound," by which he was describing the process of controlling vertical directivity of a speaker array by extending the vertical dimension. This helps to "put the sound where the people are," rather than shooting it over their heads and into adjoining neighborhoods. Hanley accomplished this stacking by mounting the speakers in towers built on the site for that purpose. Although the line array speakers now ubiquitous in live concerts were not available in Hanley's time, the physics is the same.

Today, live sound events are typically covered by a few line arrays built of specially-designed horn speakers, with separate subwoofers to handle the frequencies below 80 Hz to 100 Hz.

Although it is no longer necessary to custom-build a system such as was used for the 1939 World's Fair, covering the full frequency range of human hearing, modern live sound speaker systems owe much to the engineering skill of the pioneers of the past, without whose work our modern high-power subwoofers and mid/high-frequency compression drivers and horns for line array elements would not have come into being!

Resources

S. Klein, Akustika, 1954.

"Lansing Manufacturing Shearer Horn Model 75W5, Lansing Heritage, Harman International, Courtesy Mark Gander and John Eargle. www.audioheritage.org/html/profiles/lmco/shearer.htm

G. R. Thurmond, "Comparative Measurements of Professional Loudspeaker Components," presented at the 55th Audio Engineering Society (AES) Convention, 1976; preprint 1143.

A. G. Webster, "Acoustical Impedance and the Theory of Horns and of the Phonograph," Proceedings of National Academy of Science of the U.S.A., 1919.

How Do We Generate Measured Noise Plots for Op-Amps?

By
Michael Steffes
(United States)

Every amplifier introduction includes some effort at an input referred noise plot from measured data on a "typical" unit. Only output noise can be measured, so how are these input-referred plots generated? The details for the measurement set up will be reviewed here with the separate input noise terms extracted next.

Every amplifier will be producing an output spot noise spectrum that arises from:

- The amplifier's intrinsic input noise terms
- The gain for each term to the output (that gain might have a non-flat frequency response)
- Resistor noise terms, which will have temperature dependence

Input referring to the amplifier's spot noise terms over frequency allows an estimate of total output noise over the wide range of applications for the amplifier. Each amplifier type will have a slightly different noise model. The most complicated would be for a current feedback amplifier (CFA), as it will have a voltage input noise term at the non-inverting input with two distinct current noise terms at each input as shown in **Figure 1** for a recent CFA introduction. (See the Texas Instruments THS3491 Datasheet, Figure 21 from Reference 1.)

All of these types of datasheet plots require a spectrum analyzer to take the actual measurements. One of the more common has been the 10 Hz to 150 MHz HP3588A (which is obsolete, but still available). These spectrum analyzers have relatively high input referred noise themselves especially at the lowest frequencies where their 1/f noise is very high. These noise measurements depend on a high gain, low noise preamp that is relatively flat over frequency with very low input noise even down to 10 Hz test frequencies.

These measurement flows were developed for high-speed amplifiers but apply equally as well to lower speed voltage feedback amplifiers (VFAs) and, (with some adjustments), to fully differential amplifiers (FDAs). Since it is very common in higher speed to require a doubly terminated 50 Ω cabling interface into and out of the noise preamp, the first step in any swept frequency noise measurement is to add a 50 Ω termination to the noise preamp input and run a point-by-point noise power calibration sweep over frequency at the spectrum analyzer.

The device under test (DUT) needs to produce an output noise that will comfortably raise the measured noise power over this calibration level. When running the DUT noise sweeps, this calibration power level is removed from the DUT measurement sweeps as powers. So whatever spot noise voltage is measured in this calibration sweep, divide that by the net gain from the DUT output pin to the spectrum analyzer to get a minimum DUT output noise that can be successfully extracted. Minimum levels equal to this calibration noise level can be extracted, but higher levels will be more accurate. This, and the bandwidth for the gain of each noise term, lead into DUT test setup considerations.

Noise Measurement Preamp

This measurement preamp has evolved over time using emerging lower noise op-amp solutions. The core amplifier is always a decompensated VFA (to get the lowest input voltage noise), where the original circuits used the CLC425 but now use the 5 GHz LMH6629 (see Reference 2). This decompensated VFA shows an input referred voltage noise of 0.7 nV/√Hz but a relatively high 600 Hz 1/f noise corner (see Figure 28 from Reference 2). This higher noise at low frequencies can be corrected using a composite amplifier updated to the recent OPA1611 VFA (see Reference 3). This 80 MHz GBP audio amplifier offers a 1.1 nV/√Hz input noise with a 1/f corner below 50 Hz. Combining these amplifiers into a low-noise composite preamp suitable for lab characterization gives the full circuit shown in **Figure 2**. Since this is intended for general purpose lab equipment, several added features are included:

- Not knowing the DUT output DC level, we need to block off the DC with an input high-pass filter, set to 16 Hz here. That will roll off the noise a bit below 16 Hz but is a minor error. The final "noisy" data is fit to a 1/f model so this will usually be inconsequential as most DUTs have 1/f corner far higher than the <50 Hz input referred noise 1/f corner in this design.
- Since there will be cabling getting connected and disconnected, the 10 kΩ to ground at the input will always provide a DC reference when the input is open.
- To ensure a safe DC start up, the 464 Ω shunt resistor on the output side of the 400 µF blocking capacitor is used. That, in parallel with the 56 Ω gain resistor into the inverting input of the LMH6629, will provide a high-frequency 50 Ω match to the 50 Ω cables (often coming out of noise shielding enclosures).
- The amplifier gain is set to -50 V/V gain with the 2.8 kΩ feedback resistor—noise measurements do not need to concern themselves with gain polarity.
- One of the interesting advantages of matched input impedance inverting operation is the noise gain (NG) is much lower than the signal gain. Here that is 1 + 2.8 kΩ/101 Ω = 28.7 V/V. The LMH6629 model produces a 5 GHz gain bandwidth product. This should yield a closed loop F-3 dB of about 174 MHz in the LMH6629 high-frequency path. Since the simulation of Figure 2 showed a 225 MHz F-3 dB, there is room to increase the gain provided in this design and still hold good flatness through test frequencies > 50 Mhz.
- Not knowing the source voltages, cabling sequence, and so forth, protection diodes are added on the inverting summing junction to protect the LMH6629

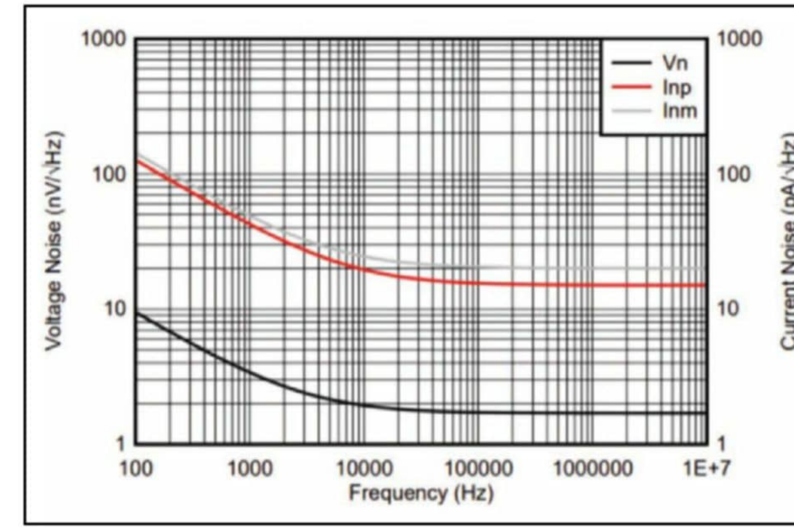


Figure 1: Input spot noise over frequency is shown for the THS3491 CFA.

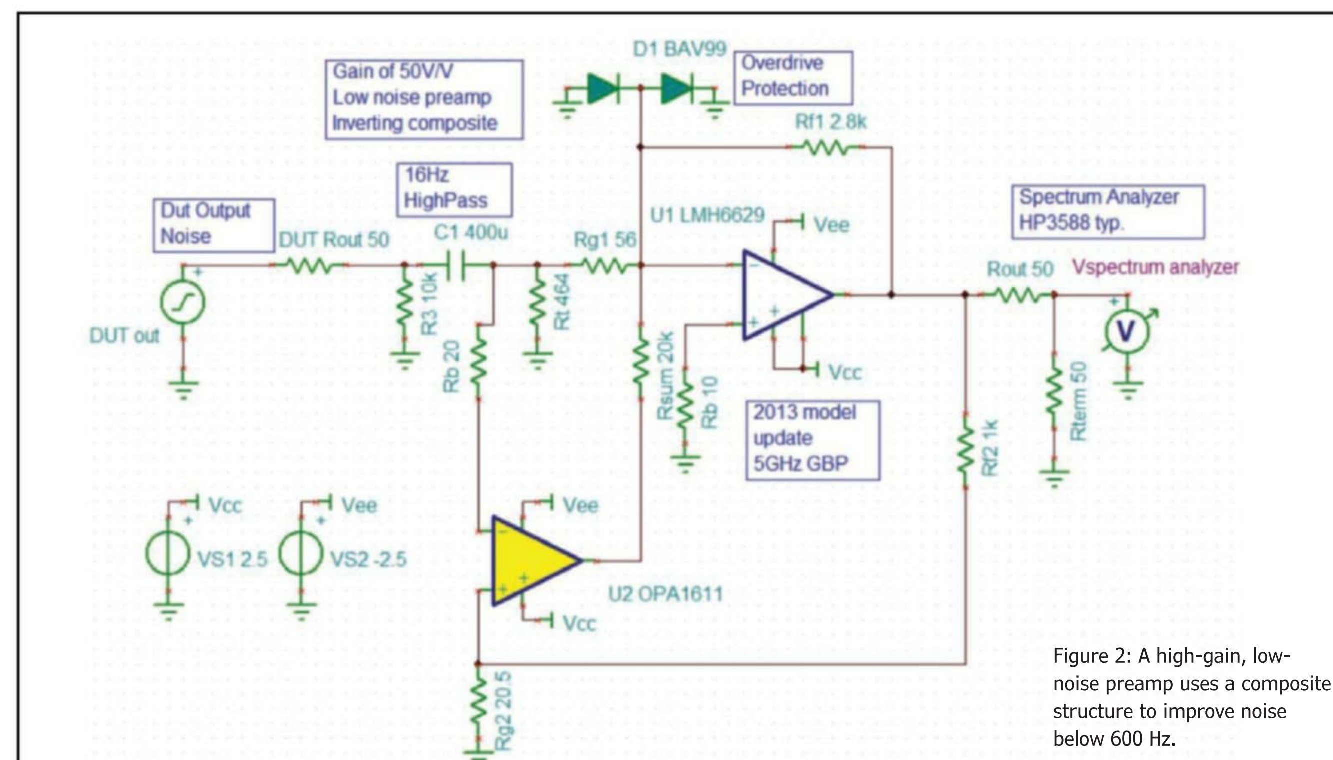


Figure 2: A high-gain, low-noise preamp uses a composite structure to improve noise below 600 Hz.

inputs from overdrive. This does add some slight parasitic capacitance, but the NG is high enough to remove that from stability concerns.

The original intent for this inverting composite circuit was to correct the poor DC precision of CFA amplifiers. And, indeed this circuit simulates to show a nominal output DC offset of 3.9 mV arising from the OPA1611 input offset voltage and its gain of 50 V/V. This is low enough to require no output side blocking capacitor.

At low frequencies, the output is controlled by the OPA1611 where it senses the input signal through the 20 Ω bias current cancellation resistor. Its output is then fed into the LMH6629 as an inverting amplifier, which means its feedback needs to come back to the non-inverting input to render the overall OPA1611 loop negative feedback. That R_{SUM} resistor can be used to tune the transition flatness.

While this circuit was intended for DC correction, this correction amplifier also controls the output noise below its bandwidth of operation. Running an input referred noise simulation (referred to the DUT output pin) shows the very low noise over a broad bandwidth for this noise preamp. It is this noise (along with the spectrum analyzer input noise) that must be exceeded for each of the DUT test configurations (see **Figure 3**).

The low frequency 1/f is coming from the OPA1611 TINA model. This circuit is passing control to the LMH6629 in that 300 kHz to 2 MHz noise transition to the lower level produced by the LMH6629 model.

The overall gain response from the DUT output to the spectrum analyzer shows a reasonably flat response over the desired 10 Hz to 100 MHz span. There will always be a gain ripple in the crossover region as the gains cannot be exactly matched. **Figure 4** shows the simulated gain from the DUT output to the spectrum analyzer input (including two 6 dB insertion losses due to matching cables).

While this preamplifier looks to be useable through 100 MHz, most characterization efforts (refer to Figure 1) span the 100 Hz to 10 MHz range where this preamp provides 0.3 dB flatness. A single gain number is usually used to refer the measured noise at the spectrum analyzer (in this case, 12.36 V/V) to the DUT output. It would be possible to measure this preamp at the same intended frequency points for DUT noise sweeps to make a more accurate DUT output referred estimate.

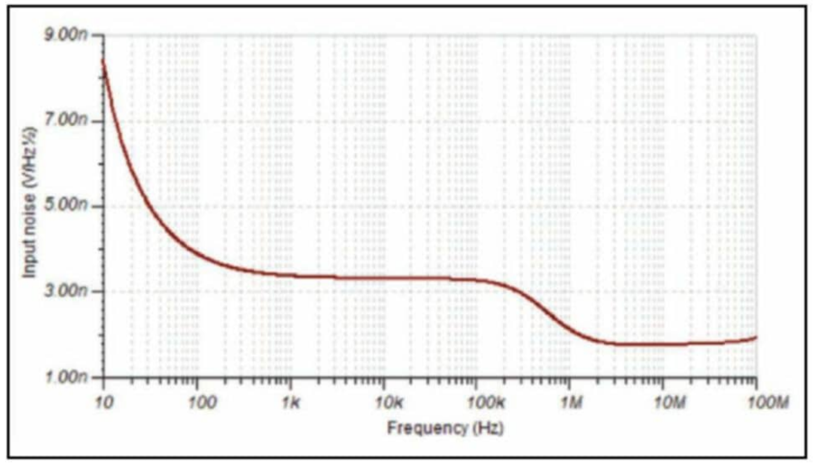


Figure 3: This graph shows the spot noise for the noise preamp referred to as the DUT output.

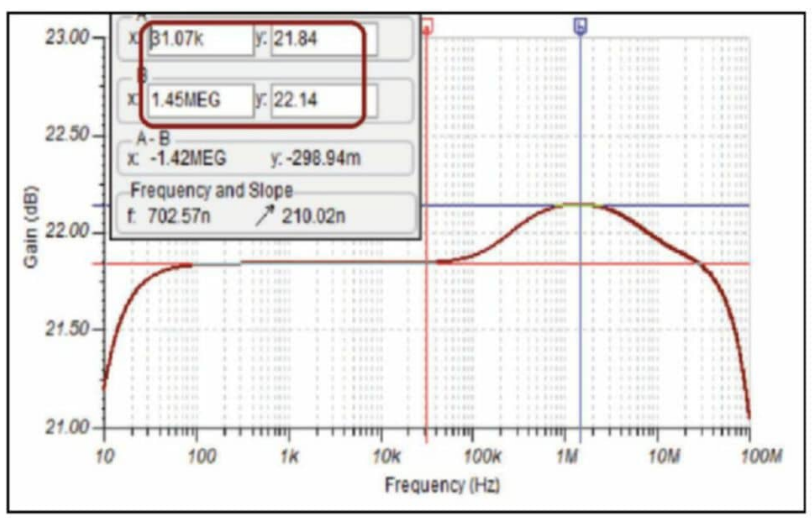


Figure 4: Here is the gain response flatness over the 10 Hz to 100 MHz span.

Next Month

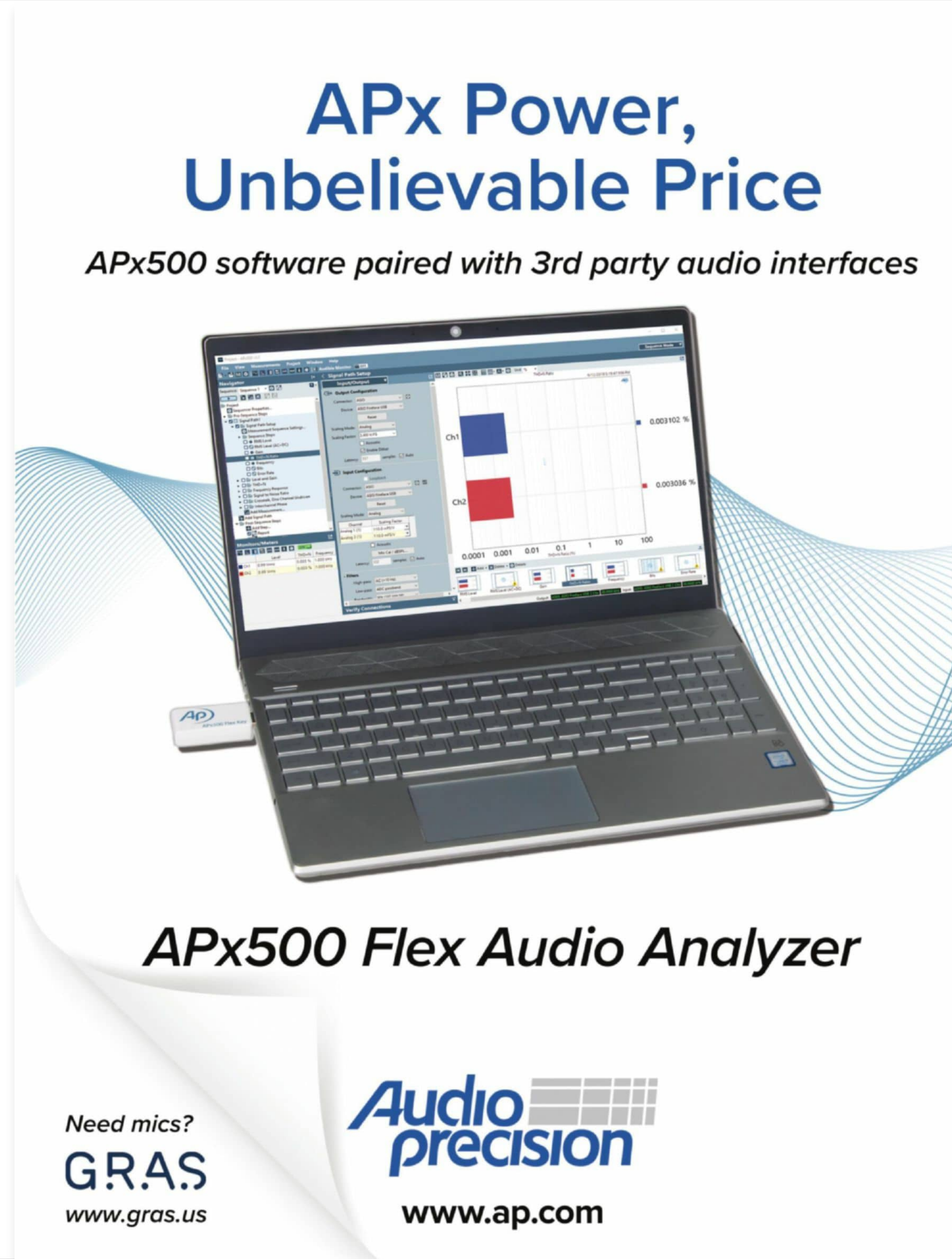
Having this very capable noise preamp available greatly eases an amplifier noise characterization effort. Since we need to model three independent input noise sources for a CFA amplifier similar to the one shown in Figure 1, three separate DUT setups are used to emphasize the different terms then solve for them input referred. This will be the topic in the next issue.

References

[1] "900 MHz, 500 mA High Power Output Current Feedback Amplifier," Texas Instruments THS3491 Datasheet, revised July 2018, www.ti.com/lit/ds/symlink/ths3491.pdf.

[2] "Ultra Low Noise, High Speed Op-Amp with Shutdown," Texas Instruments LMH6629 Datasheet, revised March 2013, www.ti.com/lit/ds/symlink/lmh6629.pdf.

[3] "SoundPlus High Performance, Bipolar Input Audio Op-Amp," Texas Instruments OPA1611 Datasheet, revised August 2014, www.ti.com/lit/ds/symlink/opa1611.pdf.



Altec Lansing's Voice of the Theatre Speakers

What You Need to Know

By
Scott Dorsey
(United States)

An official Altec Lansing photo shows the Voice of the Theatre Speaker range.

A design by John Hilliard, Altec Lansing Voice of the Theatre systems originally targeted movie theaters. The entire series was gradually expanded for use in general sound reinforcement and later converted in extreme high-end audio segments for hi-fi home use. This article explains what we need to know about these speakers that are still highly sought-after with many still in use in cinemas.

The Altec Lansing Voice of the Theatre series was one of the innovations coming out of the post-war era. John Hilliard at Altec Lansing took many of the acoustical innovations that came out of the World War II effort and developed a new loudspeaker design for large halls that was completely different from the Western Electric cinema systems of the pre-war era.

First launched by Altec in 1947, the designs were available in a number of different two-way configurations, built on the same building blocks with Model 515 woofers and the new 288 high-frequency compression drivers, with and without side baffles. Within a few years, they had introduced a number of additional configurations including the classic A7—a compact version designed for smaller cinemas but which found its way toward becoming perhaps the most popular sound reinforcement speaker of the 1950s through the 1970s.

Today, the A7 is one of the most sought-after speakers for home audio in Japan, in part because real estate is so expensive in Japan that to be able to say you have a pair of A7s is to imply that you have a living room that is so large that it requires a pair of A7s to fill it. And they still can be found in sound reinforcement service as can many of the larger Voice of the Theatre systems.

This article discusses some of the things that you should know about the Voice of the Theatre system for theater or auditorium installations. It is by no means authoritative, it's just an overview of what is out there, along with some of my personal thoughts about the system.

Bass Horns

What makes these speakers interesting today, and what makes them different from nearly anything currently made, is that they are bass horns. There is a high-frequency horn on top (a sectoral horn that at the time gave the best pattern control), but the low frequencies are also produced by a large folded horn driven by large woofers specifically intended for high-compression applications. This was in many ways the precursor to the W-Bin bass cabinets of the 1970s, but the direct-firing woofer made for response that remained flat up into mid-range frequencies.

The bad news about this is that very low bass extension was not possible, because with a bass horn, the lowest possible frequency was limited by the horn's exit width. Once the wavelength becomes wider than the horn exit, the response precipitously drops. One thing that could be done was to put multiple horns together making a super-wide horn (as used in the Altec X-1), and another was to add baffles to the side of the cabinet, which effectively increased the size of the horn as used in a variety of cabinets and available as retrofits for many others.

The good news about this is that the efficiency is very high because the coupling between the woofer and the outside air is almost perfect above that cut-off frequency. For example, 100 mW out of a Walkman headphone jack will produce room-filling sound in a large auditorium, and two 6L6 tubes putting out 20 W are enough for fairly dynamic film soundtracks.

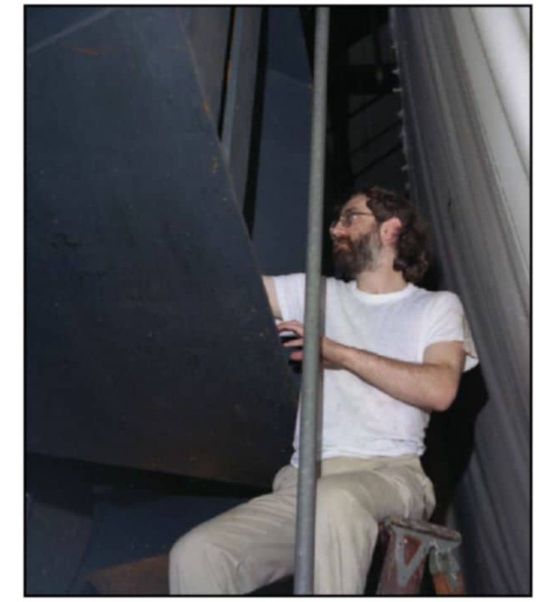
Another point that should be made is that in 1947, the physics of horn speakers was very well-understood. Engineers still did not have a grip on all the parasitic resonances bouncing around inside big horns, but the fundamental physics of operation was no longer in question. This was not the case for bass reflex speakers, which had to wait until Neville Thiele and Richard Small's paper was published in 1961. So, unlike the sealed and the vented box designs of the era, the Voice of the Theatre speakers were systematically designed.

Amplifiers

One of the major advantages of these speakers when they were new was that they were so efficient. Back in the days when the largest power pentode you could buy was a 6L6, amplifier power was expensive and amplifier distortion at high powers was quite high. As Paul Klipsch always claimed, the key to reducing the total system distortion back in those days was to improve speaker efficiency, more than to improve the speaker linearity.

The problem is that by the 1970s, amplifier power was cheap, and what was worse, many of the more popular power amplifiers were biased in such a way that the distortion at low power was much higher than the distortion at maximum power. That is, the primary amplifier distortion mechanism was crossover distortion, and some people argued that this was a consequence of advertisements and datasheets listing full power distortion, causing a push for engineers to try to reduce full power distortion at the expense of all other specifications.

Whatever was the cause of it, the end result was terrible, terrible sound. Take a Phase Linear or South West Technical Products Corp. (SWTPC) amplifier from the 1970s and put it on an A-7—you get room-filling volume with a fraction of a watt. Unfortunately when delivering a fraction of a watt,



Taken in the 1970s, this photo shows the author hanging behind an Altec A-5 loudspeaker and suspended over a stage while changing diaphragms in situ. (Photo take by Scott Norwood)

these amplifiers produced horrible shrill and shrieky sounds. This gave the Altecs an undeservedly bad name at the time, when in fact it was the amplifier that should have been blamed.

Phase Coherency

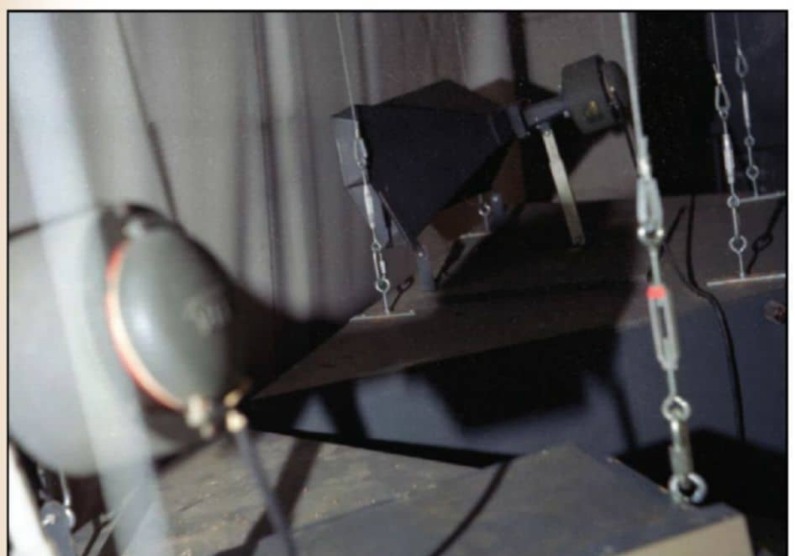
These systems are intended to be listened to at a distance, in a wide and fairly flat room with a tapered floor, so that most of the listeners are in the same vertical plane as the speaker. They might be a bit higher or lower if they are seated in the front or back rows but they won't be too far off. As a consequence, the phase coherency in the horizontal plane was not something about which the designers of these speakers ever cared.

Other companies made cabinets based on Altec plans and filled them with Altec hardware. These A-5 cabinets were built by Simplex for very small Cinemascope installations.

Right on-axis, the listener has the same distance to both the woofer and tweeter horns, but move substantially up or down and this changes. Several feet of distance between the woofer and the tweeter in most of the Altec models makes this a very serious problem for use outside of a theater.

The A-7 was available with an inverted configuration where the high-frequency horn could be mounted inside the bass horn instead of being mounted on top of the box. Although still far from a coaxial design, this gave much better phase coherence and is a huge improvement in an environment where listeners may not be on the same horizontal plane as the loudspeakers. Any A-7 being used today should be set up as an inverted cabinet with the horn mounted within the box. Not only does this make it more rugged and roadable, but the phase alignment is far better. Unfortunately, the other Voice of the Theatre systems do not permit this configuration.

In all cases, the horn/woofer alignment needs to be adjusted for the room so that the acoustic centers of both have the same distance to the listener in the middle of the room. The procedure for doing this can be found in the manuals.



Altec compression 288 drivers are pictured in place on suspended cabinets.

About the Author

Scott Dorsey has a degree in electrical engineering, during the pursuit of which he worked in the broadcast and recording industries. After several years working at a major studio, he took a job with a defense contractor. This left him time to do live concert recording for acoustical music and to design and build audio devices for personal use and on contract to several audio manufacturers and importers. Scott Dorsey is a regular contributor to several audio magazines. He has been publishing equipment reviews and DIY projects since the mid-1980s. He is probably best known in the general audio community for his retrofit electronics designs in inexpensive Oktava, AKG, and Feilo microphones.

The Various Models

I'm not going to list all the dozens of different models made over the years, other than to say that they all have 1" or 1.4" compression drivers with sectoral horns on top and bass horns with 15" cone woofers on the bottom. They also either have 800 Hz or 500 Hz crossovers.

There are some with a single 1" driver on top, and a single 416 on the bottom with an 800 Hz crossover. At the other end of the spectrum, there is the A1X with four 1.4" compression drivers on top and six 515 woofers on the bottom. This is what I mean by a "modular system" in that there are a huge variety of different products that were sold by mixing and matching.

In addition to the standard line, Altec also sold some special order items with nonstandard configurations. (For instance, two 1" compression drivers into a single manifold was a curious arrangement that you will not see in the catalog.) And, given how long the systems were sold, there have also been a large number of curious hybrids that people have built with Voice of the Theatre components, many of which were passed off as production speakers but actually were not.

It's not my intention to describe all the combinations available, and the original Altec documentation is available online if you want to look them up. The A-5 and A-7 speakers were the most common, although they both came in several different variants over the years.

In addition, a lot of Altec products that use these drivers or similar drivers are frequently sold as Voice of the Theatre speakers even when they are not. A tremendous variety of similarly sized Altec speakers are sold today on eBay as A-7s, when they were never sold originally as A-7s and do not have a component lineup that was ever used in the A-7s. So make sure you know what you are buying.

My personal feeling is that the speakers using the 1" compression drivers with a 500 Hz crossover sound better than the ones using the 1.4" driver, if only because it keeps the crossover artifacts farther out of the vocal band and the driver seems less prone to parasitic resonances in the top octave. (Also, availability of high-quality aftermarket diaphragms for the smaller compression drivers is better and that can lead to better sound as well.)

On the other hand, a friend of mine who does cinema sound installations does not like the 1" drivers because they are comparatively delicate. He says every time he has ever encountered one, the diaphragm was blown and that he much prefers the 288. There is something to be said for that, because any driver will sound better than a blown one. And in sound reinforcement applications, it is already far too easy to damage the 288 with extreme transients.

It is true that Altec overstated the ability of these speakers, and it was not unusual to see systems installed that were undersized for the room. Take the manufacturer's recommendations for acceptable auditorium sizes with a grain of salt.

If you have a horn designed for a 1.4" driver and you want to change to a 1" driver, the appropriate adaptor is Altec part 21216. Be sure you have enough power handling in the smaller driver to get enough sound for the room, though.

Note that there also are two different phase plug designs for the 1" drivers—the original circumferential plug and the new Tangerine one. The two sound a little bit different and I personally think both of them are fine as long as you make sure that you use all of the same kind. Since so many systems out there were assembled from bits and pieces, this is an important thing to check.

This is a simple first-order 500kc crossover.

The Treble Horn

All of these drivers were available with a variety of diaphragms from horrible phenolic things to conventional aluminum ones. There were various specialty ones with names like "Pascalite" and "Symbiotik." I won't give details on any of these because you don't want any part of them. It doesn't matter what diaphragms you have in there because it's almost certainly in poor condition. Put new ones in.

Materion Truextent makes replacement diaphragms for the smaller compression drivers, which have better high-frequency response, lower distortion, and better power handling. There is no reason not to be using them other than price, and the price is likely worth it in a commercial installation.

A number of companies are making third-party replacement diaphragms of varying quality. Some are very bad, and appear under a variety of different names—sometimes names that have been used for other manufacturer's diaphragms in the past. I would urge you to avoid these if at all possible.

Note that these drivers were sold with 8 Ω, 16 Ω, and 24 Ω diaphragms, so make sure to get a replacement of the same impedance, if you are using a passive crossover like the original configuration.

This image shows an Altec 515 woofer in place.

Resources

Altec Corp., "Construction Plans for Low-Frequency Enclosures," Altec Lansing Applications Notes AN-11, http://alteclansingunofficial.nlenet.net/publications/techletters/AN-11_EnclosurePlans.pdf.

A. Budmaieff, "Loudspeaker Enclosures, Their Design and Use," Altec Corp., Catalog No. AL 1307-3.

J. Markwart and J. Tucker, "Altec Voice of the Theater Speakers for Hi-Fi," *Sound Practices*, Summer 1993.

J. Stronczer, "La Voice of the Theater Chez Nous: Taming the Altec A5 Classic for Domestic Use," *Sound Practices*, Volume 2, Issue 11.

Some of the Altec compression drivers, most notably almost all of the 288 drivers, have alnico magnets, which can suffer from loss of magnetism over the years especially if dropped or mechanically shocked. These can be remagnetized by Great Plains Audio (https://greatplainsaudio.com), which has the remaining Voice of the Theatre parts inventory from Altec as well. I don't know anyone else set up to remagnetize them. The later drivers use ferrite magnets and are more stable.

In addition to upgrading the diaphragm, there are two very popular modifications for the top end. The first is to remove the bug filter from the horn's throat. The bug filter is a metal mesh mounted into the front of the compression driver to keep insects out. Removing it makes the device more delicate and prone to damage from insects and debris, but the improvement in high-end detail at high levels is considerable and the expense is nil.

The second modification is to damp the horn so it no longer rings. Strike it with a screwdriver and the stock horn rings like a bell. Put automotive undercoating on the outside or a mixture of tar and sand, and the ringing will stop. This is not expensive and is a tremendous improvement in the top end.

Some people have advocated cutting the braces at the horn's mouth and filling in the broken parts with RTV, so as to isolate the braces from the horn itself mechanically. This emulates some of the later Altec horn designs. I have not tried it and cannot recommend it. It seems difficult mechanically.

Bass Horns

If you knock on the side of the cabinets, you will find that the bass horns on these systems severely ring. They sound hollow, rather than giving a dull thud. Stiffening the box with external sheets of Baltic Birch can help a lot, along with adding internal wood blocks for bracing. I have seen some of these cabinets where internal bracing was added with cheap pine 2 × 4s and that seems like a good improvement, if you are on a budget.

Of course, if these systems are being flown, you have to worry about structural support and about adding too much weight. But if they are on the normal platforms behind a projection screen there's no reason not to add as much weight as needed to make them go thunk instead of sounding hollow when struck.

Often as these cabinets age, the wood shrinks and a gap will appear along the bottom edge of the bass horn. Filling that gap with RTV or wood filler will help a lot.

Some people have used Bondo or tar behind the curved flare sections to damp them down and keep them from ringing. Other people have installed a wooden baffle and pumped the area between the baffle and the flare with expanding polyurethane foam. I do not know how effective this is—on some of the smaller speakers, those sections do not ring much when knocked with the knuckles, but I could see this being a help on the larger models with longer unsupported sections where the flare ringing is a serious problem and a big source of midrange mud. I'd pick Bondo as the first way of dealing with that since it seems like the easiest one to undo if it is a problem.

Wings

Many of these speakers had "wings" or bass baffles on the side of the cabinet. If yours do not, adding them is an easy way to get nearly an octave better bass extension for minimal cost. Altec published plans for all of the Voice of the Theatre cabinets, including plans for the wings. Use a high-density plywood (e.g., Baltic Birch) if you can afford it. The majority of installations never used these wings, which is a shame since they are an easy upgrade. However, this is not of much use for a system that has to go on the road.

Crossover

The existing crossover is a first-order (6 dB/octave) filter using an iron core inductor on the bottom end and an oil capacitor on the top end. It is strongly recommended that you ditch it and biamp these speakers using a modern active crossover.

If you cannot biamp, consider replacing the crossover with a modern design. In John Stronczer's *Sound Practices* article from 1990, he details two modern crossover designs using second-order filters, and I recommend either one of them. Do note that some of these speakers were built for 16 Ω or 24 Ω operation, in that case components need to be scaled up appropriately.

If you absolutely must use the first-order crossover, at least replace the capacitors with modern film types and the inductors with air core inductors that won't easily saturate. But really, you want to biamp if at all possible.

Repairs and Modifications

Please watch out for aftermarket parts. Although I highly recommend Radian Audio Engineering and Truetexent's aftermarket diaphragms for the 1" throat drivers, there are some companies selling some very doubtful replacements.I do not know of a good Altec 288 replacement diaphragm right now other than the one from Altec (although the situation may have improved since I last looked). There is a Taiwanese supplier making a 288 replacement diaphragm, which has serious mid-range problems.

The same goes for replacement cones for the 416 and the 515 woofers. The aftermarket cones all appear to be far less rigid than the originals and will break up at normal listening levels. The aftermarket cloth surrounds appear to be okay, however. These aftermarket components have often been sold fraudulently as being original Altec material, so watch out.

If you see the Altec 30904 attenuator box on any of the Voice of the Theatre speakers, remove it. It was intended for some of the studio monitor systems, but a lot of people installed them into A-7s for rock band use. Take them out when you're taking out the crossover. If you can't afford to upgrade the crossover yet, take them out beforehand.

Conclusion

The Altec Voice of the Theater series was a milestone in loudspeaker design and these speakers can still perform well in a large auditorium today. They remain an excellent choice for cinema systems, but many of the speakers are suffering from deferred maintenance issues after so many decades of sitting unattended behind a screen.

Even well-maintained ones can benefit from some upgrades. Since there were so many different models of these speakers, there is a choice for nearly any large hall.

Loudspeaker Measurements at Home

A Guide for Building Your Own Test Chamber

By
Geoff Hill
(United Kingdom)

A lot of Geoff Hill's work is now based at home, and indeed his front room. And like so many acoustics builders, designers, and engineers in these crazy times, he no longer has access to an anechoic chamber, so how is it possible to still make useful loudspeaker measurements? In this article, he explains how to build a basic tetrahedral chamber to use at home, and then put together a measurement system and check the results.

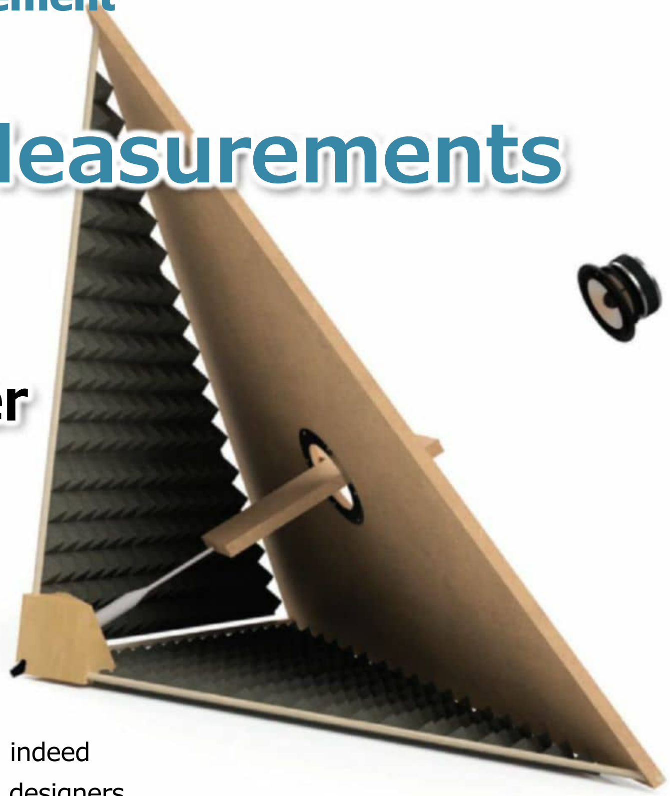


Photo 1: This is the completed tetrahedral chamber, which you can use to to test your loudspeakers when you don't have access to an anechoic chamber.

Like so many of us, I am currently stuck at home in isolation. I know you're probably faced with a similar situation and it's even worse for us acoustic folks—without an anechoic chamber we are stuck!

We all know that anechoic chambers are big and expensive, originally built for the military during World War II to test loud sound sources without creating a public nuisance. So how did we end up making loudspeaker measurements in anechoic chambers?

Around the same time as the Second World War, engineers were still struggling to measure loudspeakers. The measurement equipment was limited, and the anechoic chamber was a big step in the quest toward making accurate, repeatable loudspeaker measurements.

New Normal for Loudspeaker Measurements

As the pandemic forces us to do a lot of things differently, it's time to take a fresh look at how we can make loudspeaker measurements during the new normal.

First, let us take a look at the alternatives that we have used:

- Free-field measurements
- Ground plane measurements
- Anechoic measurements
- Anechoic IEC baffle or JIS test box measurements
- Windowed/Gated measurements

All of these methods have their advantages and disadvantages, are expensive, and rely upon having a large physical space in which to work. This is not possible when working at home or even in most workplaces, but there are modern standards created to overcome these issues. Using them, we can make accurate, repeatable, calibrated, and transferrable loudspeaker measurements, as an alternative to these traditional methods. We can do this by building our own basic tetrahedral chamber (see **Photo 1**)—giving us a convenient, low-cost approach during the new normal.

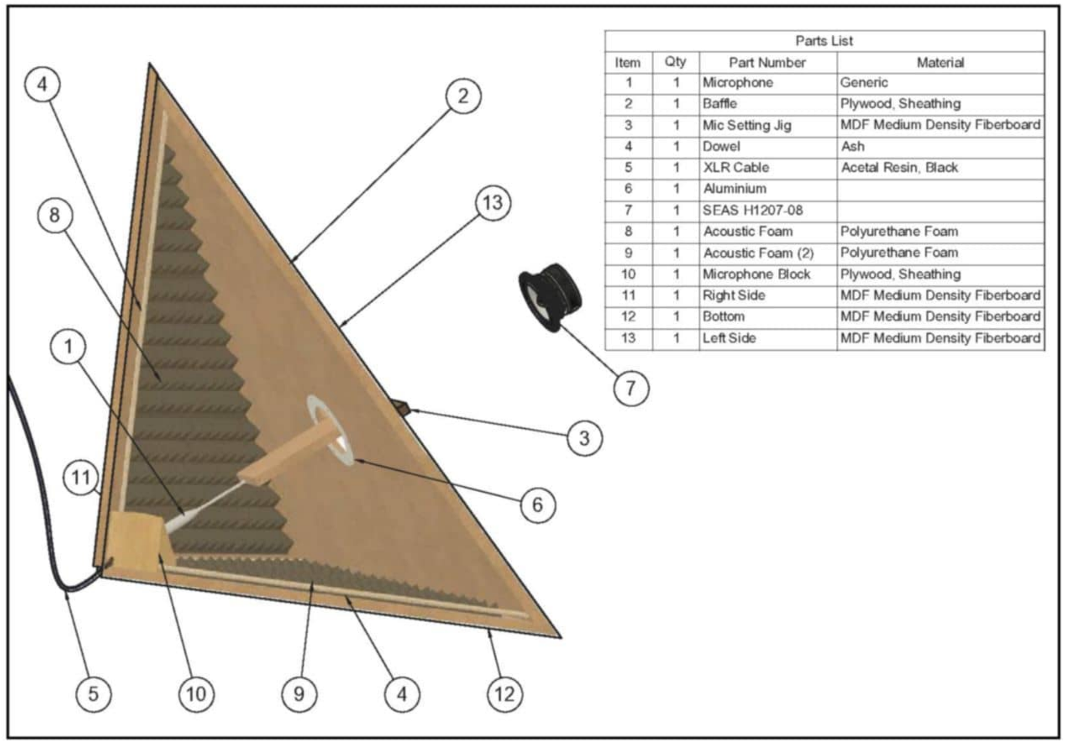
The tetrahedral chamber concept was launched at the Loudspeaker and Headphone Conference in Helsinki, Finland, with the theory being published at the Audio Engineering Society (AES) convention in New York. The theory was based upon a tetrahedral structure that can fit into a corner anywhere.

Following peer review, this type of chamber has now been incorporated into international standards [(1) IEC 60268-21-2018, (2) draft IEC 60268-22, and (3) AES 73id-2019] and widely accepted as a valid methodology for comparative loudspeaker measurements. These standards give us a compact and convenient, defined structure, which is small and relatively low-cost and meets our requirements for making accurate, repeatable loudspeaker measurements.

Can We Use These Modern Standards At Home?

You should be able to build a tetrahedral chamber with a minimum number of tools—this will enable you to take loudspeaker drive unit measurements in the new normal. Further, I will base many of my other measurement suggestions around free, open-source software and hardware, assuming that most people will not have access to professional test equipment.

When compared to a full Tetrahedral Test Chamber (TTC), this implementation will be constrained in a few ways. First and foremost, it will not have the physical integrity of a TTC, and the structure will need to be larger than a TTC would be to measure the same size of loudspeaker. Also, it will be more difficult to calibrate than a TTC, it will not have the stability and accuracy of a TTC, and it will not have the acoustic damping of a TTC. Having said all of that—it's still better than an anechoic chamber!



Parts List			
Item	Qty	Part Number	Material
1	1	Microphone	Generic
2	1	Baffle	Plywood, Sheathing
3	1	Mic Setting Jig	MDF Medium Density Fiberboard
4	1	Dowel	Ash
5	1	XLR Cable	Acetal Resin, Black
6	1	Aluminium	
7	1	SEAS H1207-08	
8	1	Acoustic Foam	Polyurethane Foam
9	1	Acoustic Foam (2)	Polyurethane Foam
10	1	Microphone Block	Plywood, Sheathing
11	1	Right Side	MDF Medium Density Fiberboard
12	1	Bottom	MDF Medium Density Fiberboard
13	1	Left Side	MDF Medium Density Fiberboard

Figure 1: The full assembly is shown with acoustic damping on the bottom and the parts list. The chamber is shown without the left side fitted.

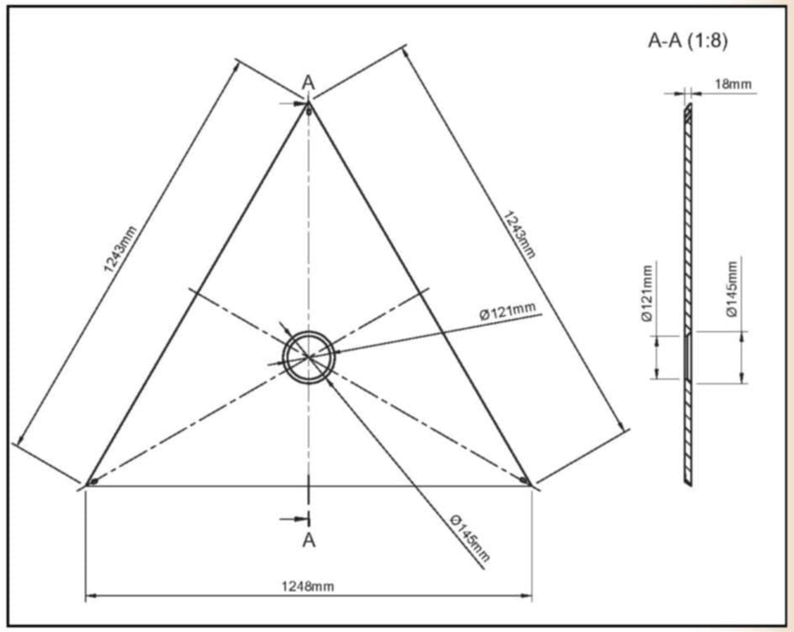


Figure 2: This triangle baffle has been cut in the center to fit my SEAS 4" driver.

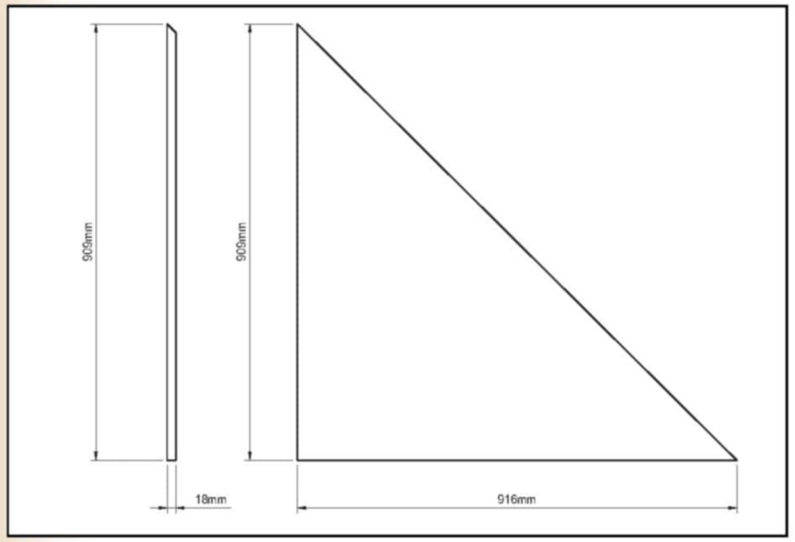


Figure 3: This diagram details the construction of the chamber's sides and bottom.

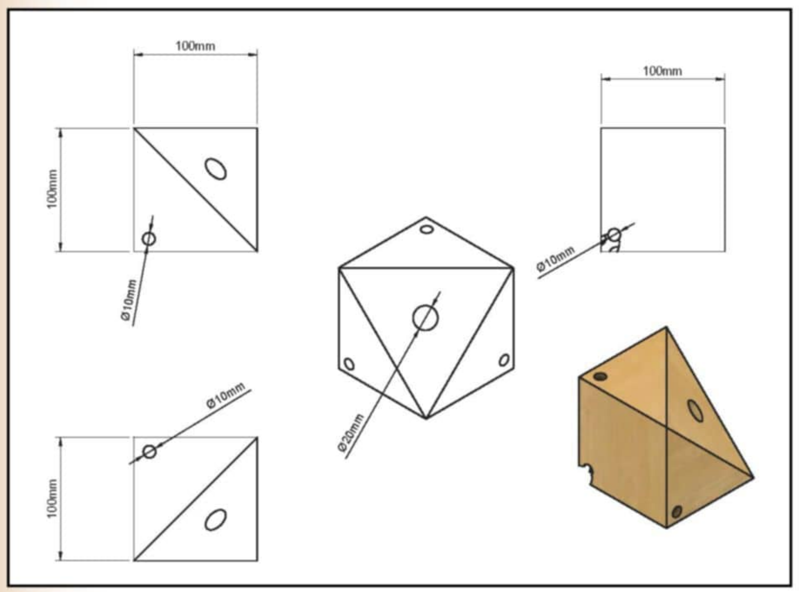


Figure 4: Here is the microphone block assembly.

Photo 2: Here we have cut the mic block, note the hole drilled for a dowel.

Assuming you have a loudspeaker to measure, we will also need the following equipment:

- A microphone—we will use an affordable measurement microphone (e.g., a Behringer ECM8000) and/or a USB microphone (e.g., the miniDSP UMIK-1).
- A computer with a freeware analyzer and a sound card
- An audio amplifier
- Leads and connectors

We'll build this basic tetrahedral chamber into an existing corner of a room. For those who have access to woodworking equipment, it will undoubtedly prove easier but not essential. To build the chamber, we will need the following parts:

- One sheet of wood, 18 mm thick (from this you'll make the baffle)
- Chamber sides and bottom
- Mic setting jig
- Mic block

For the chamber's sides and bottom, we will use three lengths of wooden dowels—800 mm long × 10 mm plus diameter—and acoustic damping, foam, or fiberglass insulation, 25 mm thick. We also need a piece of aluminum sheet, 0.8 mm to 1 mm thick.

The final assembly with acoustic damping on the bottom is shown in **Figure 1**.

Our first task is to produce these parts, and we should start by making the triangular baffle. I have drawn an example (see **Figure 2**) to accept the SEAS 4" driver with a hole through the middle of the baffle.

Next, you should make the chamber's sides and the bottom (see **Figure 3**), cut the dowels to length and from the 18 mm sheet cut five squares each 90 mm on the side. Use these five layers of plywood to produce a cube. Drill the three holes for the dowels on the sides of the cube. This is then cut on the diagonal into two parts (see **Figure 4** and **Photo 2**).

Then, make the Microphone Setting Jig. Produce the aluminum ring, which ensures the loudspeaker is flush with the baffle and is essential. Then screw it into the baffle as this prevents the loudspeaker from falling through the baffle. Next, cut some acoustic foam into three parts and put the parts together as shown in Figure 1. More detailed diagrams can be found in the Supplementary Materials section on the *audioXpress* website (see Project Files).

About the Author

Geoff Hill, the inventor of the Tetrahedral Test Chamber (TTC) and CTO at Hill Acoustics, has been working in the loudspeaker and audio industry for more than 40 years. He is the vice-chair of the Audio Engineering Society's AES SC-04-03 Working Group on loudspeaker modeling and measurement and a member on the International Electrotechnical Commission's IEC Working Group TC 100/TA 20/PT 60268-22. Hill is also the author of *Loudspeaker Modelling and Design: A Practical Introduction* (Routledge 2018). He welcomes your comments and can be contacted via email at geoff@hillacoustics.com.

The Measurements

So how do we go about making our measurements? I will describe the basic steps.

First, we start by measuring the external very-near-field frequency response of the loudspeaker (Step A). Next, we measure the internal pressure frequency response (Step B) and we create the difference curve between A and B (Step C), allowing us to create the correction curve for the low frequencies (Step D). Finally, we apply the correction curve to get the equivalent free-field measurement (Step E).

Our Step A is to make a measurement outside of the chamber! Yes, I know that sounds crazy but let me explain. What we need to do is to measure the very-near-field response of the rear of loudspeaker not the front which we usually do. This yields the true low-frequency response. However, the high frequencies are not correct because of four things: reflections from the magnet and basket;

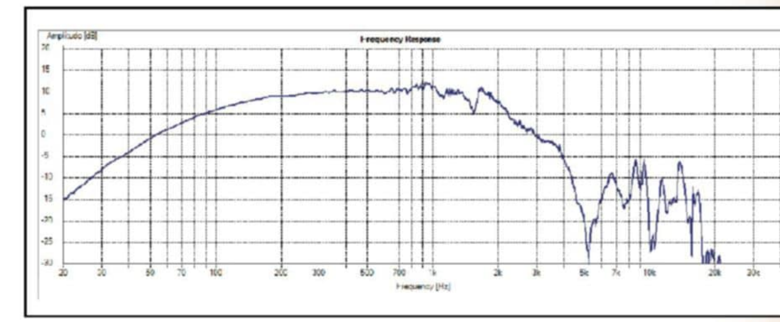


Figure 5: This is the external very-near-field response.

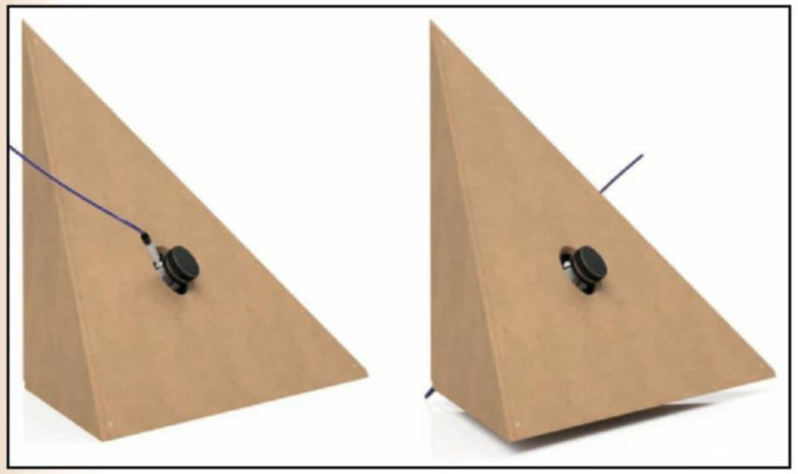
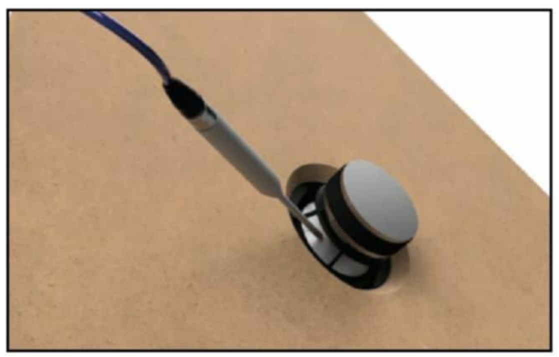


Photo 3: The test chamber is shown during the external near-field microphone measurement and during normal internal measurements.

Photo 4: This image shows a microphone 2 mm to 3 mm from the rear of the cone making the external very-near-field frequency response measurement. Note: The microphone must not touch the cone.



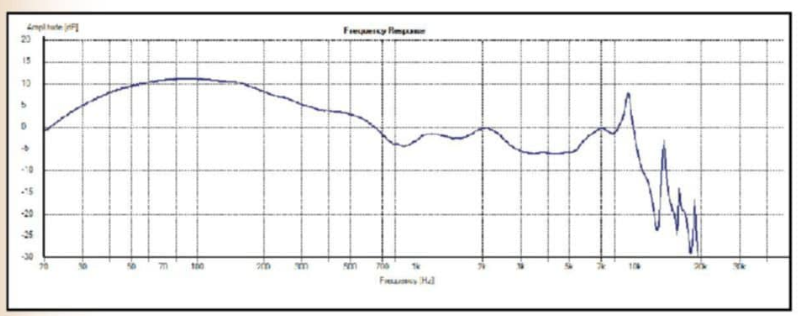


Figure 6: The internal pressure frequency response is obtained as part of Step B—placing the microphone inside the tetrahedral structure.

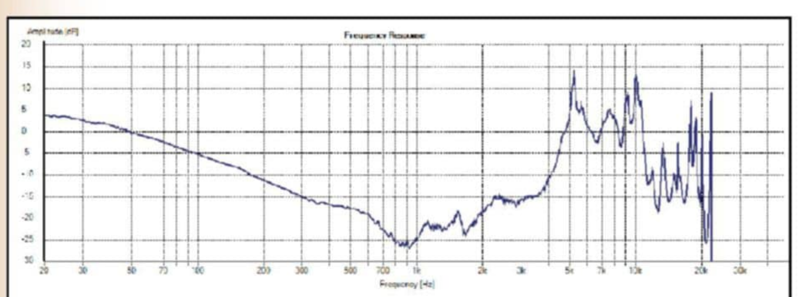


Figure 7: We obtained this difference curve between the two measurements shown in Figure 5 and Figure 6.

the magnet is physically blocking the sound; the cone is not working pistonically at frequencies above 1300 Hz for a driver of this size; and because the front of the cone is not contributing to the high-frequency response. See **Figure 5** for the measurement results.

We save this response and move to Step B by placing the microphone inside the tetrahedral structure. We need to use the microphone setting jig placed inside the chamber to guarantee the critical mic to baffle distance, which ensures the repeatability of the measurements (see **Photo 3**). Then, we make a conventional pressure measurement inside the airtight tetrahedral structure (see **Photo 4**), which yields the true high-frequency response. However, the low frequencies are boosted because of the pressurized environment, which raises the sound pressure level. This is shown in **Figure 6**.

In Step C, we create a difference curve between the above two measurements as shown in **Figure 7**. And for Step D, we create the correction curve from the difference curve in Figure 7. Keep in mind when creating the correction curve that above 1300 Hz, the difference is meaningless and the difference is set to the last valid value (see **Figure 8**).

In the final step, Step E, we apply the correction curve to the internal measurement resulting in an accurate, repeatable, and transferrable loudspeaker measurement. **Figure 9** shows the resulting frequency response of the loudspeaker in the chamber.

With this simple but effective loudspeaker measurement system we have a solution for the new normal and into the future.

Project Files

To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Resources

International Electrotechnical Commission (IEC) standards: Sound system equipment. Acoustical (output-based) measurements, IEC 60268-21-2018; Sound system equipment - Electrical and mechanical measurements, IEC 60268-22 (Draft due 2020).

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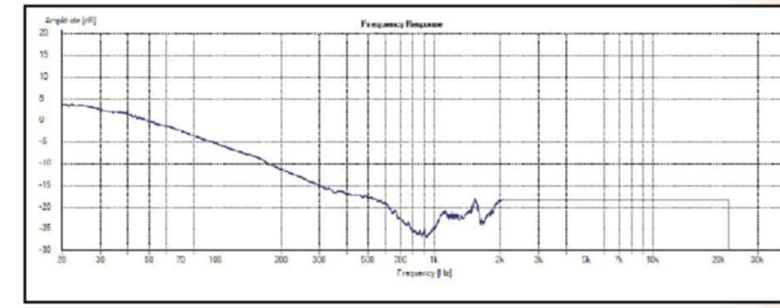

Figure 8: To create the correction curve, we ignore information above 1300 Hz.

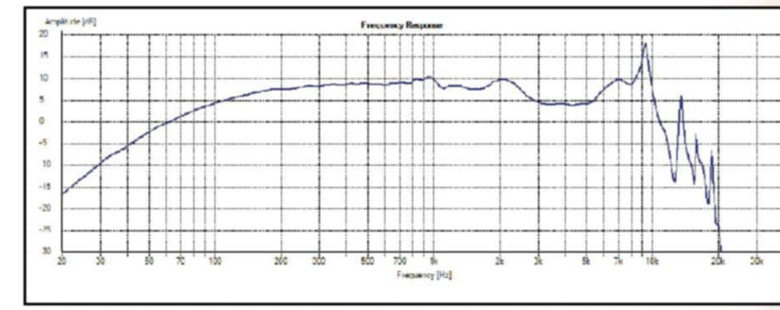

Figure 9: Applying the correction curve, we obtain the frequency response of the loudspeaker in the chamber.

Fast and Reliable End-of-Line Acoustic Defect Detection

This article discusses end-of-line testing, a crucial step in production, which requires very specific procedures and measurements. Specifically, the authors focus on testing irregular acoustic defects (also known as Rub & Buzz) to reduce the test time and optimal solutions to speed automatic testing in a production environment.

By
Stefan Irrgang and Wolfgang Klippel
(Klippel GmbH)

The primary goal in end-of-line (EoL) testing is to reliably separate faulty from fault-free devices under test (DUT) and to minimize false positive or false negative verdicts. However, some defects only generate minor symptoms, which might not be audible or critical during the EoL test but could become unacceptable during product life in the final application and should not be shipped to customers.

While acoustical, electrical and sometimes mechanical checks of specified properties such as frequency response or speaker parameters (Thiele-Small, stiffness asymmetry, and voice coil rest position) are mandatory for comprehensive testing, they are discussed in greater detail in the articles mentioned in our References [1], [2], [3]. This article focuses on testing irregular defects (also known as Rub & Buzz) [4], which in most cases can only be detected by acoustic measurement, because detecting these defects, unlike fundamental or harmonic testing, is the most limiting factor in reducing the test time. This article investigates the physical constraints limiting sensitivity and speed of automatic testing in a production environment and searches for optimal solutions that can be realized with minimal effort [5].

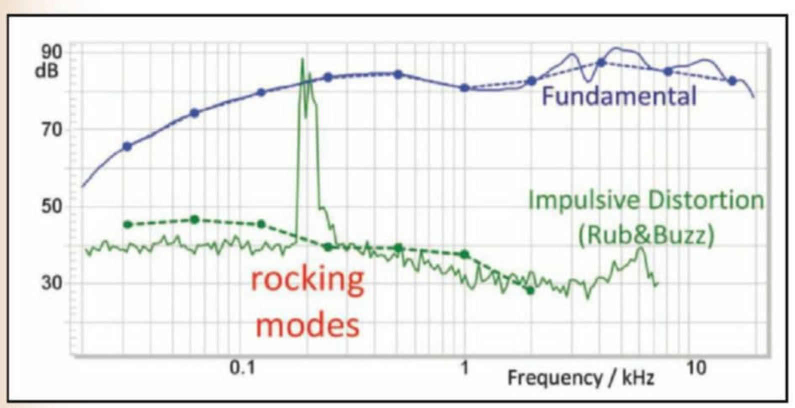

Figure 1: Frequency response of SPL fundamental component and impulsive distortion measured with a stepped sine stimulus (dashed line) and continuous log chirp with speed profile (solid line) using the same total stimulus length of 200 ms.

Particularities of EoL Testing

Typical measurements performed during product development are usually not time restricted and are applied to selected samples only. A high signal-to-noise ratio (SNR) can be achieved by averaging the monitored signals (noise suppression) and by using an IEC baffle in an anechoic and climate-controlled environment.

The production environment has restrictions that need to be addressed to make the testing of all DUTs (100% testing) feasible. Those restrictions include a short test time, and the fact that the properties of DUT may change over time (e.g., glue not completely dried, higher temperature from drying process, break-in effects, etc.). Further restrictions include the need to use a measurement microphone in the near field, test boxes that provide ambient noise shielding, and differing conditions at EoL test stations.

Efforts must be made to suppress undefined conditions and influences that limit the reproducibility of the results such as clamping and positioning of DUT and sensors, handling by human operators, acoustic load changes (box leakage) and connection problems. Variable conditions can also cause verdict (Pass/Fail) corruption due to external acoustic or mechanical disturbances, as well as significant temperature/humidity variations.

Repeatability and reproducibility of test results should be verified. For EoL testing, this is more important than comparability with standard R&D measurements. Responses from the DUT are captured and analyzed, extracting characteristics for Pass/Fail classification.

Optimum Stimulus

The stimulus is crucial for fast EoL testing. A chirp is the most popular stimulus for acoustic testing in manufacturing.

Steady-State Measurements

The most accurate way of measuring a DUT's behavior is through steady-state measurements. The acquisition begins when the amplitudes of all state variables (e.g., pressure, excursion, and current) are settled and constant. The settling time depends on the resonance frequency and quality factor of the fundamental and other higher-order modal resonators (cone breakup modes).

For example, a subwoofer operated in a sealed enclosure generating a quality factor Q0 = 1 and resonance frequency f0 = 50 Hz must be excited for 40 ms until an amplitude accuracy of 0.2% is reached. In a vented enclosure, the higher quality factor Qp = 10 of the port resonance of 50 Hz would increase the pre-excitation time to 400 ms. Instead, allowing an error of 4% in the measured amplitude would cut the pre-excitation time in half, an exchange that is accepted and inconsequential in EoL testing.

Stepped Sine Stimulus

The stepped sine stimulus contains multiple steps, each containing full oscillations at fixed logarithmically spaced frequencies. With knowledge of the maximum quality factor found in the regular and irregular modal resonances of the DUT, the optimum frequency spacing and number of periods for every step could be calculated that ensures every critical resonator is properly excited. In practice, the minimum period number is between 2 and 5.

The minimum number of excited frequencies in one octave is the most critical requirement for exciting resonators by stepped sine stimuli. If too low, a defect in the transducer could be missed. This is illustrated in a practical example in the next section.

Logarithmic Chirp

A continuous sinusoidal chirp is defined as a sine-based signal with continuously changing frequencies using a constant sweep speed. The benefits of a chirp will be illustrated with a woofer with two rocking modes near 200 Hz having a high quality factor Q ≈ 25. These generate impulsive distortion due to voice coil rubbing.

Figure 1 shows the result of a measurement in which both the stepped sine and the chirp stimulus have the same length of 200 ms. The chirp signal excites all frequencies and provides symptoms of voice coil rubbing in the impulsive distortion (ID) at high resolution [6] [7]. The frequency spacing of the stepped sine stimulus at this speed is unable to sufficiently excite the rocking modes. In fact, the optimum frequency spacing and period number for this woofer results in a stimulus length of 4.4 s (20 Hz to 20 kHz). Furthermore, the low number of excitation tones also severely limits the frequency responses' resolution.

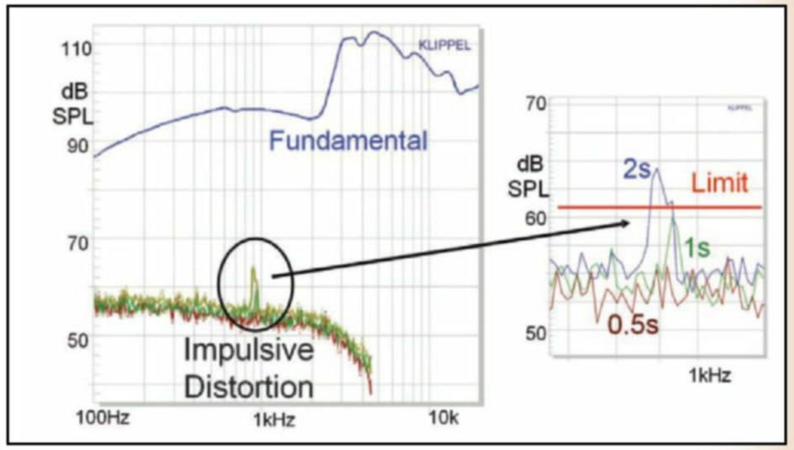

Figure 2: Impulsive distortion generated by a parasitic resonator (defect) with high Q-factor in a headphone measured with chirps of different lengths (0.5 s, 1 s, and 2 s)

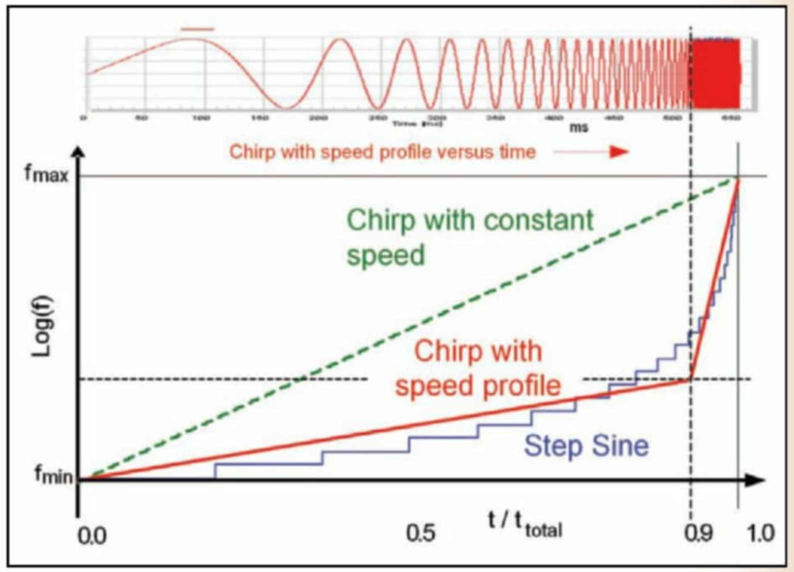

Figure 3: Time-frequency mapping of a stepped sine sweep, a logarithmic chirp with constant sweep speed, and a logarithmic chirp with two different sweep speeds

While the chirp is more effective for fast EoL testing, a defect could still be missed if the chirp is too fast. The relationship between sweep speed and amplitude of the resonator response is illustrated in **Figure 2**. A loose part generates parasitic vibration in a headphone at 900 Hz with a high-quality factor of Q > 20. With a 2 s long chirp from 100 Hz to 20 kHz, the critical vibration of the resonator can be excited and detected as impulsive distortion. Doubling the sweep speed to 1 s reduces the symptom by approximately 4 dB. At 0.5 s length, the symptom is covered by measurement noise. Increasing the stimulus amplitude in the critical resonator frequency band (applying frequency dependent amplitude shaping) by 3 dB can partly compensate for the reduced energy resulting from doubling the sweep speed.

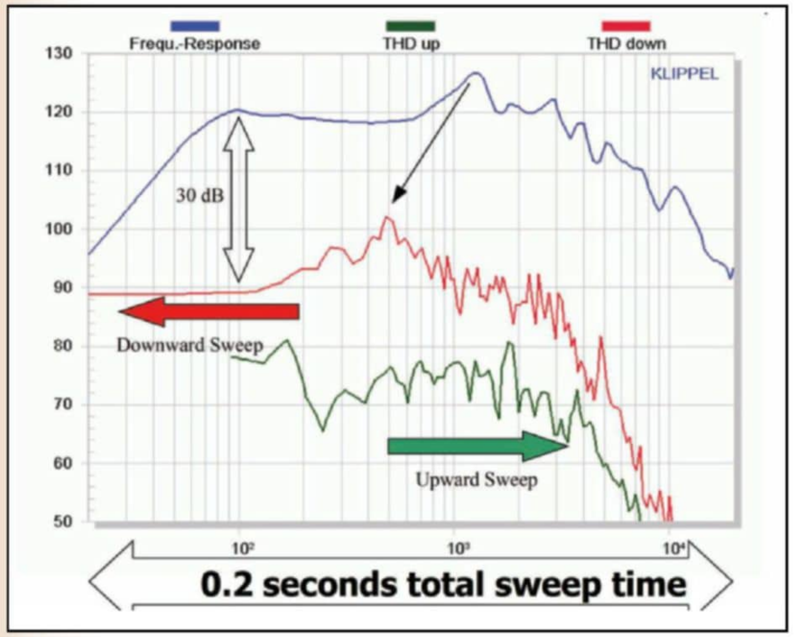

Figure 4: Errors caused in the harmonic distortion of DUT with modal resonances using a chirp sweeping downward in 200 ms

About the Authors

Stefan Irrgang studied electrical engineering at the University of Technology in Dresden, Germany and received a Ph.D. in Technical Acoustics in 1997. In 1998, he joined Klippel GmbH as one of the first employees. Since 2016, Irrgang is the company's head of development for measurement applications. His team provides unique test solutions for lab use, quality assurance, and product validation to the audio industry. His special interests are signal processing and test optimization. He gives lectures and workshops on loudspeaker testing, diagnostics, and optimization.

Wolfgang Klippel studied electrical engineering at the University of Technology in Dresden, in the former East Germany, where his initial studies focused on speech recognition. Afterward, he joined a loudspeaker company in the eastern part of Germany where he was engaged in transducer modelling, acoustic measurement and psychoacoustics. He later returned to his studies and received a Ph.D in Technical Acoustics in 1987.

After spending a post-doctoral year at the Audio Research Group in Waterloo, Canada and working at Harman/JBL in Northridge, CA, he returned to Dresden in 1997 and founded Klippel GmbH, a company that develops novel control and measurement systems dedicated to loudspeakers and other transducers.

Klippel has also been engaged as Professor of Electro-acoustic at the University of Technology in Dresden since 2007. His papers and tutorials on loudspeaker modelling and measurement—particularly those on large signal behaviour and physical distortion mechanisms—are considered reference works in the field.

Variable Sweep Speed

Figure 3 compares a chirp signal with constant sweep speed (dashed line) with a stepped sine wave (stair-case line) within the same total measurement time. The slope of the chirp in the time-frequency mapping at low frequencies is greater than that of the stepped sine but smaller at higher frequencies. The optimal sweep speed is limited by the maximum quality factor at lower frequencies. This slow speed is not required at high frequencies and unnecessarily increases the measurement time.

For fast testing of electroacoustical devices, the ideal stimulus is a combination of the dense excitation inherent in the chirp while using the time-frequency mapping of the stepped sine shown in Figure 3. Thus, the sweep speed is not constant but rises with higher frequencies. This is the basis for ultrafast testing at the physical limits. The chirp would be fastest if the sweep speed was continuously changing. In practice, only two sections of different but constant speed are required to get a sufficient approximation, shown as a thick solid line in Figure 3. This technique is implemented in the Klippel QC system [8]. For a full audio band chirp (20 Hz – 20 kHz) with a sweep speed above 1 kHz that is five times higher than at lower frequencies, the test time is reduced to 53% of a traditional chirp with constant sweep speed.

Sweep Direction

The direction of the sinusoidal sweep is crucial in fast EoL tests because the ringing of modal resonances generates artifacts in the harmonic distortion measurements. For example, **Figure 4** shows that sweeping downwards within 200 ms generates 12 dB more distortion at 500 Hz than sweeping upwards. The modal resonance at 1 kHz corresponds with the THD maximum at 500 Hz in the downward sweep due to the measured second-order component. By sweeping downward, the modal resonators in the DUT, box or room will be excited and the post-ringing generated by high quality factors will be interpreted as harmonic and higher-order distortion. Sweeping upward gives the correct total harmonic distortion (THD) values that match the results of slower steady state and chirp measurements by generating the harmonic components before the excitation causes ringing at the modal resonances in higher frequencies.

Another benefit of sweeping upward is that initial low frequencies help break-in the transducer when operated for the first time after production. An additional low-frequency, high-displacement signal can be used before the first measurement for not only breaking-in the transducer but also for settling the response for the low start frequency.

Production Noise

R&D tests are performed in well-defined conditions without significant external disturbances—this is not the case in the production environment. Acoustic and structure-born disturbances are unpredictable and are in the same magnitude as the defect symptoms being detected. There are multiple strategies for coping with production noise.

Typical passive solutions include well-damped test enclosures that attenuate disturbances up to

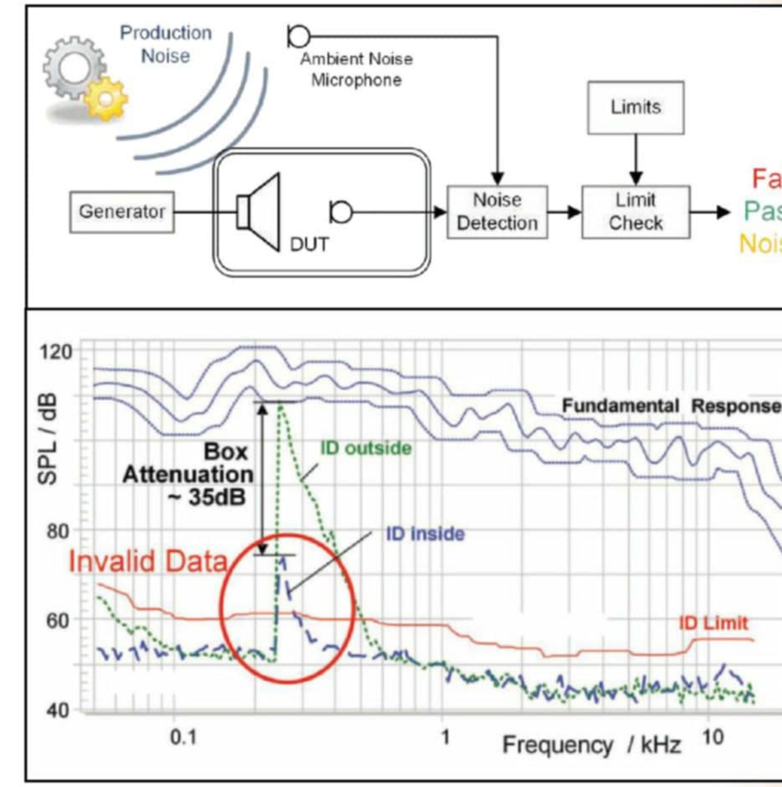

Figure 5: Detection of invalid data is found by comparing the impulsive distortion (ID inside) at the test microphone with the impulsive distortion (ID outside) at the ambient noise microphone located outside of a well-designed test enclosure.

40 dB. While this is sufficient for amplitude frequency response, and usually also THD, it is insufficient for impulsive distortion.

Figure 5 shows a corrupted measurement caused by an impulsive ambient noise (e.g., dropped part) outside a well-made test enclosure with almost 35 dB attenuation. However, this disturbance generates ID at the test microphone that exceeds the allowed limit by 15 dB and would cause a false Fail verdict.

Thus, in addition to insulation, further steps are required. Simple approaches such as an automatic repetition of failed tests do not reliably determine the root cause (defect or production noise). The repeated measurement may also be disturbed, so there is no reliable prevention of false verdicts.

Using additional mechanical or acoustical sensors in parallel to the test microphone, as shown in Figure 5, to monitoring ambient noise can identify corrupted data. Since ambient noise is usually random, the number of test repetitions can be minimized by merging valid parts of several partially corrupted repetitions together to form a completely uncorrupted data set. More details and available products can be found in the References [2] [9].

Timing in EoL-Testing

A complete EoL test cycle consists of:

1. Positioning the DUT on the test station
2. Fixing the DUT and connecting
3. Excitation of the DUT and signal acquisition
4. Release of the DUT
5. Moving the DUT out of test station, proceeding with step 1

Initial setup and product changeover are ignored in this discussion since they are usually negligible relative to the total test time for large production batches.

The actual measurement (point 3) is defined from the trigger of the test (e.g., a barcode scan or a hard- or software switch) until the last captured signal sample. The final verdict may appear slightly later without consequences as long as the start of the next test is not delayed.

The order of the measurement tasks or steps also influences the total measurement time. It is useful to start with the task that requires the largest processing load and use the acquisition time of the following task for finishing the calculation of the previous task. When using a chirp, some processing can even be started before the acquisition is completed.

Modern test systems can capture sound pressure, voltage, current, and displacement in parallel, so checking some electrical or mechanical properties might not increase the measurement time.

Conclusions

The stimulus properties limit the speed of the measurement and sensitivity for detecting defective units [10]. Adjusting the stimulus to the transient behavior of the DUT through frequency dependent speed and amplitude shaping is required for minimizing the error in the PASS/FAIL decision. The chirp with rising sweep speed at higher frequencies is the optimal stimulus for speeding up EoL testing. The time savings provide interesting opportunities for manufacturing.

Partly or completely repeating measurements is the best way to cope with a high probability of random ambient noise that cannot be completely attenuated by passive means. An optional second measurement may also be useful for verifying inconsistent defects like loose particles in case of suspicious symptoms (e.g., just below limit threshold) in a first measurement. This may significantly reduce the amount of costly field rejects. The saved time can also be invested in additional test steps for nonlinear parameters [1], [3] or other criteria [4], which are helpful for deeper diagnostics.

Machine learning and defect classification can reveal the root cause of a failed unit, automatically assigning it to a known defect class [2]. The more independent aspects of the DUT that are measured, the better these processes work. If a failed unit does not fit into an existing defect class, an operator can investigate it at a diagnostic station close to the production line by performing additional tests, listening to its acoustical output and dissembling it to find visual clues for the root cause.

Thus, the smooth combination of a fast EoL measurement system with machine learning, automatic classification and further testing at a diagnostic station generates a learning process in manufacturing and engineering. This helps maximize the yield rate, improve product reliability, and design future products with higher benefit-cost ratio that are easier to manufacture.

Resources

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How Virtual Product Development Improves the Quality of Consumer and Pro Audio Products

This article addresses how modern methods of acoustical virtual product development make it possible to assess and optimize quality and performance problems throughout the entire product development process. Using a combination of modeling, simulation, and measurement tools, everything can be analyzed in-depth before building the physical product, the two authors explain.

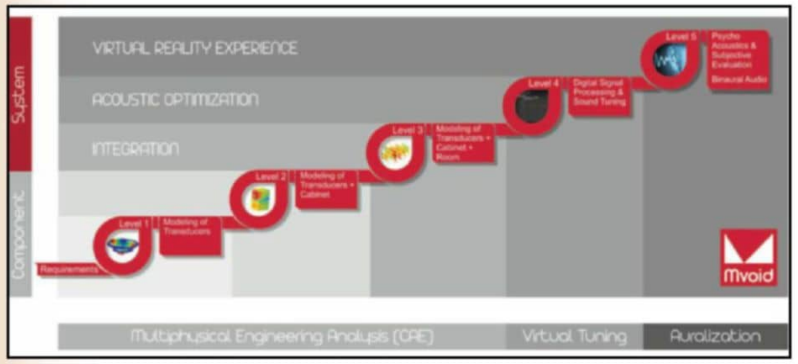
By
Alfred Svobodnik, CEO and President
and Thomas Gmeiner, Director of Engineering
(Mvoid Group)

All images courtesy of Mvoid Technologies GmbH

The big day has arrived: The prototype of the newly developed transducer is mounted into the loudspeaker cabinet for the first time. The loudspeaker is connected for a listening test. The test reveals unexpected rattling noises. Possibly an asymmetrical air load on the transducer causing a rocking mode or unwanted structural vibrations. This means that the product developers have to review the design for the next iteration step: They have to create a new model.

Running those tests during simulation stages could be extremely beneficial for everyone involved and even more if the results could be audible via headphones. Otherwise any modifications become expensive and time-consuming. Acoustical virtual development enables users to feed back the experience gained from the virtual listening test into the multiphysical simulation, make the necessary adjustments, and then use the newly calculated model for the next listening test, which can be shared between remote locations. Production and development costs can be saved as prototypes are designed, while products are improved at the same time.

The development of consumer and pro audio products is an increasingly complex and costly process. The faster incorrect approaches are rejected or modified, the faster a product reaches the market—and the lower the development costs. Acoustical virtual product development methods provide valuable help in optimizing the development of consumer and professional audio products while optimizing quality at the same time.



There are five tier levels of Mvoid methodology for a complete virtual product development environment.

Acoustical Virtual Product Development

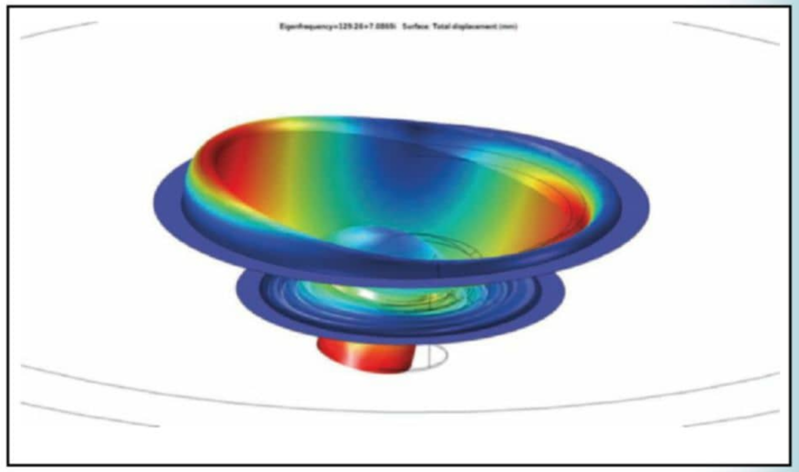
Despite the experience and long-established practices in the development of loudspeakers and audio systems, acoustic experts are always faced with new challenges. Unexpected noises and unbalanced sound reproduction must be analyzed, evaluated, and eliminated at an early stage. The earlier virtual methods are used in the product development process, the greater the advantages with regard to three essential objectives: quality, time, and cost.

Acoustical virtual product development methods enable designers to turn loudspeakers and audio systems into a realistic experience for the user—without the need for any physical object. The multidisciplinary simulation of a loudspeaker, as previously described, is only one possible application.

Accessing the deepest corners of cabinets and even listening rooms, experts can analyze measurement points in the virtual domain, which are not accessible in the real world. This allows analysis and evaluation of undesired results that aren't otherwise possible, leading to completely new development approaches. Various materials, changes in shape or a specific directional characteristic can be analyzed effortlessly in a short time without any prototype.

Every aspect of the system from sub-components through the cabinets, the ducts, the ports or the horns can be designed, auralized, and optimized. Multiple virtual prototypes can be modeled, assembled as systems, evaluated with virtual tuning and be validated in real acoustic venues through auralization.

Thanks to virtual mock-ups, decision-makers receive faster and better feedback from members of the development team, as the audible models provide a more realistic impression even for those at management level who are less familiar with design. The audible model of a new product makes it easier for team members to form opinions and it speeds up the decision-making process. Suggestions for improvement can be incorporated before even a single real model is produced. As a result, the number of iteration cycles during product development is reduced, while product quality is improved.



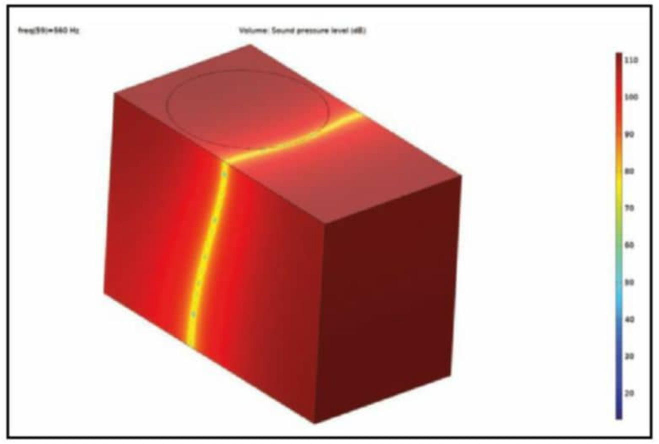
A loudspeaker's unwanted oscillation (rocking) is analyzed by means of 3D multiphysical loudspeaker model.

Many Factors Influence the Sound

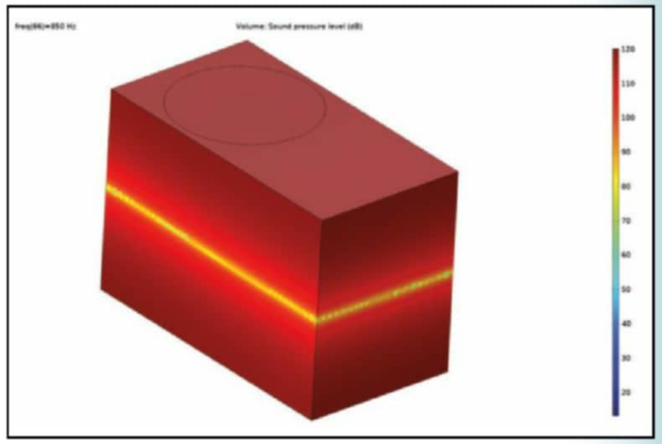
In order to create a holistic virtual model, it is necessary to examine the complete system. In doing so, it is necessary to consider all involved subsystems and to investigate their reciprocal interaction as it will demonstrate their impact. In a listening situation, the overall system is significantly more than a loudspeaker—the loudspeaker, potentially in an array, room acoustics, and the perception of sound by humans.

To be able to listen to an audio system based on a virtual prototype, the following engineering disciplines need to be added to the analysis tools—digital signal processing, psychoacoustics, subjective evaluation, sound tuning, and auralization (binaural audio).

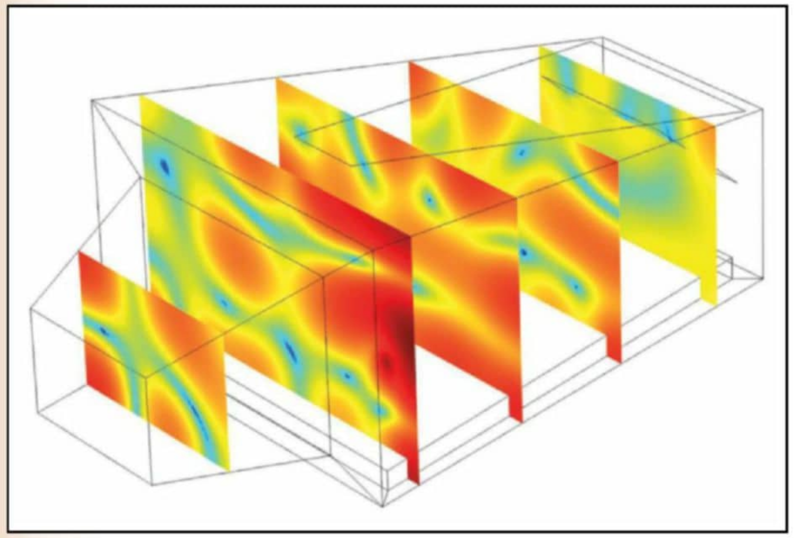
A step-by-step approach has proven successful, combining five different multidisciplinary levels. When all five levels are applied, a complete virtual product development environment is created, which allows a realistic listening experience of the virtual audio system at any time of the project. The methodology has been used for many years, especially in the automotive industry and is currently being adopted by pro and consumer R&D labs.



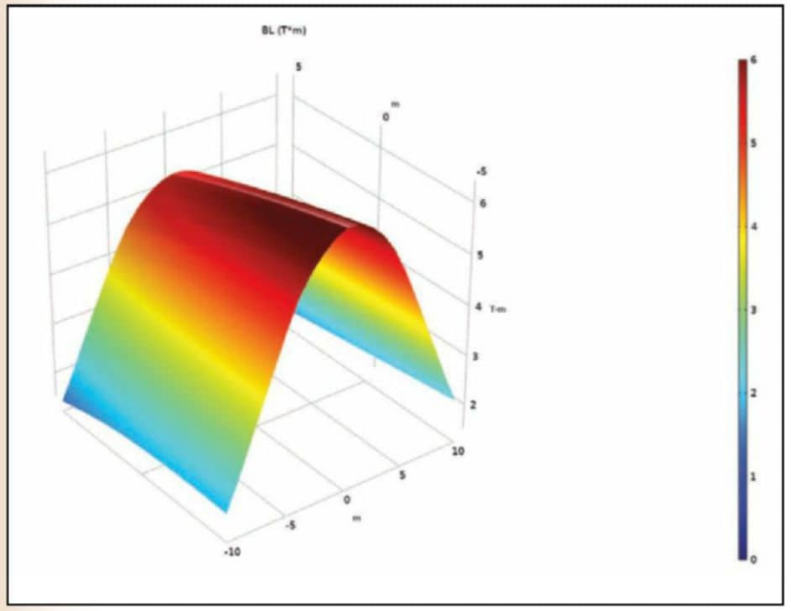
The multiphysical loudspeaker model includes the cabinet and the sound; cabinet resonance at 560 Hz.



The multiphysical loudspeaker model includes the cabinet and the sound; cabinet resonance at 850 Hz.



Multiphysical modeling of transducers, cabinets, and room is shown; sound pressure distribution at a cross section.



Example of a graph representing the force factor depending on two variables.

About the Authors

Dr. Alfred J. Svobodnik is President & CEO of the Mvoid Group. He is an entrepreneur, thought leader, engineer and scientist. He has been researching for 30 years in the areas of Multiphysics and virtual as well as computational acoustics. In 1990 Alfred completed his doctorate at Vienna University of Technology with the degree "Doktor der Techn. Wissenschaften" (equivalent to PhD).

Thomas Gmeiner leads Mvoid's business unit Professional Audio. Thomas has more than 22 years of experience in acoustics, especially in audio engineering, business to technology development, microphones, headphones, digital wireless transmission and loudspeakers. Thomas holds two master's degrees, one in electrical engineering from University of Technology Graz, Austria, and one in acoustics from Penn State University in Pennsylvania.

Multiphysical Simulation Modeling of Transducers

Within the first tier level, the goal is to create the multiphysical model of the transducer. The greatest challenge lies in analyzing the different physical domains, as each domain interacts bi-directionally with each other—electromagnetism (motor/drive system), structural dynamics (mechanical vibration system), acoustics (surrounding air), and sometimes also heat conduction and flow—since these are strongly coupled with each other. In addition, nonlinear effects that lead to disturbing distortions must be considered. With the support of multiphysical simulations, the acoustic experts are able to analyze interactions of all physical domains. One dimension-lumped parameter models, 2D, 2.5D, and 3D finite element models are used for multiphysical simulation. Moreover, it is possible to generate small and large signal transducer reports that correlate to the physical samples.

Multiphysical Simulation Modeling of Transducers and Cabinets

Loudspeakers require an enclosure to produce a net volume flow at low frequencies. The enclosures and adjacent parts can create unintended noises. The interaction of the loudspeaker and the cabinet must be analyzed.

Practical problems are asymmetric air load on the transducer causing rocking mode, directivity of subwoofers, or structural vibrations. Multiphysical simulation allows examining structural dynamics and acoustics. Acoustics experts are able to quickly detect undesirable directivity effects, noises, or disturbances and find new solutions for enclosure design (e. g., alternative placement of components and alternative cabinet shapes).

A Full-System Model

Multiphysical simulation models are able to determine distribution of sound and reflections in a room. The listening room has an important impact on the perception of the reproduced sound and acoustical comfort. There are many challenges to consider when designing for example a new line array system for touring, a concert hall, or theatre: the size of the hall, the shape of the ceiling, flat or curved walls, the position of the stage itself. All major factors that affect the sound in the interior must be taken into account. Practical problems include the directivity pattern and placement of a loudspeaker for excitation of room modes or directivity of an array for sound coverage in a given space.

Moreover, every space has a different transition band between these regimes:

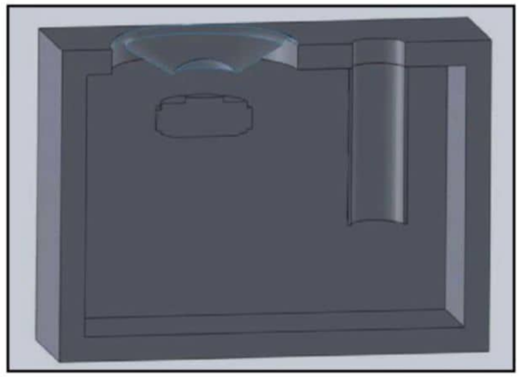
- Low frequency: Modal range
- High frequency: Statistical area
- Transition range depends on the room size

By combining different simulation methods, the audible room acoustic effects can be analyzed. The 3D sound field is determined by physical mathematic parameters. Based on 3D CAD data, the interactions of loudspeakers and enclosures including the listening room are analyzed and evaluated in the multiphysical simulation model.

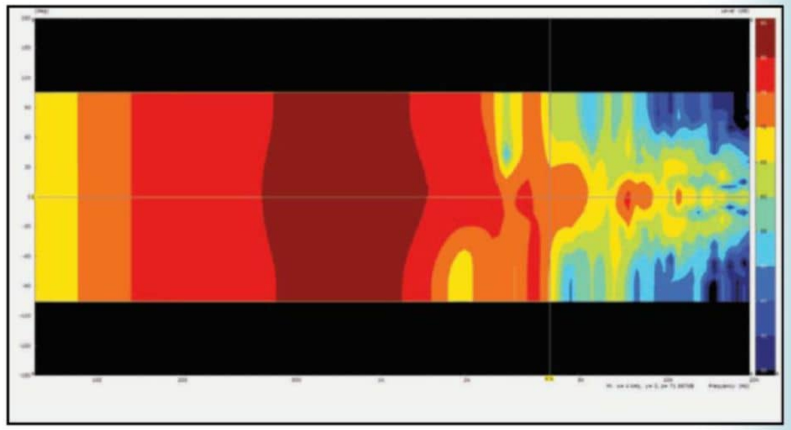
The simulation model allows visualization of all essential factors influencing the sound. The multiphysical simulation predicts the acoustics in a room. However, the multiphysical simulation does not predict the acoustic sound quality of a particular seat, on the basis of the calculated spectrum. This is possible in the next step.

Better Sound Through Virtual Tuning

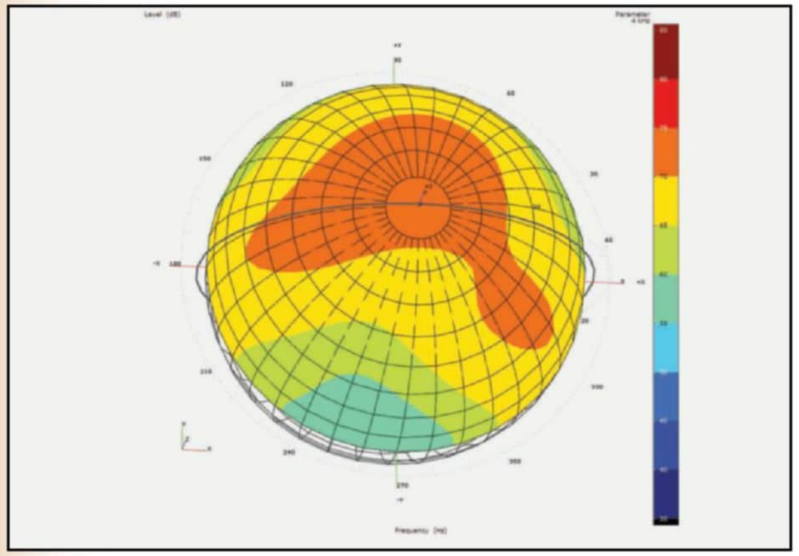
The human ear perceives all loudspeakers in a room as a group. In multiphysical simulation, the spatial attributes of human sound perception are not yet sufficiently taken into account. The analysis of the individual loudspeakers and their frequency



A parametric cabinet design allowing further optimizations.



Isobaric representation of the directivity of a loudspeaker, including the cabinet, is shown.



This is a balloon plot representation of the directivity of a loudspeaker, including cabinet.

behavior alone are not sufficient. In virtual tuning, the overall system is now evaluated on the basis of the simulation data.

Virtual tuning establishes the acoustic sound quality at a specific location. The interactions of the loudspeakers with each other are analyzed. There may be areas within the architecture of a room or venue in which sound waves cancel each other. In addition, irregularities or resonances can occur at certain positions. Certain frequencies may be too loud or too soft for an enjoyable listening experience. Typical problems are caused by resonances in a control room or unbalanced sound in an audience area.

The acoustics experts check whether there are any constructive or destructive influences. They examine whether the individual loudspeaker's characteristics and placements can be influenced in such a way that their acoustic energy adds up optimally to create a sweet spot large enough for the entire audience.

At this stage, the acoustic experts put the complete virtual system through its paces. They predict reliably whether cost-intensive hardware changes will be necessary, such as the replacement of loudspeaker components or an integration adjustment. Or whether the desired sound reproduction is cost-effectively achievable—by means of tuning. To determine the perfect sound experience, they are able to analyze any specific location in the room, and adjust speaker characteristics for the best possible sound in each specific location.

Acoustical Virtual Reality by Auralization

Virtual tuning is supported by auralization—the process of rendering an acoustic situation to make it audible. Auditory perception relies on the physiognomy such as ear shapes, head circumference, and body volume, which differs from person to person—it is not universal. The physical properties of sound arriving at the ears is transformed into a perception by the auditory system. Our ears transform sound waves into electrochemical impulses, which are transmitted from the nervous system to the brain where further processing and evaluation of the signal takes place.

For a realistic virtual listening experience, all important psychoacoustic effects are considered. This includes the detection and localization of incoming sound waves on the left and right ear. The incoming waves to our two ears are diffracted and reflected in different ways due to our anatomy (head, neck, shoulders, and ears) producing two signals that our brain processes into spatial information.

The auralization process makes it possible to hear digital prototypes of audio systems in a specific room, which might exist only on the drawing board, and to assess their product quality. The resulting soundfield is made audible through headphones, allowing acoustics experts to examine the interactions of audio products within a listening environment, evaluating important psychoacoustic effects of their spatial impressions.

To achieve realistic reproduction, calculations are extended with binaural data. The virtual listening environment uses a multiphysics model of the room impulse response as the starting point, determined by the positions where persons are sitting and their ears are located. Additionally, binaural effects caused by diffraction of the incident sound waves at the head and torso are added to these simulation results.

Real-time processing is crucial in the auralization process, in order to create an acoustical virtual reality. This is because any change of any tuning parameter must immediately produce an audible response via headphones and a visual response on the monitor (i.e., graphical representation of the frequency response or impulse response).

The auralization procedure allows acousticians to draw precise conclusions about the sound of the system while in the planning stages, being able to judge them by subjective listening tests—without real prototypes. Sound reproduction faces the ultimate instance—the human ear.

Better Quality, Fewer Prototypes

Modern methods of virtual product development make it possible to assess and optimize quality and performance of consumer and pro audio products throughout the entire product development process.

Acoustical virtual product development enables concept ideas and technical development statuses to be assessed, interpreted, and optimized. Shortcomings and errors, which would become costly in later phases of development and seriously impact scheduling, are detected early. Working in the virtual domain, consumer and professional audio developers and product managers are able to analyze the entire concept, from initial ideas through to the final results.

The experts can safely assess whether the product concept meets the desired requirements. Important development tasks can be carried out using multidisciplinary simulations, and VR applications before prototypes serve as an alternative to production of several physical models. This shortens development times, reduces costs, and improves quality.

Mvoid—A Partner for Acoustical Virtual Product Development

The use of virtual sound systems requires a high degree of know-how in acoustics and multiphysical simulation as well as the combination of advanced tools. Mvoid is a pioneer and expert in virtual acoustics with the focus on simulation of all hardware in the audio chain.

The company guides manufacturers and R&D departments in Acoustical Virtual Product Development, a process that uses advanced multiphysical simulations and combines modeling of all aspects of an audio system, culminating in real-time auralizations that allow actual listening evaluation and validation of design decisions.

Mvoid provides a virtual development environment based purely on computer-generated models and its own VRtool, a virtual tuning and auralization solution where acoustic results can be auditioned over headphones. The VRtool, was fully developed in-house and is one example of Mvoid's innovations.

The company uses other proprietary modules that integrate with industry standard software packages to realize and implement multidisciplinary simulations. The virtual development environment reduces the need for physical prototypes to an absolute minimum and offers the right solution for every step of the development process. The "virtual acoustical reality" provides more efficiency and promotes innovation in the development of advanced audio systems. The development environment is designed to meet customer requirements at any level and scale.

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The APR-17 Aperiodic Loudspeaker (Part 2)

Measurements and Final Details

By
George Ntanavaras
(Greece)

Photo 1: This is the completed version of the APR-17 aperiodic loaded loudspeaker.

In the second part of this article series, George Ntanavaras shares the APR-17's crossover design and discusses the measurements he took to evaluate the aperiodic loaded loudspeaker (see **Photo 1**). He also describes a passive equalizer that he used to boost the loudspeaker's low-frequency response.

In designing the crossover for the APR-17 aperiodic loudspeaker, my two primary requirements were the flat on-axis response and the uniform horizontal polar response. For good spectral balance in a room, the horizontal polar response curves should be smooth replicas of the on-axis response. This is possible up to a limit because of the tweeter's natural roll-off at higher frequencies and larger off-axis angles.

To begin designing the crossover for the loudspeaker, I mounted the woofer and the tweeter in the enclosure and I measured their polar responses. For the measurements, I placed the loudspeaker on a stand in the middle of a large room with the tweeter at 1.35 m above the floor (half of the room's height) and placed the microphone at 0.6 m, giving a gate window measurement of 6.5 ms, which offers a measurement accuracy above 150 Hz.

I used a miniDSP device to equalize the two drivers for a flat frequency response. The woofer was equalized up to 6 kHz, while the tweeter was equalized down to 800 Hz. This gives a more clear view of the capabilities of each driver without the confusion that could be introduced from the irregularities in the frequency response.

The horizontal polar response measurements are shown in **Figure 1**. The woofer's polar response goes very smoothly up to about 2 kHz and then starts to roll off. A small suck-out is shown on the on-axis response around 1.6 kHz, but the dispersion is wider around this region. A mild resonance in the region of 4 to 5 kHz is also evident.

The tweeter's polar response is very smooth up to about 3.5 kHz. A wider dispersion in the response at the region of 1.8 kHz to 3.5 kHz is also evident. Above 3.5 kHz, the response rolls off as expected from its 1" diaphragm until 26 kHz, where a resonance is evident.

I chose a third-order Butterworth filter with the nominal crossover frequency set at 2 kHz for the woofer and 2.7 kHz for the tweeter. The 18 dB/octave slope gives sufficient attenuation to the mild resonance of the woofer in the region of 4 kHz to 5 kHz and protects the tweeter from high excursions at the lower frequencies.

The dimensions of the box's front baffle are 33 cm × 40 cm and according to the Edge diffraction simulator program (see Resources), its frequency response will have a baffle step between 140 Hz and 300 Hz, which will be compensated by the crossover.

I used the LspCAD 6 demo version for the crossover's simulation. The demo version contains all the functionality of the unlimited version with a few exceptions:

- It is not possible to save the projects
- It is not possible to export data
- The crossovers are limited to two drivers

I measured the frequency response and the impedance of each driver with ARTA and LIMP and imported the data to the program as text files. The loudspeaker's total response with the crossover connected to the drivers can be easily computed by the program, and it is instantly displayed. When the value of any component is modified, the result is immediately shown in the frequency response, in the impedance curve, and on many other useful displays.

When I was satisfied with the simulation results, I used a piece of plywood to quickly assembly the prototype crossover. I re-measured the loudspeaker's frequency response to confirm that everything was as the simulation predicted, and I proceeded to the listening tests.

The crossover's final circuit diagram is shown in **Figure 2**. Both drivers are connected in phase. I indicated the internal DC resistance next to the value of the each inductor. I used inductors with a ferrite core and 1.4 mm oxygen free copper (OFC) round-wire that gave a low internal resistance. I used metallized polypropylene capacitors with a very low loss factor and metal-oxide power resistors. I glued them onto two pieces of wood—one for the woofer and one of the tweeter because it was easier to place them inside the enclosure. The completed crossovers are shown in **Photo 2**. In the image, I have shown the woofer on the left side and the tweeter part on the right side. **Table 1** shows the components used for the crossover.

I placed the crossover to the back side of the cabinet using screws. Then I used 1 mm² cable for to connect the woofer and 0.75 mm² cable to connect the tweeter.

Loudspeaker Measurements

When the loudspeaker construction was completed, I continued with the measurements to evaluate its performance. **Figure 3** shows the loudspeaker's on-axis response, the acoustic response of each driver, and the total on-axis response with the tweeter inverted. A mild peak in the woofer acoustic response is apparent at 4.5 kHz but it is almost -20 dB down.

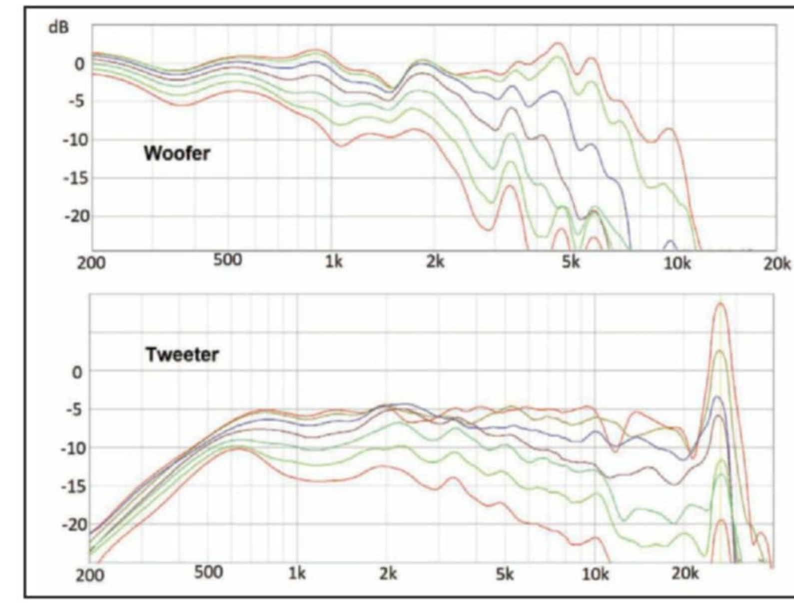

Figure 1: The horizontal polar response of the woofer and the tweeter

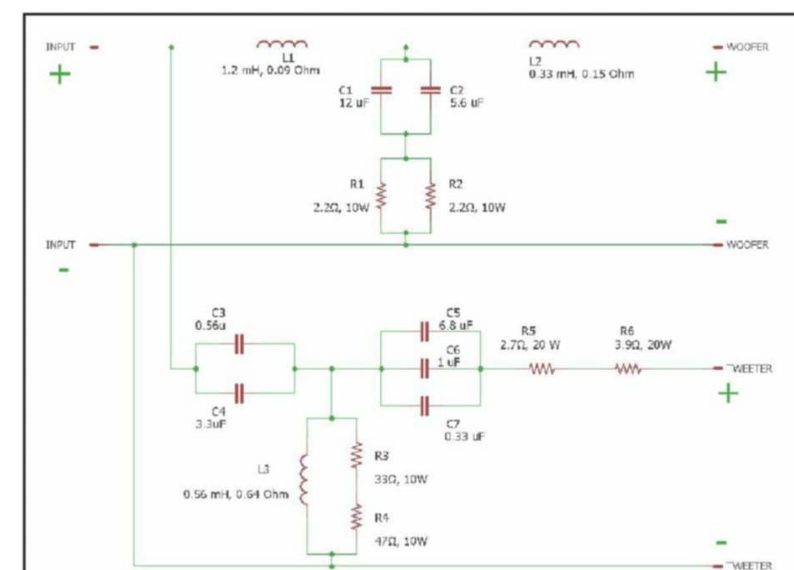

Figure 2: The electronic diagram of the crossover

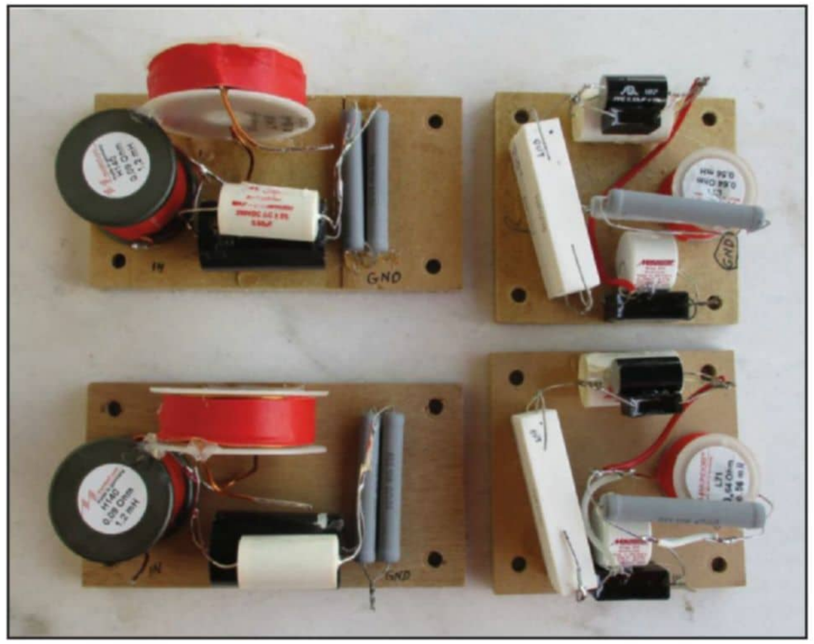

Photo 2: I designed these crossovers to complement the APR-17 loudspeakers.

Components List for the Crossover

Capacitors (All capacitors 250 V, MKP)

C1	12 µF
C2	5.6 µF
C3	0.56 µF
C4	3.3 µF
C5	6.8 µF
C6	1 µF
C7	0.33 µF

Inductors (All inductors Cored with non ferrite material, except as noted)

L1	1.2 mH, 0.09 Ω
L2	0.33 mH, 0.15 Ω, air-coil
L3	0.56 mH, 0.64 Ω

Resistors

R1, R2	2.2 Ω, 10 W
R3	33 Ω, 10 W
R4	47 Ω, 10 W
R5	2.7 Ω, 20 W
R6	3.9 Ω, 20 W

Table 1: These are the components I used for the crossover.

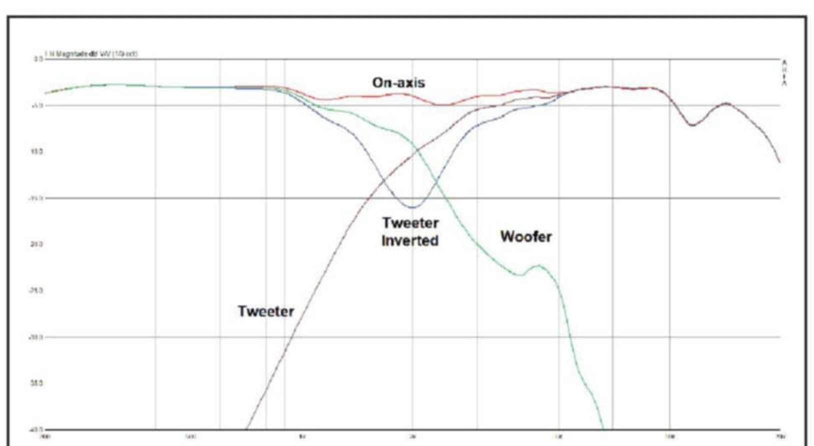

Figure 3: On-axis frequency response of the APR-17 loudspeaker

The phase relationships around the crossover frequency were determined by reversing the tweeter leads and noting the changes in the on-axis frequency response. The notch at the crossover frequency indicates that they are in phase around the crossover frequency.

Figure 4 shows the loudspeaker's polar responses. For the horizontal response, I took measurements on-axis and at 15°, 30°, 45°, 60°, 75°, and 90° off-axis. The response is smooth except in the region around 2.5 kHz, where the loudspeaker shows wider dispersion.

For the vertical polar response, I took measurements on-axis and at 5°, 10°, and 15° above and below the axis of the loudspeaker. The response is very good, except for a small suck-out that develops in the crossover region at 15° above the measurement axis. This means that the loudspeaker's tonal balance will not be affected if the listener has his ears level with or just below the tweeter.

I measured the impedance using the LIMP software. Its magnitude and phase are plotted in **Figure 5**. The aperiodic loading of the woofer results in a more uniform load at the frequencies below 200 Hz, where the magnitude drops to a minimum 3.7 Ω at the region of 170 Hz to a maximum of 8 Ω at the region of 55 Hz while in the same region, the phase varies from +24° to -20°. The maximum impedance value is 19 Ω at 2.1 kHz.

Figure 6 shows the near-field frequency response of the woofer and the aperiodic vent. The response of the aperiodic vent was scaled down by -4.55 dB since the woofer's radiation area is 129 cm^2 and of the aperiodic vent 76.4 cm^2. The band-pass response of the aperiodic vent is centered at 110 Hz with the -3 dB points at 63 Hz and 168 Hz. The woofer response falls down with a rate about 7.5 dB/octave similar to that of the aperiodic vent.

After building the loudspeaker and just before I finished writing the article, I discovered another interesting method to measure the loudspeaker's low-frequency response. The method is suggested in the Klippel Application Note AN39 and concerns the measurement of the Bass reflex loudspeakers (see Resources). The microphone is placed on the axis between the driver and the port, where the sound pressure contributions of the driver and the port cancel out each other at the very low frequencies. The microphone is placed in several locations until the sound pressure is minimal at the very low frequencies (f < 20 Hz). At this location, the sound pressure components generated by the driver and the vent have the same amplitude but opposite phase can cause a dip in the sound pressure response.

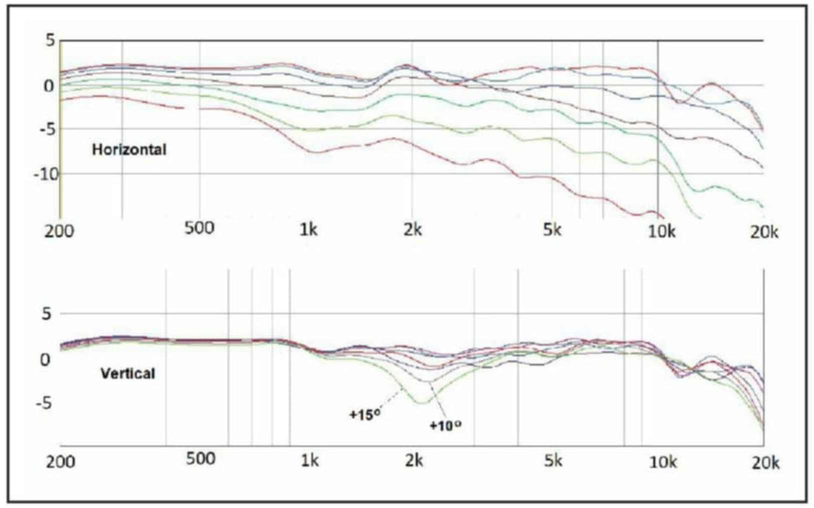

Figure 4: The horizontal and vertical polar responses of the APR-17 Loudspeaker

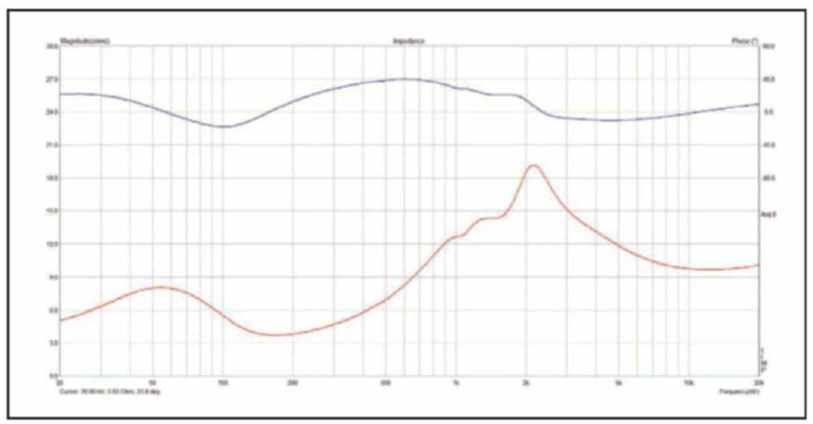

Figure 5: The impedance of the APR-17 loudspeaker

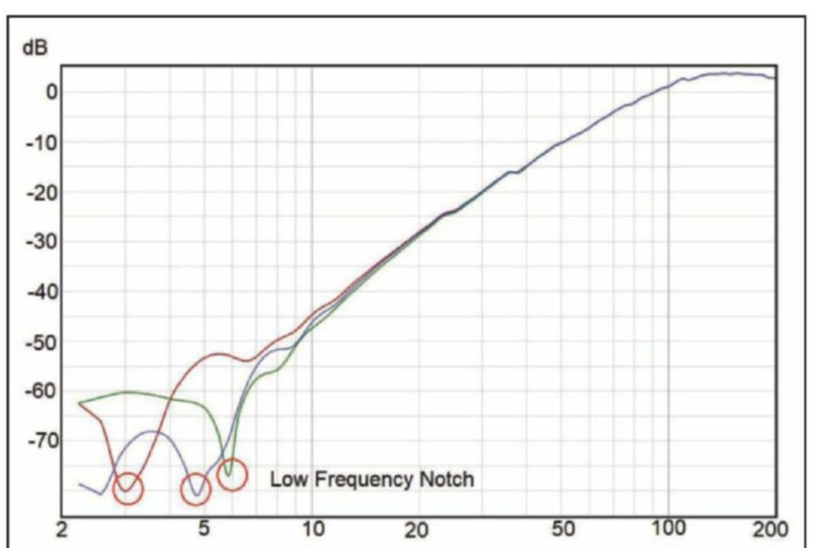

Figure 7: Low-frequency response measurement of the loudspeaker

I measured the APR-17 loudspeaker's low-frequency response, using the Klippel method. **Figure 7** shows three measurements that I took with the microphone at different positions, which were very close to each other (within 2 mm to 3 mm). The location of the microphone that will give the best notch requires some experimentation to be found. All the responses are very similar down to 20 Hz. As the green curve in Figure 7 shows, the slope of the response at the low frequencies approximates 18 dB/octave

Measurements with an Accelerometer

I used my accelerometer system to perform some other interesting measurements. The system consists of the ACH-01-03 accelerometer, an amplifier that also computes the velocity and the displacement, and an external power supply adapter. It is an improved version of the design that I presented in *audioXpress* magazine and is now posted online on the audioXpress website (see Resources).

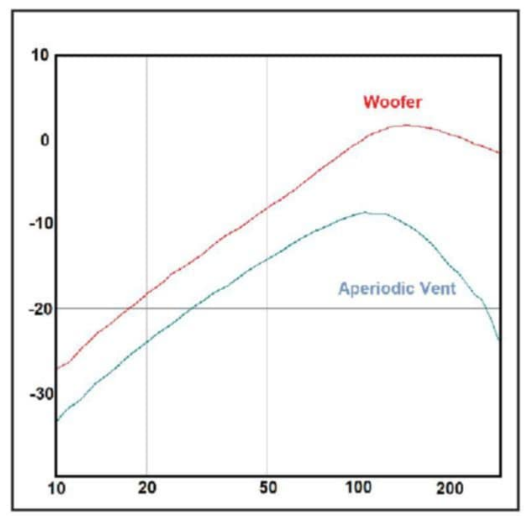

Figure 6: The near-field responses of the woofer and the aperiodic vent

Resources

ARTA & LIMP software, www.artalabs.hr

EDGE program, Tolvan Data, www.tolvan.com/index.php?page=/edge/edge.php

Klippel Application Note AN39, www.klippel.de/fileadmin/klippel/Files/Know_How/Application_Notes/AN_39_Merging_Near_and_Farfield_Measurements.pdf

LspCAD program, IJData, www.ijdata.com/LspCAD_demo.html

G. Ntanavaras, "Accelerometer Testing of Loudspeaker Drivers," *audioXpress*, June 2011, http://audioxpress.com/article/Test-your-speakers-performance-with-this-do-it-yourself-measurement-system.html

The vibration of the loudspeaker cabinet was measured by attaching the accelerometer to the cabinet's panels and using the ARTA software. The excitation was a pink noise signal and a power amplifier drove the loudspeaker with this signal at a high sound level. It was necessary to use an additional gain of 20 dB on the accelerometer preamplifier because the signals measured by the accelerometer were very low in level. The additional gain was then compensated using the "Scale" function of the ARTA software by -20 dB.

Figure 8 shows several measurements. The top curve is the reference and it was taken with the accelerometer attached to the center of the woofer's cone with a very thin double sided tape. The other five curves are vibration measurements of the loudspeaker box's sides as described in the figure. They are almost 50 dB lower than the reference, except around 440 Hz where they are around 30 dB to 35 dB lower.

I used the accelerometer to measure the woofer's cone excursion and to compare the aperiodically

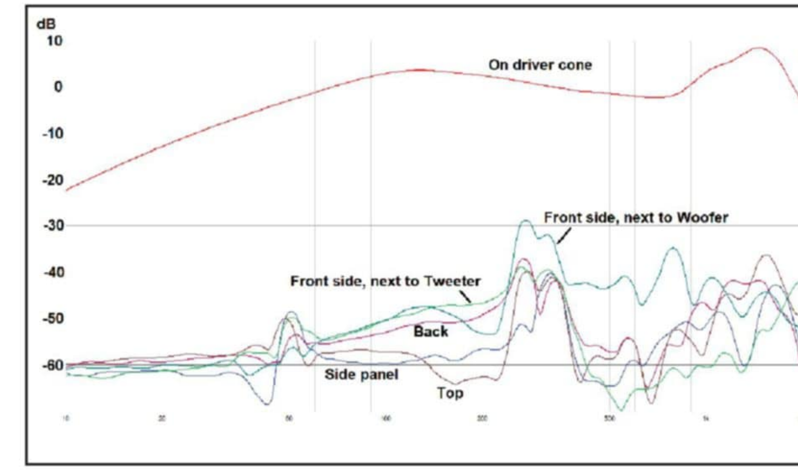

Figure 8: Vibration measurements

loaded box and the closed box as shown in **Figure 9**. The aperiodic has lower excursion in the region between 45 Hz to 150 Hz. Below 45 Hz, the excursion of the aperiodic loaded woofer increases while the excursion of the closed box remains almost constant. This means that the aperiodically loaded woofer can be overloaded more easily if the program contains very low frequencies below 40 Hz at a high level.

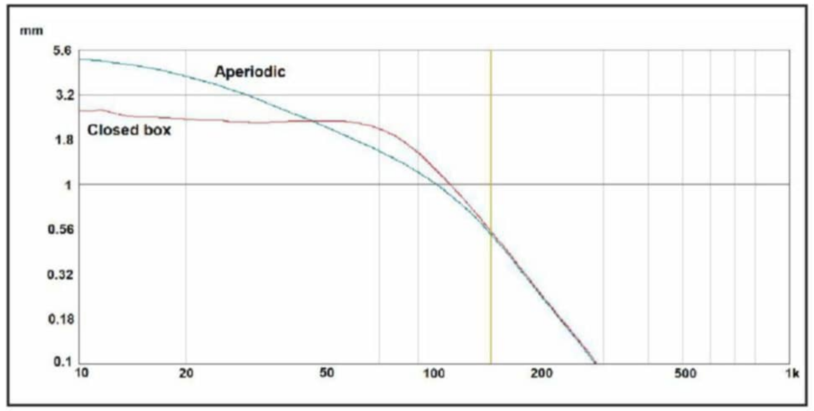

Figure 9: The woofer's cone excursion measurements

Figure 10: Distortion measurements at the low frequencies

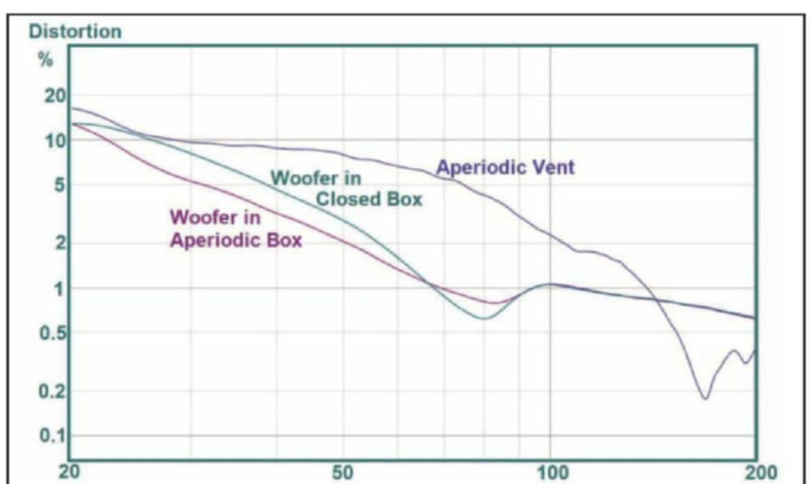

Figure 11: Distortion measurements with the accelerometer

About the Author

George Ntanavaras graduated from the National Technical University, Athens, Greece, in 1986 with a degree in Electronic Engineering. He currently works as a technical consultant for Greek electronics companies. He is interested in the design of preamplifiers, active crossovers, power amplifiers, and most loudspeakers. He also enjoys listening to classical music.

Figure 10 shows a comparison between the distortion of the woofer in the closed box and in the aperiodically loaded box. The measurement was done with the microphone in the near field. The loudspeaker was driven with 4 Vrms. The distortion seems similar for both cases but the interesting thing here is that the distortion of the aperiodically loaded woofer is a little higher at the region between 40 Hz and 150 Hz, exactly where the excursion is lower.

Figure 11 shows three distortion curves and may offer an explanation. Two of them show the distortion of the woofer as measured by attaching the accelerometer on its cone and the third one shows the distortion of the vent as measured with the microphone in the near field of its opening. The measurements show that the distortion of the woofer when it is aperiodically loaded is lower than the distortion of the closed box. The output of the aperiodic vent is not very linear and the distortion is much higher. Probably this has a strong contribution to the total distortion of the woofer as measured in Figure 10 due to their proximity, and it is an issue that needs further investigation.

A Passive Equalizer for Low Frequencies

As would be expected from a 17 cm driver, the loudspeaker is not particularly extended in the low-bass frequencies. For this reason, I was looking for a way to increase the extension of the loudspeaker at the low frequencies. My intention was to use a low-cost simple circuit that would not require a power supply.

The result was the passive equalization network shown in **Figure 12**. The circuit is placed between the output of the preamplifier and the input of the power amplifier that drives the loudspeaker.

The total value of resistor R1 in series with the output impedance of the preamplifier should be 3.3 kΩ. My preamplifier has an output impedance of 560 Ω and I used 2.7 kΩ for R1.

The total value of the resistance R3 in parallel with the input impedance of the power amplifier should be 15 kΩ. In my setup, the input impedance of the power amplifier was 34.7 kΩ, and I used 27 kΩ for R3.

The minimum input impedance of the passive equalizer is 5.2 kΩ at the frequencies above 300 Hz and higher at the lower frequencies. The circuit's frequency response shows a maximum loss of 8.6 dB at the high frequencies, but this can be easily compensated from the gain of the preamplifier. The loss at the low frequencies is lower and results in a boost of 3 dB at 80 Hz and 5.5 dB at 20 Hz.

I built the circuit on a perforated board and housed it in a small plastic box with of dimensions 100 mm × 50 mm × 25 mm, with permanently-connected flying leads for the output to minimize losses. **Photo 3** shows the internal construction of the device.

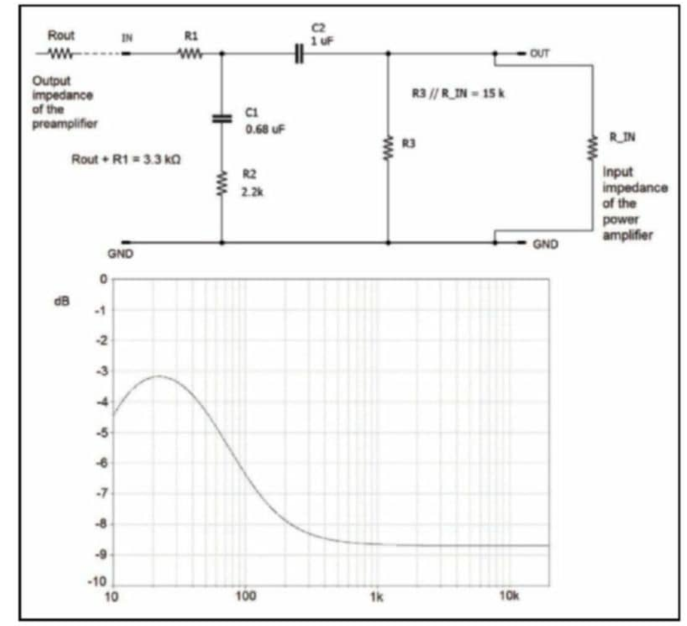

Figure 12: The passive equalizer for the APR-17 Loudspeaker

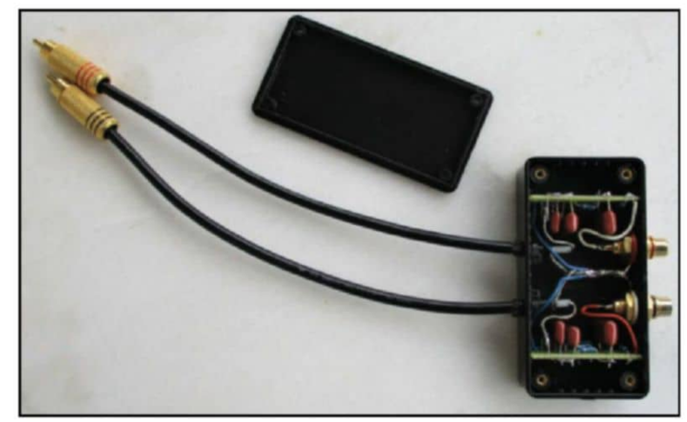

Photo 3: The internal construction of the passive equalizer

With the passive equalizer, the loudspeaker's tonal balance becomes more even. The negative is that the cone excursion is increased, requiring some caution when the loudspeaker's volume is increased and the music contains a lot of low frequencies. During my listening tests, at listening levels around 80 dBSPL to 85 dBSPL with peak levels up to 92 dBSPL to 95 dBSPL, I didn't noticed any problem, but as this depends on the kind of music and the sound level that is being reproduced, some caution is always required.

Conclusion

The APR-17 is the result of my investigation into the aperiodic loading technique. It is a good sounding loudspeaker, and with the addition of the passive equalizer becomes a well-balanced loudspeaker. The loudspeakers should be mounted off the floor, preferably on stands. Corner positions should be avoided. In my opinion, its bass response is very detailed, tight, and clear, and I think this is the strong point of the aperiodic loading.

The RSX

A Large, Modern DIY Speaker

Chris Biese shares his design for the RSX speakers, which won first place in the Dayton Audio category at the 2019 Midwest Audiofest hosted by Parts Express.

Photo 1: The RSX speakers were displayed during the Parts Express competition in 2019.

By
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(United States)

There they stood, at a relatively imposing 4′ high, almost 2′ deep, and over 1′ wide. My ever-so-patient wife walked in and saw them for the first time. Being polite and keeping her real feelings to herself, she asked "Don't you think they're kind of big?"

"Yeah!" I proclaimed. I wanted a big speaker, you know, like the ones that adorned the front of the Sunday paper ads for the big box stores, marketed as parts of a rack system, which are often shown next to an equally giant rear-projection TV. I flipped through those ads and drooled every Sunday. Sure, they probably broke almost every rule of proper speaker design, probably sounded terrible, but who cared—they were big and awesome!

So fast forward to my mid-30s. I can say I have matured in my perception of sound quality, but this does not mean I have stopped listening to aggressive music or that I haven't occasionally forgotten where three o'clock is on the volume control. I wanted a big awesome speaker, but one that fit two criteria. One: It had to have enough fidelity to have progressed with my ears since my teens. Two: Despite its size, it needed to pass the visual approval process to be allowed in the house. **Photo 1** shows the result of my labor—The RSX Speakers.

Driver Selection

To begin, I wanted a larger diameter driver as part of my design. I decided 10″ would be the minimum cut-off and I initially wanted to do this as a two-way design. This was going to require one heck of a tweeter and/or a waveguide. There are some very good waveguides out there that could work or be made to work for this application (e.g., the SEOS line). But, I just wanted something different. I wanted something that would aesthetically separate itself from an econowave or sound reinforcement speaker. I was left to consider either a coaxial design, where the cone could act as a waveguide, or a standard two-way, either of which would require a well-behaved 10″ woofer. I also had to balance having some reasonable level of efficiency and low-frequency response.

I knew I had a challenge ahead of me. I looked at drivers from all the sources and the manufacturers I could find. While searching for the right driver, I made up my mind that I was going to build a coaxial. This further strained my limited number of candidates as I needed a large enough voice coil to accommodate a tweeter that would be up to the task.

The driver I kept coming back to was the Dayton Audio RS270P-8A. This paper cone woofer had some low-level breakup, but overall a nicely controlled roll-off, and it could get into the high 20s for bass response—all while exceeding 100 dB within Xmax. It offered a specified 2″ voice coil, which wasn't massive but allowed me to likely find a suitable tweeter to fit in its confines. The Dayton also offered a pointed pole piece extension (phase plug), which I had seen other DIYers remove in smaller RS woofers to make coaxial designs. I had previously used the smaller RS225 woofer and found the line to offer a nice balance of build quality, price, appearance, and performance.

I also found a few options for a tweeter candidate and settled on a Dayton Audio ND28. The mounting flange was too wide for the woofer's coil gap, but the plastic flange could easily be removed with a band saw and fit nicely. The ND28 offered nice performance for a small neodymium tweeter at a low cost. As its name suggests, it is a large 28 mm dome. It has a low (for its design) resonant frequency of 1097 Hz and a very benign impedance peak at this frequency of only 12 Ω. This indicated to me that I may have a reasonable chance of making it work in this challenging situation.

DIY Coax

The RS270 woofer is a simple but nice design. It features a cast-metal basket, a paper cone, a rubber surround, a ferrite magnet, and a beautifully machined aluminum pole extension glued into what would function as a pole vent had the woofer employed a dust cap.

In the reading I have done on DIY forums, I learned from fellow DIYers that the pole extension can be easily removed with a punch and a hammer from the driver's rear, via the pole vent. I placed the woofer upside down on the supplied cardboard packaging, and using a 3″ long 1/2″ bolt and a solid whack with a hammer, the heavy aluminum piece popped right out. This allowed me to observe that this aluminum piece also appeared to function as part of the shorting path for the motor as the pole was absent a non-ferrous material on top of the pole.

As mentioned earlier, I took the ND28 tweeters to the band saw. This made quick work of removing the mounting flange and cutting the tweeter to size. I needed to form a mounting solution, so I picked up some hardware, including 1/4″ nuts, 1/2″ flat washers, 1/4″ cap screws long enough to reach through the magnet assembly, and some 1/2″ rubber washers. I found a brass screw and the majority of the washers were copper. I chose these materials as the pieces I removed were non-ferrous and I wanted to try not to disrupt the woofer parameters. My hope was the copper washers would also help re-establish a shorting path and be thermally beneficial.

I soldered and double heat shrinked insulated some leads to the tweeter tabs. Using a bolt cutter, I cut openings in some of the washers to allow the wires to be routed to the interior of the washers and

Photo 2: Construction of the tweeter assembly. a) Mounting flange is removed on the bandsaw. b) Copper washers and a nut are epoxied on. c) Washers are stacked until adequate spacing is obtained. Note the gap for the wires to be routed to the inside. d) Mounting screw and wires are routed out the rear.

Photo 3: This is a close-up image of DIY coaxial driver constructed from a Dayton RS270P and a ND28.

run out the rear, via another cut washer. I used JB Weld to glue on all the hardware. I used the rubber washer and a few smaller washers to establish a moderate torque on the screw without bottoming it out on the tweeter magnet (see **Photo 2**). The completed assembly is shown in **Photo 3**.

Enclosure

I did my enclosure simulations in Jeff Bagby's Response Modeler and also Woofer Box and Circuit Designer (see Resources). I have found his Excel-based programs accurate and easy to use. He offers them free to the DIY community. I knew I wanted to get the best combination of power handling and low-frequency response I could. Being a large speaker, I targeted an F3 (-3 dB point) in the 20 Hz to 30 Hz range. This was doable with the space I had available to work with using the RS270P.

Dayton Audio provides links to datasheets for its drivers. I have found Dayton Audio's data to be typically quite accurate, which is nice for DIYers who lack a tool to measure the data themselves. Just to be safe, I used the Dayton DATS module to re-measure my specific drivers. The parameters were close enough to the published specs that either data set yielded a similar result.

The simulation showed the woofer capable of 105 dB with an F3 of 29 Hz in a 64L vented cabinet tuned to 28 Hz. The F6 was 26 Hz and F10 showed 23 Hz (see **Figure 1**).

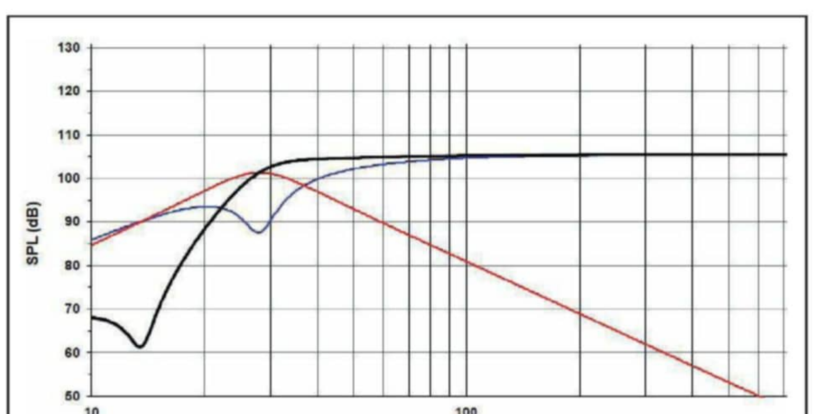


Figure 1: This is the simulated low-frequency response and SPL for a single RS270P with 75 W in the RSX enclosure.

The simulation was with a bass reflex design. Tuning low, 28 Hz in this example, will require a relatively long port. A large woofer at high output levels will need a large diameter port to prevent port turbulence, which is an audible chuffing noise. The larger the diameter, the longer the port, and I wasn't fond of using this long of a port. The enclosure volume needs to be increased to compensate for the volume of the port as well if the port is inside the speaker.

An alternative to a port is a passive radiator (see **Photo 4**). I like the looks of a passive radiator, it prevents chuffing, and also prevents midrange leakage that can occur with a port. The passive radiator is basically a speaker driver with no magnet or coil. It comes with a fixed set of parameters and can be tuned by adding mass to a bolt that is affixed where the motor usually is. Just like an active driver, the passive radiator also has mechanical limits and if exceeded, will damage the part. I was able to model the passive radiator in Woofer Box and Circuit Designer, just like the port.

The 10″ Dayton Designer Series Aluminum (DSA) passive radiators appeared to be within spec for this design. They were also reasonably priced and cosmetically attractive. The general rule of thumb for passives is to have twice the surface area of passives as you have active drivers. Two 10″ DSA passives appeared perfect in the simulation. This was ideal as I could place one on each side of the cabinet. I find this visually appealing and had always admired the quality of the bass in a pair of EPI A500 speakers I once owned. These employed a 10″ woofer with two 12″ opposing passive radiators. The bass just seemed to blend well with the room and not be overly fussy with placement. It also is ideal to have opposing radiators as they can generate a fair amount of force and cause a speaker to walk or rock. Opposing units cancel these forces.

Using two of these radiators I needed to add 140 g of mass to each radiator to accomplish my tuning frequency. This was heavy, but I have not had any issues. I used large steel washers and a digital scale to do the tuning. I chose large diameter units to keep the weight close to the spider to prevent unnecessary leverage pulling on the suspension. It is important to measure the weight when adding to each unit, as washers of the same diameter can vary within thickness and weight—even when from the same package.

Measurements and Initial Design Mockup

I was initially undecided about exactly how I wanted the final design to be executed. It is more typical to build the final enclosure first, load the drivers, measure the drivers in the enclosure, and then design the crossover. I was frankly unsure if my coaxial experiment was going to work out. I knew it would make noise, but would it be good sound and be worth my time? I had not previously experimented with coaxes. The woofer employs a pretty heavy cone, which has the potential to have a muddy midrange, and I wasn't sure if a wimpy little dome tweeter could do what a compression driver would usually be asked to do in a driver this size.

Because of these concerns, I cut an oriented strand board (OSB) baffle from some scrap to the width I anticipated to be that of the final design and screwed it to a sealed test enclosure. This test enclosure only offered flat bass to about 50 Hz, but this enabled me to gauge the baffle step correction needed for the test crossover.

Once I loaded the DIY coax into the test baffle, I used Omnimic to take a set of measurements. I do my measurements in-room and typically have no issue with this. I was unconcerned with the response below about 500 Hz, which is where the long wavelengths typically have significant room interaction, causing significant variations in the measurements. I knew I need not be concerned as I anticipated my crossover to be above 1000 Hz. Even with the variations in the bass frequencies, I am able to determine the baffle step correction without additional measurements. If one desires smooth flat measurements to lower frequencies, measurements can be made outdoors or free programs can be used to splice near- and far-field measurements.

An important measurement is obtaining the relative acoustic offset of the drivers. A point-source driver like this will be a lot closer to a zero offset than a tweeter on baffle, but it cannot be assumed to be zero. Jeff Bagby published a white paper on how to use his programs to determine the offset, using a combination of individual driver measurements, combined measurements, and constant distance/volume. This can be done in a variety of programs and ways, but I typically use Bagby's method. Despite the tweeter located at the base of the cone, it still measured 8 mm ahead of the woofer's acoustic center. This can easily be accounted for in the crossover design software.

Crossover Design

I used XSim to design the final crossover. This and most of the better DIY programs offer great features and are very accurate. I consistently find the final measured response to match the sim very closely. I plugged the Omnimic and DATS measurements into the programs and played around with various crossover points and slopes. The crossover design is a balance of many compromises. In this situation, I was bordering on too big of a woofer and too small of a tweeter. Too low of a crossover point on the woofer would not meet the small tweeter, and too high risked a muddy midrange. Too low or too shallow of a slope on the tweeter risked mechanical damage or buzzing. Too steep of a slope on the tweeter close to Fs may cause ringing from the high Q of the knee. Once these factors were balanced, I needed slopes to have acoustic phases that aligned as well. The balance on top of this is to ensure the electrical impedance and phase angle was not such that it was too taxing on the amplifier.

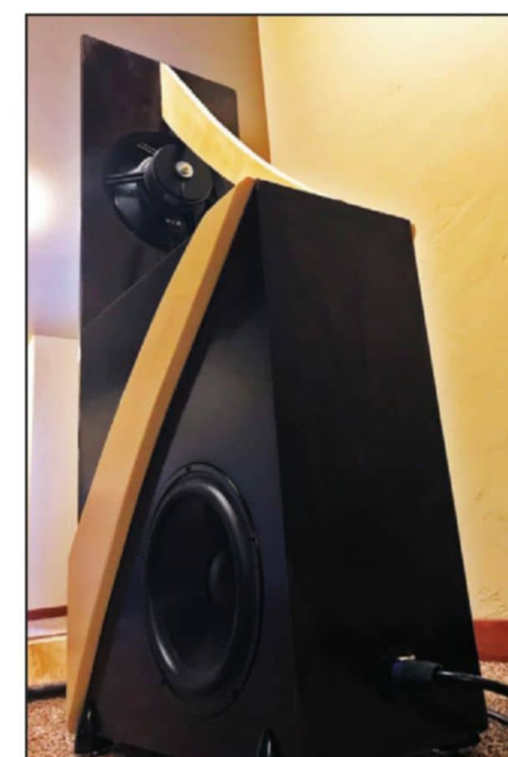


Photo 4: The back of one of the two 10″ aluminum passive radiators is shown, along with the Speakon connection for the crossover and the open-baffle mid.

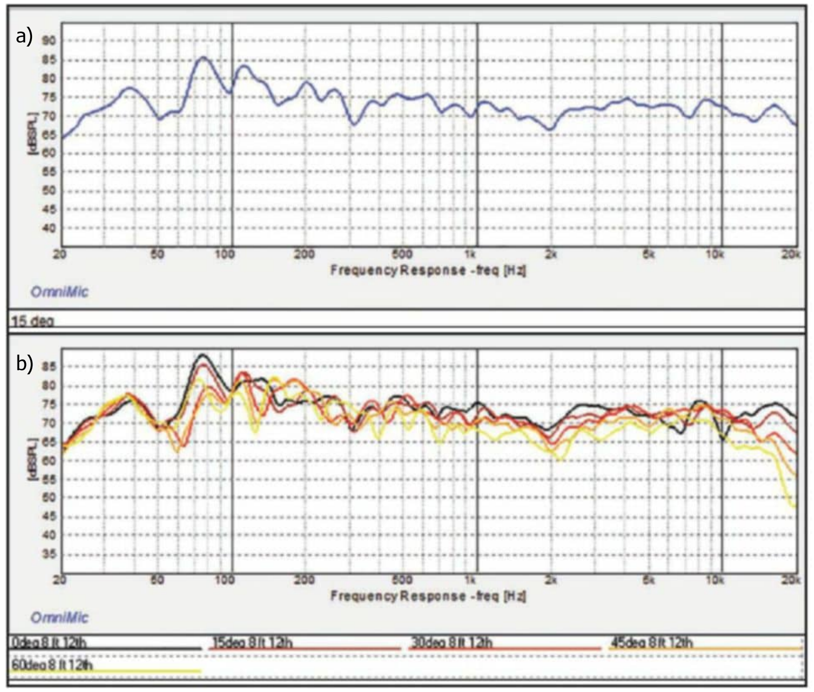


Figure 2: Here are the measured responses in a room approximately 16′ × 30′: a) 15° off-axis yields the best response. b) Various axis plots are shown. Note the measurements are far-field indoors with no splicing, so below 500 Hz has reflections present.

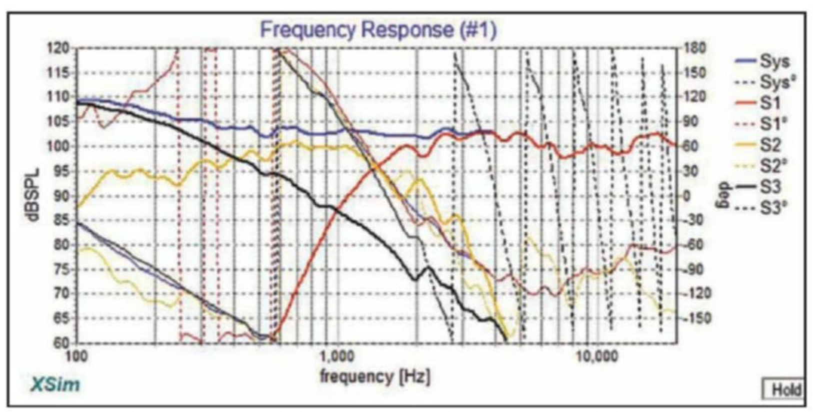


Figure 3: The crossover slopes and driver phase alignment are shown here.

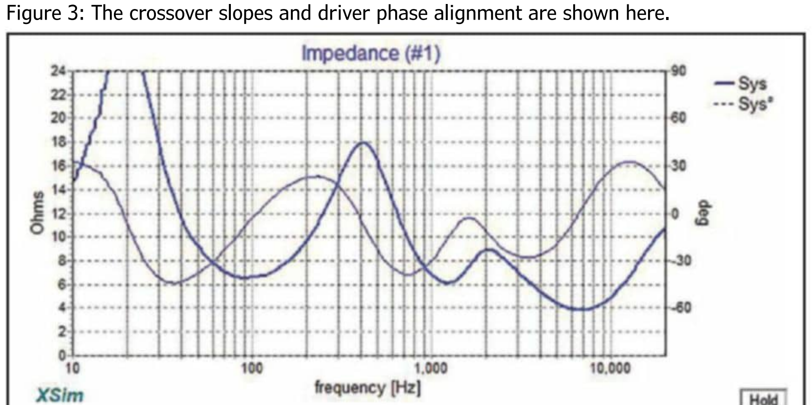


Figure 4: This graph shows the results for the final three-way open-baffle mid design.

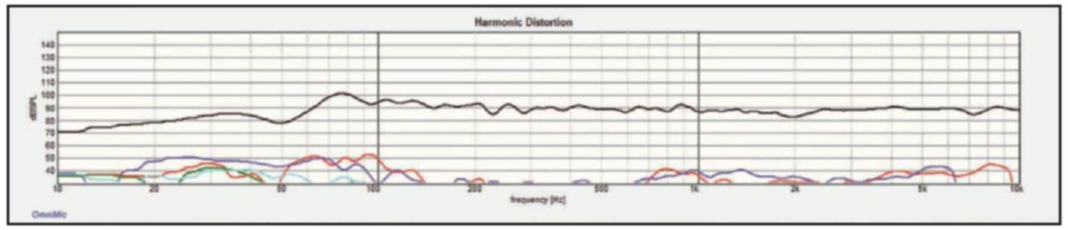
I had two things on my side for the tweeter situation, both of which related to the waveguide formed by the woofer cone. The first is the low-frequency boost offered by a waveguide. I gained 12 dB on the tweeter's lower end, which allowed me to cross more closely to the tweeters resonant frequency than normally possible. The second is the controlled directivity and tweeter location of this type of design allowed a higher crossover frequency than I could get away with in a standard two-way design. The one uphill battle I found though is one can expect odd dips at various angles caused by interference from the woofer's surround. Fortunately while dips look ugly on a chart, they typically are not as annoying as peaks. **Figure 2** shows the response measurements in-room.

Once a basic workable filter was drawn up for the two-way, I used alligator clips and spare parts to piece together a prototype crossover and found that the combination could work relatively well. I did not necessarily determine amplitude modulation to be much of an issue (distortion caused by the moving woofer cone surrounding the tweeter). The more I thought about things, I decided it would be nice to have additional output by placing a woofer near the floor (less bass is lost than when mounted high due to the boundary reinforcement of the floor), and an open baffle 10″ midbass would be kind of fun and unique. I ordered a second pair of RS270P-8A woofers. Once this was simulated, I made the addition of an open baffle to my more functional than beautiful OSB and plywood test box. I took additional measurements as necessary and began designing the new crossover.

I tried to target a Bessel acoustic slope on the tweeter for the softer knee. My hopes were this would reduce strain and potential ringing. Once the final three-way open-baffle mid design was completed, dealing with the acoustic offsets, I ended up with the best combination of phase alignment and response using approximately second-order Bessel at 450 Hz for the woofer's low-pass, a first-order Butterworth on the mid high-pass, a fourth-order Linkwitz Riley on the mid low-pass at 1300 Hz, and a fifth-order Bessel on the tweeter (see **Figure 3**). I intentionally used a very shallow slope on the mid high pass to make the most of the big driver. The resulting impedance and phase was not as flat as I had hoped, but appeared it would be no problem for an amp or receiver only designed for 8 Ω. Impedance is shown in **Figure 4**.

To further reduce strain on the tweeter, I employed a LCR notch filter on the tweeter at its resonant frequency. Items of note in the crossover design was the addition of a 0.1 mH inductor on the

Figure 5: This is the distortion at 90 dB/ 6′, in-room on a 10 dB vertical scale.



tweeter output to curb a rising tweeter response. I took some off-axis measurements and while the dips moved some, I found good consistency among almost any angle. I would call 15° the sweet spot. Distortion appeared reasonable measured at 90 dB at 6′ (see **Figure 5**). The other note is the typically odd 1 Ω resistor on the woofer. Despite concerns of heat, it actually does not dissipate that much power.

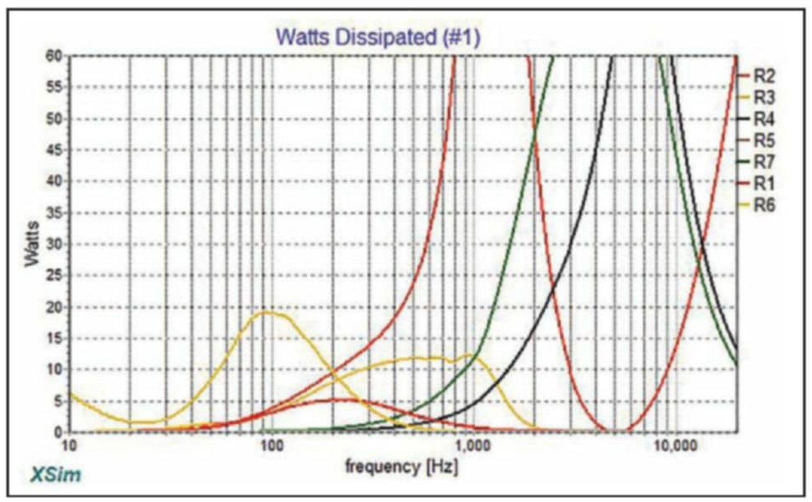


Figure 6: The graph shows the resistor power dissipation at 150 W. This is not a representation of music power so the numbers are not likely to be seen in normal playback, but this can provide a good guide as to which resistors may need to be upgraded from the standard 10 W, or which may be cut to smaller 5 W units.

Photo 5: The crossover was constructed using a separate board for each driver. Zister resistors were mounted to the front heatsink for the higher dissipation positions. I used XSim to calculate how much power each resistor will potentially need to dissipate as heat.

I found the bass blended better with this resistor in place. If a smaller gauge air core inductor was used as opposed to a steel lam, the resistor likely could be eliminated. I found it less expensive to go with the steel lam and resistor.

The crossover design came in at 20 components. I used a combination of various capacitor values that I had on hand to make some of the circuits, so there are more physical parts in my particular crossovers than would be necessary. I opted to do external crossovers just to do something different. There is plenty of room in the enclosure, if need be.

I wanted the crossovers to look like monoblock amps, so I used a heatsink as the face. This is a functional part as I affixed the hotter resistors to it as well. **Figure 6** shows the XSim heat dissipation sims used to get an idea on resistor size. I used Speakon connectors to connect the speakers to the crossover. **Photo 5** shows the layout I chose for the crossover's interior.

Enclosure Construction

The enclosure is primarily fabricated from hardwood plywood purchased at a home improvement store. I knew I would be veneering the baffle, so grain pattern was not a concern. I used 3/4″ Birch here. I happened to find a sheet of 3/4″ Maple that had a very nice grain pattern, so I selected this for the woofer box, knowing I would not need to veneer it. I selected solid Aspen for the decorative side pieces and baffle brace for its clean and clear grain.

I doubled up the plywood on the baffle, separating the two pieces with Green Glue acoustic compound, in hopes this would help deaden baffle resonances (see **Photo 6**). Construction adhesive was used at the perimeter to hold the panels together. Once dry, I used clamps and a drywall square as a guide to bevel the baffle edges with a circular saw. The bevels were asymmetrical to help with diffraction. The baffle was veneered with PSA (peel 'n stick) figured Maple and I used a router to cut driver holes. The drivers were flush mounted and the backsides were treated with a 45° chamfer to open up the area around the back of the woofers.

I used a table saw to cut the panels for the trapezoidal woofer box. The shape does help deal with standing waves, but frankly in this situation is purely aesthetic. I again used the router for the passive radiator holes, which do not require chamfering. A simple but effective cross brace was used inside to tie all panels together (see **Figure 7**).

Next, I glued the baffle to the enclosure with PL 8X construction adhesive. The interior of the enclosure was internally treated with a very small amount of carpet padding. I affixed the exterior Aspen trim pieces only at each end. The middle was treated with Green Glue.

I dyed the entire assembly (aside from the Aspen) with Transtint coffee brown dye diluted in denatured alcohol. I sprayed them with multiple coats of Bullseye Sealcoat and subsequently several coats of clear gloss lacquer, lightly wet sanding between coats. I allowed this to dry for a couple weeks and buffed it with three levels of compound and completed it with an automotive wax. **Photo 7** shows the finished product.

Tuning/Voicing

I spent a long time listening and making small tweaks. A crossover sim can be close, but the smallest changes can sometimes make a big difference. The biggest thing I kept coming back to was going back and forth on notching out the midrange cone breakup. The heavy paper cone does breakup some, but not necessarily in an annoying way like an aluminum cone can. I applied various designs of notches to address this. I found that too much notching of the breakup really took life out of the sound. I ended up with a simple capacitor around the 1.0 mH inductor, and this was really there for impedance purposes more than anything. **Figure 8** shows my final schematic for the RSX speaker.

Listening/Sound

I enjoy listening to these speakers. They are large and have big full-range sound. They are easy to drive with just a receiver and work very well doing double duty for a home theater. A surprising positive for me was the flexibility of room positioning. Having never built or had a dipole speaker, I was afraid they would be very particular about positioning in relation to room boundaries, but they are almost no different than any other large speaker.

One of the most enjoyable aspects to me is the quality of the bass. The Reference Series drivers from Dayton Audio have always had a consistent following and I can see why. These are nothing short of impressive. With response into the high 20s, I had some fun throwing in some electronic music, including some old bass tracks from my car audio days, and making these things work. The bass is solid and tight.

Photo 6: : I used Green Glue between 3/4″ baffle layers. The Green Glue provides no adhesion qualities, so construction adhesive is used to bond the panels near the perimeter and across the center.

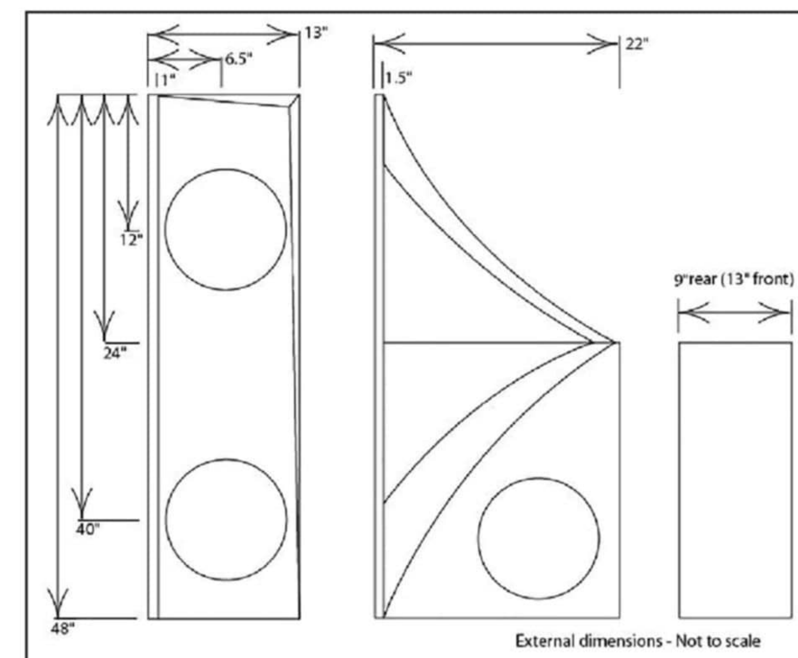
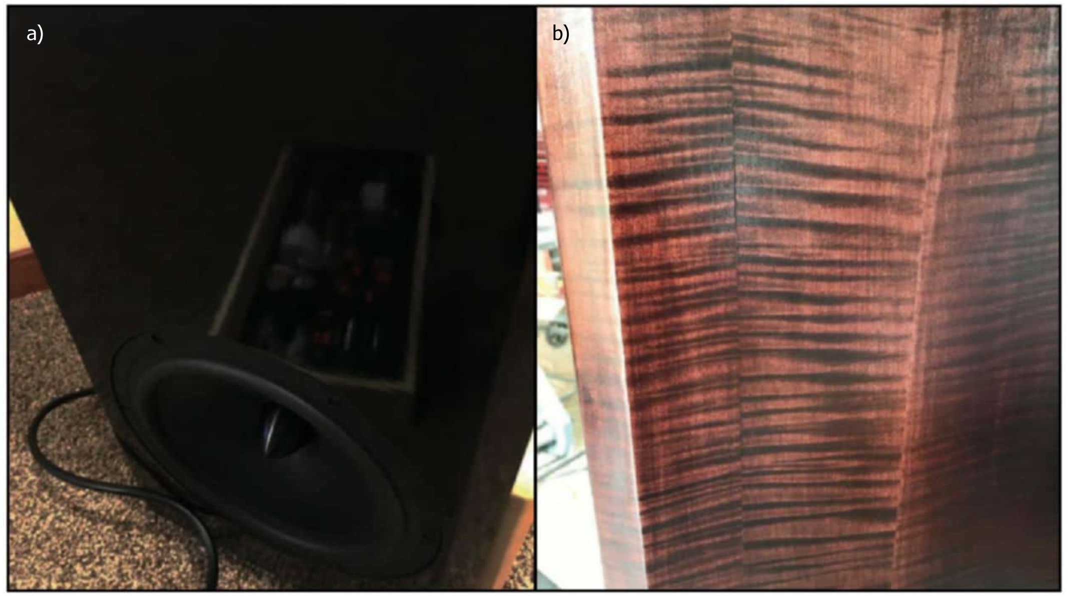


Figure 7: This diagram shows the external dimensions for the trapezodial woofer box.

About the Author

Chris Biese says he has no formal schooling in audio design or engineering. He started taking apart speakers and swapping drivers at a young age, which eventually led to doing new designs using basic box calculators and off-the-shelf crossovers. Once he was old enough to drive, Biese really enjoyed doing custom car audio installs. He returned to doing home audio projects again after he discovered the DIY forums and says he has learned so much from fellow DIYers. They helped guide him through learning how to design crossovers and patiently answered all his questions. Biese says that he learns something new with each project.

Photo 7: I created a glossy finish using several sprayed coats of de-waxed shellac and gloss clear acrylic (a). Curly Maple veneer was used for the baffle, dyed with Transtint Coffee Brown (b).



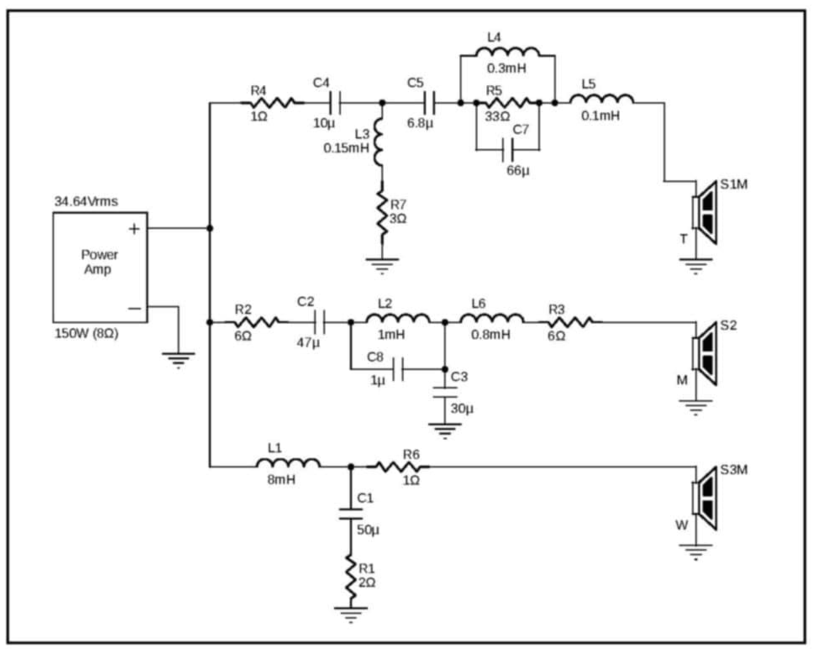


Figure 8: This is a schematic for the RSX speaker.

I debuted the RSX speakers at the 2019 Midwest Audiofest, hosted by Parts Express. I was fortunate to take first place with them in the Dayton Audio category. On a 1–10 scale, the three judges gave an average score of 9 in clarity, 9 in dynamic range, 7.6 in soundstage, and 7.6 in tonal balance. The 7-8 range is "above average" and 9-10 is "excellent." Some positive judge comments included: Great rock speakers, Good vocals and details, Good bass, Fun. Criticisms were: Upper bass a little forward, Overall balance heavy. I provide these numbers and comments as I am not a good judge of my work.

The competition at Midwest Audiofest allows three individuals to provide their thoughts and I appreciate the opportunity to have had their feedback provided to me. I have since made some revisions to a few crossover values based on feedback. I think the comment "Fun" is about the best description to sum them up. They are not the last word in neutrality or sonic perfection, but they are universal, loud, and when they are required to suit up for a serious listening session with a good recording, well, they can do that too.

Editor's Note: Jeff Bagby and his loudspeaker software are mentioned several times throughout this article. He was instrumental in the DIY speaker building community, and the Facebook group he and a friend started, the DIY Loudspeaker Project Pad, grew to have more than 23,000 members within just a few years. Sadly on March 24, 2020, Jeff Bagby, Chrysler engineer, math whiz, and devoted family man became the first person to die of the coronavirus in Howard County, IN. He was 60 years old.

Resources

"Jeff Bagby's Loudspeaker Design Software,"
http://audio.claub.net/software/jbabgy/jbagby.html

Parts Express, www.parts-express.com

XSim, Free Download Manager Library (FDB lib),
https://en.freedownloadmanager.org/Windows-PC/XSim-FREE.html

Sources

RS270 woofer, **ND28 tweeters**, and a **DATS module**
Dayton Audio | www.daytonaudio.com

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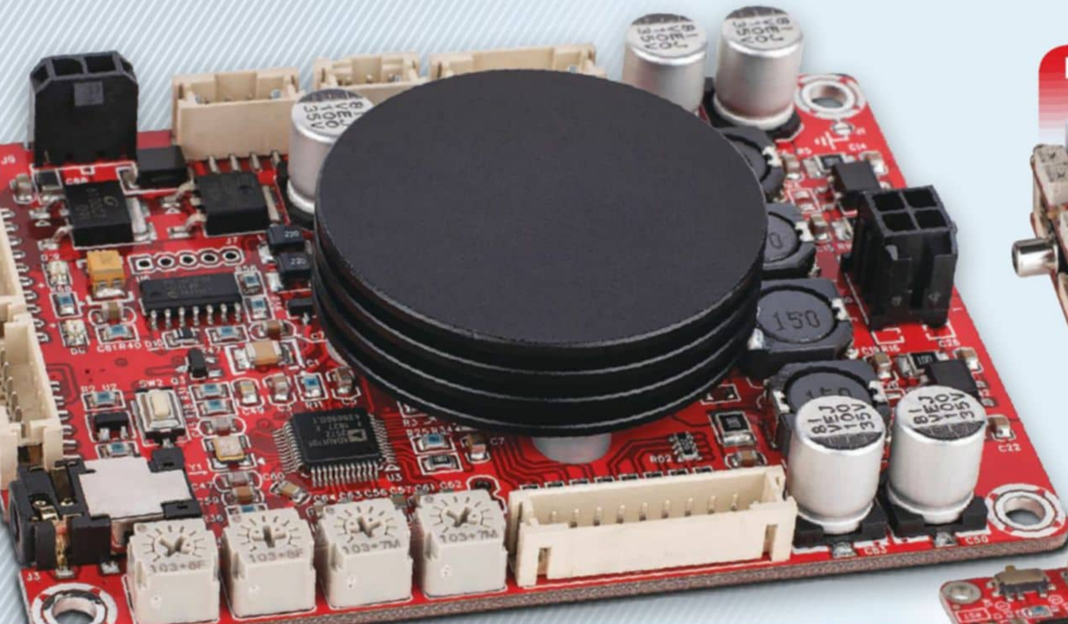
Digital Signal Processing **Made Easy!**



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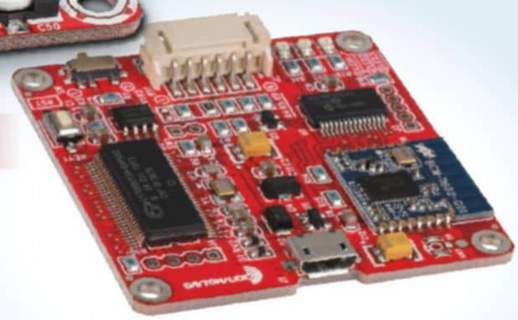
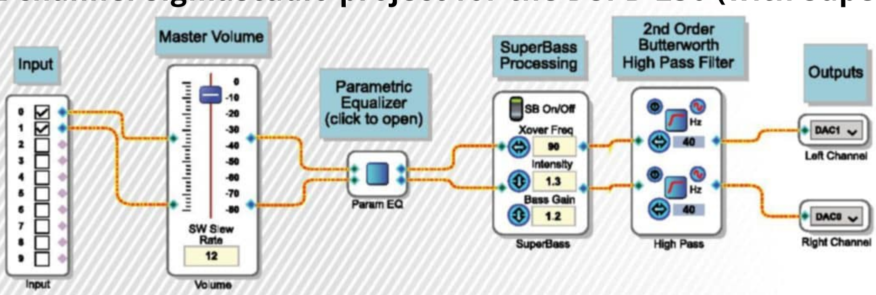


Dayton Audio DSPB-ICP1 In-Circuit Programmer for DSPB-250/100 and DSPB-K Boards

2 channel SigmaStudio project for the DSPB-250 (with Superbass processing).



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