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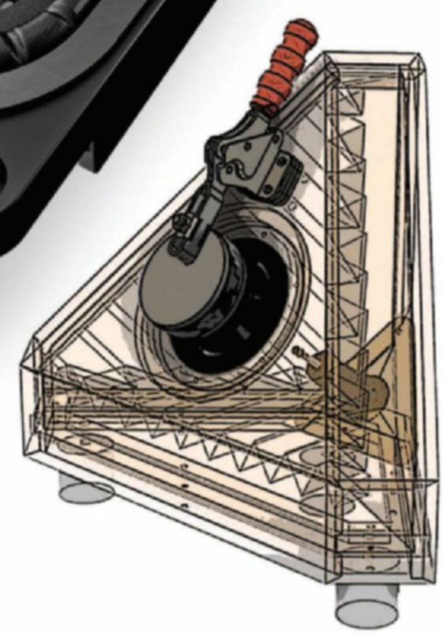

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Beyond Stereo, Speakers, and Reproduced Sound

There is still a lot to learn and improve with speakers, and how electroacoustic systems translate audio sources. Loudspeakers were originally designed to translate and amplify direct sound sources, starting with speech. Since the invention of sound recording, multitrack music recording, and the transition from mono to stereo, the simplistic goal of "sound amplification" and realistic reproduction has turned into a complex equation.

Should the ultimate goal of a loudspeaker be to faithfully reproduce sound as it was originated at the source? Or translate the recording itself? Or should it translate the result of a mix of sound sources (some of which have no acoustic origin and are purely synthesized) as it was produced in the studio? And what about the unique sound signatures of microphones or electronic circuits, which were intently used for its "coloration" effect in the sound? And what about acoustics and electroacoustics interactions?

All those questions were meticulously explored by our late and highly esteemed author Ron Tipton (1936–2018) in his fascinating and extensive article series titled "The Continuing Quest for Realistic Recorded Sound Reproduction." Ron expressed that exact complexity in the first sentence of the first of more than 40 articles on the topic, addressing how "conventional stereo" was recorded and the different microphone techniques. The more Ron explored the multiple efforts for "realistic sound reproduction" in his series, the more we understood how that quest was an enjoyable, but never-ending journey.

In this edition, *audioXpress* pays tribute to Roger Russell (1935-2021), who also devoted his life to the pursuit of audio reproduction and speaker design. With a fascination for stereo recordings and realistic sound reproduction, Roger designed speakers that aimed to translate the live performance experience. He was also an enthusiast of high fidelity and how audio equipment could translate recordings in home environments. He understood the basics of human hearing science and acoustic interactions and was not afraid to empirically accept and play with them. In fact, he enjoyed how modern recordings (and music genres) could generate emotions from those interactions.

"Stereo is still limited in part by recording methods, speakers, and room acoustics. Binaural recording and headphone playback eliminate the effects of the speaker and room but lacks the body perception of high level sound and lower frequencies as well as the freedom of turning the head to localize sounds," he wrote.

Today, we are able to enjoy those emotions he described with much more sophisticated immersive audio content using 7.1.4 Dolby Atmos speaker systems, and we are extremely close to realizing binaural spatial audio processing on headphones with head-tracking. The goal seems to go "beyond stereo," even if the intended goals for music are not yet clear.

In this Speaker Focus edition of *audioXpress*, we feature the work of Peter H. Lehmann, an author who has focused his efforts to enhance reproduction and "fix stereo." As he wrote, "In a stereo recording of music, instruments recorded to be heard as positioned directly in front of the listener are heard less distinctly than those instruments recorded predominantly to either the left or right channel of the recording." Not happy with the resulting phantom image that normally should result of correctly positioned left/right speakers in a listening environment—which strongly depends upon the original recording translating the "multi-directional perspective" that results from a truly "stereophonic" effect—Lehmann designed an "upmix" solution to derive a "real" center channel from the left and right channels of any conventional stereo recording.

Many companies are also currently researching the holy grail of "beyond stereo," now conveyed predominantly as "spatial audio" resulting from the reproduction of multichannel, object-based, or higher-order ambisonics (HOA) generated content.

It is my firm belief that we need to realize the spatial audio potential first and foremost using loudspeakers in order to convince consumers. And one could hope that more consensus about basic reproduction systems, such as stereo music, could have been achieved before departing on the "spatial" voyage.

J. Martins
Editor-in-Chief

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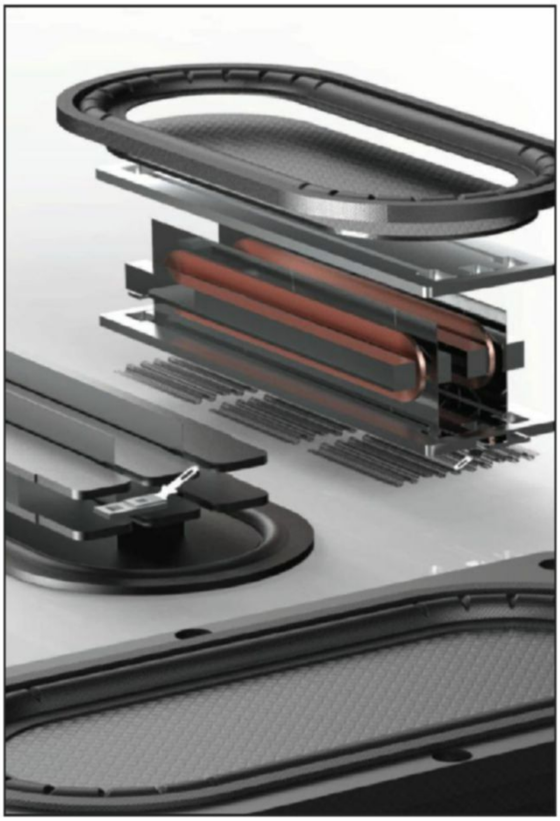

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Parasound Halo JC 5 Stereo Power Amplifier

Traditional Goodness

Free of extraneous ornamentation, the JC 5 makes a strong, simple statement.

Oliver A. Masciarotte was looking for an amplifier that he could respect without emptying his bank account. Richard Schram offered to send the Parasound JC 5, his company's largest stereo power amplifier for evaluation. Anticipating the fun of enjoying the pleasures of this John Curl design, Oliver accepted the challenge, and Stuart Yaniger tested the JC 5 to confirm the amplifier's merits with actual measurements.

By
Oliver A. Masciarotte and Stuart Yaniger

Richard Schram's Parasound Products, Inc. occupies a stealth address in a gritty industrial section of San Francisco, CA, south of Cesar Chavez. Like its gear, a modest facade belies the sophistication hidden from view. Parasound is a small, independent, owner-operated company that manufactures old skool, refined analog designs. Its hallmarks are very high slew rate, DC-coupled circuits with no output inductors, as well as high Class A power for a Class AB design at real-world pricing.

I was first made aware of Parasound long, long ago when I was part of the team at Sonic Solutions working on the then-new DVD-Audio format. Parasound's implementation had solved a fundamental flaw in the Forum spec*, so, from then on, I kept an eye on the company. Years later, I introduced myself to Schram, the founder and driving force, and have since enjoyed hanging with him both on and off the trade show floor. At the last AXPONA in 2019, Schram suggested we work on reviewing the JC 5, the second John Curl-designed amplifier for Parasound.

The JC 5 delivers an aggressively biased 12W into 8Ω in Class A mode, "spectacular," according to Schram, plus 400W Class AB into 8Ω and 600W per channel into 4Ω. In stereo mode, it is stable into temporary loads as low as 1.5Ω, and can deliver 1200W into 8Ω when bridged. Full power is available down to 5Hz. Twenty-four 60MHz bipolar transistors rated at 15A each provide peak current capacity of a substantial 90A. To supply that on demand is a power supply with a 1.7kVA shielded and encapsulated toroidal transformer with a premium H18 grade steel core (for "pressure" applications) and independent windings for each channel. It has independent secondary windings for its dual-mono power supply. The beast is supplied with a premium quality, 9', 12-gauge IEC power cord, with silver soldered connectors. In standby mode, it consumes an eco-friendly 1W.

JC 5
Parasound Products, Inc.
2250 McKinnon Ave.
San Francisco CA 94124
Cost: $5,995

High-quality binding posts and multiple inputs with gain make for a versatile solution.

Dual mono complementary differential JFET input stages, with extra beefy ±110VDC rails, feed dual push-pull MOSFET drive stages. As with its JC 1 grandparent, input B+/B- rails are 10V above the output rails to eliminate any chance of distortion caused by supply sagging when the amp is under load. Independent power supplies for left and right channel input stages employ 8,880µF of Nichicon Gold filter caps. Driver and output stages employ a massive 132,000µF of filtering from Rubycon.

Vampire 24-karat gold-plated RCAs offer unbalanced inputs, while discrete electronically balanced inputs via Neutrik locking XLR connectors are also supplied. RCA looped or parallel output jacks are present for daisy chaining, such as passive bi-amping or, in my case, a send out to a subwoofer transmitter. Excellent, easy torquing, 24-karat-plated five-way "propeller" binding posts are custom manufactured for Parasound by CHK Infinium. They accept up to 8-gauge bare wire.

As with most modern designs, the JC 5 employs a DC-coupled servo. It can dive well into subsonic territory, and soar into the ultrasonic. This guarantees that, in the 20Hz to 20kHz audio passband, its phase and amplitude behavior is exemplary. Schram told me, "I'm pretty happy with 100 kilohertz (bandwidth)... we don't find extremely high bandwidth absolutely necessary."

The user manual notes that, when the JC 5 is powered up, "...the dim blue glow around the On-Off button will become brighter and the two blue channel indicators will illuminate." The chunky, 30A protection relays click off and... I'm not quite ready to rock. I've found the JC 5 takes a bit of time to come up to temperature equilibrium and sound its best. Patience is a virtue.

Subjectively Seen

The Parasound JC 5 has several convenient features, including input attenuation for each channel with a 29dB range, from totally off to full gain. This allowed me to optimize gain staging as I typically feed any amp directly from my exaSound e22 Mark II. Also, the inclusion of those RCA pass-thrus made a comparison between the JC 5 and the Cambridge Audio Edge W (which I reviewed in the November 2019 issue of *audioXpress*) much easier since I could feed both amps simultaneously.

On a variety of material, from symphonic bombast and cool blue bebop to dance floor classics and hayseed Americana, the JC 5 offered a wider, more defined soundstage, an airy extended top end and a more solid bottom. Rosen on bow was more palpable, reverb more spacious and enveloping. Improved low amplitude detail meant each instrument or voice was better delineated.

If I had to provide a pithy comment on the JC 5, I'd say clarity and control are its chief attributes. If you were listening to highly manipulated "laminated" productions, as legendary engineer George Massenburg calls it, the differences are not as noticeable. But, on even slightly more purist productions, the differences become obvious. Though my better half, Kari, preferred the casework aesthetics of the Edge W, she found the sonics of the JC 5 more compelling. To her, the Edge W was "cloudier, somewhat muddier" when listening to Amanda Shires' HRA duet with Jason Isbell [*The Problem* 96kHz/24-bit Qobuz].

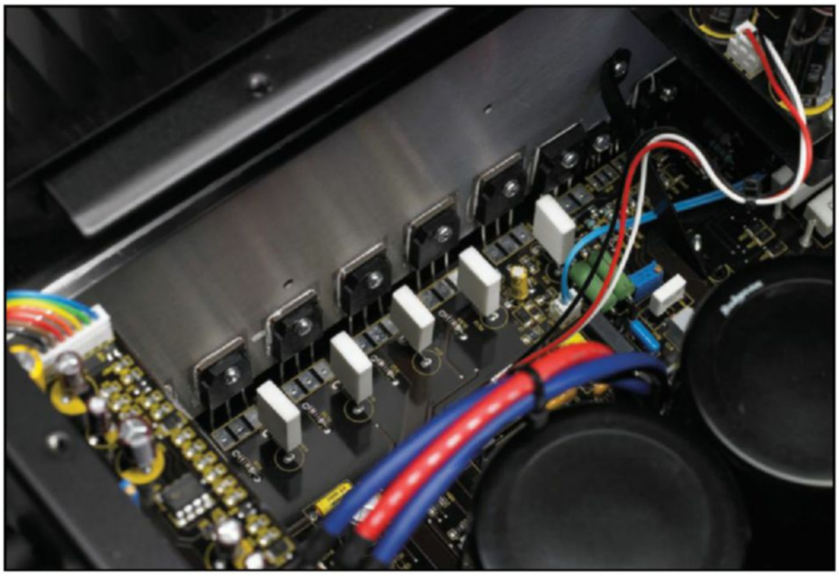
With particular recordings, however, that "fuzzier, more vintage" sound was for her an asset, as with John Denver's "Take Me Home, Country Road" [44.1 CD rip]. Not to my taste, but the shading lent an antique patina to the piece. For us both, the JC 5 offered tighter imaging with a more enveloping soundstage, crispness and more detail with a better balance between instruments while revisiting Steve Earle's "Transcendental Blues" [44.1 CD rip]. In short, the Parasound was, according to Miss Kari, "...a lot nicer," and this when compared to another excellent amp in its price range.

Different But the Same

At about the same power as the original JC 1, the JC 5 packs the equivalent of two into the footprint of one. Via e-mail, I asked Schram about how they

About the Author

O. A. Masciarotte has spent more than 40 years immersed in the tech space, working on manufacturing, marketing, and product development for many pro and CE audio manufacturers including dbx, a/d/s, Lexicon, Sonic Solutions, and Minnetonka Audio. His client roster is as diverse as Apple, Harper Collins, NASA Johnson, NPR and Universal. His writings include a book covering file–based music for the home and more than 100 articles for sundry trade publications. A member of the Audio Engineering Society (AES), Society of Motion Picture and Television Engineers (SMPTE), Project Management Institute (PMI), and Digital Cinema Society (DCS), he is currently co–founder and CMO of MAAT Inc., a digital audio software manufacturer.



Twelve beta-matched 15A, 60MHz bipolar output transistors provide brawny power.

differ, and what changes and/or compromises were made to accomplish that. He told me that "...the JC 5 topology is similar to the JC 1 but isn't identical. The most obvious 'compromise' is the lower Class A operating range—25W Class A in the JC 1 is the maximum possible without fan cooling. So we had to halve the Class A power in the JC 5. The filter caps are made by Rubycon and are smaller in consideration of cost and real estate. The Nichicon Gold Tune caps in the JC 1 are extremely expensive. Limited real estate limited the size of the power transformer and thus the rated power output. The -3dB low end is 5Hz, (for the) I/O (it's) 2Hz."

At a philosophical level, the A 21+ differs from the JC 5 mostly in the use of more expensive parts. Plus, more time was spent in refining every aspect. Schram confessed that the A 21+ is their "...current bargain of the century. One reason its price is so much lower is that we sell many more of these, as you'd expect, than the JC 5. Rather than inflate its price just because we could, we priced it to be even more attractive than the original A 21 so we could keep the large cohort of buyers for whom even $2995 is a stretch."

Since Parasound sells both traditional analog and Class D PWM designs, I was curious to know if Schram thought that Class D performance will someday equal that of Class A. He considers Class D capable of offering outstanding performance. "When we've asked listeners, including pros like Jack Vad, the Producer/Senior Engineer for the San Francisco Symphony (SFS), to audition the ZoneMaster 2350 he was absolutely smitten. He has numerous A 21s and his wife's system (she's a cellist in the SFS) uses JC 1s driving Magnepan 3.7is. There are a lot of prejudices and commercial reasons for disparaging Class D."

From a manufacturer's point of view, Class D is "overly democratic." Anyone can buy a really good quality Class D module, put it into a chassis and sell it for far less because it requires virtually no investment, overhead, or R&D. The highly specialized parts required for outstanding Class A or AB are increasingly rare. As an example, Parasound uses Toshiba 2SJ74 and 2SK170 for input stages. These parts have not been manufactured for at least 20 years.

"Only two other companies that I know of, other than Parasound," opined Schram, "...have any (and those are)... Pass and Ayre. Parts used in Class D are largely generic and manufactured in large quantity because they have many applications outside of audio. Specialty parts for Class AB amplifiers were always a niche product and as demand dropped back in the 1980s, companies like Toshiba were happy to discontinue them (and they destroyed the tooling for the dies). The only company to attempt to emulate the Toshibas is Linear Systems. According to Curl, their best efforts for the P channel FET don't match the venerable Toshibas."

Familiar Territory

When I first pitched the idea of a JC 5 review, my usually sanguine *audioXpress* editor J. Martins asked, "What do you expect to find? It's a John Curl MOSFET design that our readers know well. What will be the angle?" I had my views on that subject, but what to tell you, dear reader? When asked that same question, Schram was a bit sardonic. "Hmmm," he said. "How about 'It's a John Curl MOSFET design that your readers know well." Once the wink and nod had been delivered, he went on to say that, "... seriously, John is always looking at better ways to design amplifiers. The JC 5 is the best we can do to achieve virtually the same performance as a pair of JC 1s in the same real estate and less cost." To that I'll add, dear reader, that sometimes traditional is good. Especially in these trying times, exceptional build quality, guaranteed reliability, and accomplished execution are all virtues worth pursuing.

As I've mentioned before, high-end hi-fi isn't only a rich man's hobby but, as with cars, boats, and other luxuries, the more dollars you throw at it, the more likely you are to close in on your own image of "perfection." The 4U, 73lb Parasound Halo JC 5 stereo amplifier carries a suggested retail price of $5,995, and includes a fine fat power cord, rear handles plus front rack mount hardware. I was unable to compare the JC 5 to unobtanium amps that start at three times the price and spiral upward in cost from there, but I can confidently say that you would have to throw down for much more to better the subjective sound quality. Your entire system—

from sources to wiring to speakers—would also have to be exemplary to even match this amp's pedigree and performance. This behemoth is worthy of unqualified admiration; highly recommended.

** Author's Note: Until the DVD-Audio format passed into irrelevance, the standard setting DVD Forum insisted that the opaque user interface was perfectly acceptable to consumers. This was despite the fact that a television was needed to reliably play the supposedly audio-centric delivery format. Even when DVD-A champions at Warner Brothers insisted that a few mandated extra buttons be added to the spec, the Forum was unwavering. In response, Parasound released the only player I know of that incorporated a tiny video display on the front panel simply to make disc navigation possible without an external display. I asked Richard Schram about that product, and was told they had "...engaged a well-known Norwegian designer to create a DVD player with the inspired model number D 1. It had a TFT display that was pretty cool. The designer overlooked the difficulty of actually manufacturing (the product) and many dollars later we had to let it go." As an aside, I long ago sold my reference DVD-A player from my authoring days, and have since ripped all high-resolution DVD-A tracks to files on my server.*

At over 33kg, the JC 5 is built for any challenge.

Parasound Halo JC 5: The Measurements

By
Stuart Yaniger

I'm sure Oliver Masciarotte will comment on this further, but my first impression of this amp was solid, massive, and... well, large. The toughest part of this review was hauling this beast upstairs to my lab, then after unpacking it, hoisting it up onto my workbench. After living mostly with Class D amps for the past year, this felt like a blast from my past.

The back panel is nicely laid out, and the I/O connections are clearly marked and easy to use. I also appreciate the input gain controls, which makes life easier and safer for those of us who are connecting and disconnecting upstream equipment and don't want to accidentally fry speakers—a real concern at the power levels on tap here. It's also useful for gain adjustment in multi-amped systems. One small note: Make sure that if you drive this amp with both balanced and unbalanced sources, you remember to use the toggle selector switches on the back to choose which one you're using. Otherwise, you can have some very puzzling losses of gain... not that I'd ever be forgetful and do that, of course.

The topology of this amp will come as no surprise to anyone familiar with John Curl's designs. His basic philosophy is wide bandwidth and low distortion before applying feedback, and he tries to minimize feedback as well. He tends to use FETs liberally in his driver circuits, runs his output stage idle currents a bit rich, and uses complementary circuitry wherever possible.

For the Parasound Halo JC 5, the inputs are buffered by discrete constant current source (CCS) loaded junction-gate field-effect transistor (JFET) source followers, which are direct coupled to a fully complementary bridged JFET differential amplifier. The JFETs are cascoded with bipolar transistors to decrease Miller effect and linearize the input stage by holding the drain-to-source voltage nearly constant. The differential signal at the cascode collectors is converted to single-ended and amplified by a bipolar Vas stage, again fully complementary, with their collectors connected together via a bias spreader (constant voltage) circuit. Complementary MOSFET source followers then drive the output stage, which consists of six paralleled NPN-PNP pairs, for a total of 12 output devices per channel. Output offset is corrected via an op-amp configured as an inverting integrator to avoid the usual large electrolytic capacitor in the feedback loop and any input capacitors.

After my exertions getting the amp unpacked and sited, I first gave it a quick listen, substituting it for the nCore amps in the NAD M10 that drive the midrange-tweeter section of my speakers; once the levels were adjusted, the JC 5 sounded just as clean and transparent as the nCore. The JC 5's extra power made itself known on a few recordings at brisk sound pressure levels, where I managed to clip the NAD, but the JC 5 played flawlessly. Using a Purifi Eigentakt amplifier in place of the NAD amp, I was also unable to hear any significant difference. Satisfied that it is audibly as clean and transparent as a good amplifier should be, it was time to take it out of my system and grunt it to the lab benchtop.

On the Bench

As usual, my test setup includes an Audio Precision (AP) APx525 audio analyzer and various home-built dummy loads—I leveled up the dummy loads for this review to accommodate the JC 5's rated power. I did miss one thing, though, which I'll talk about shortly. For ultra-low distortion measurement, I also use

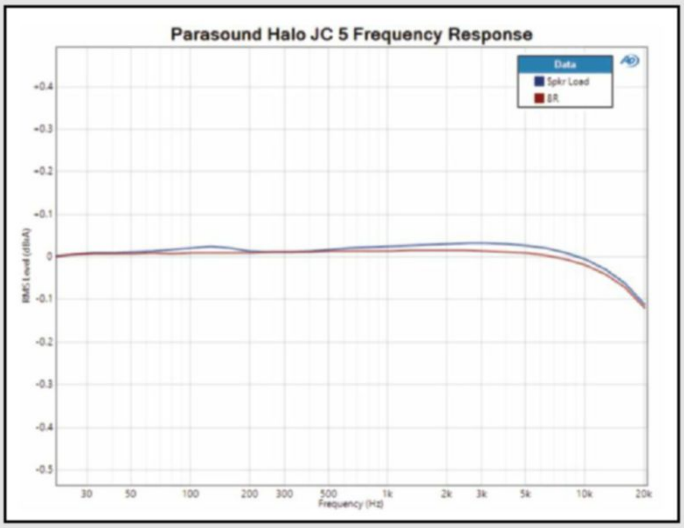


Figure 1: The frequency response of the Parasound Halo JC 5 is extremely insensitive to the load impedance.

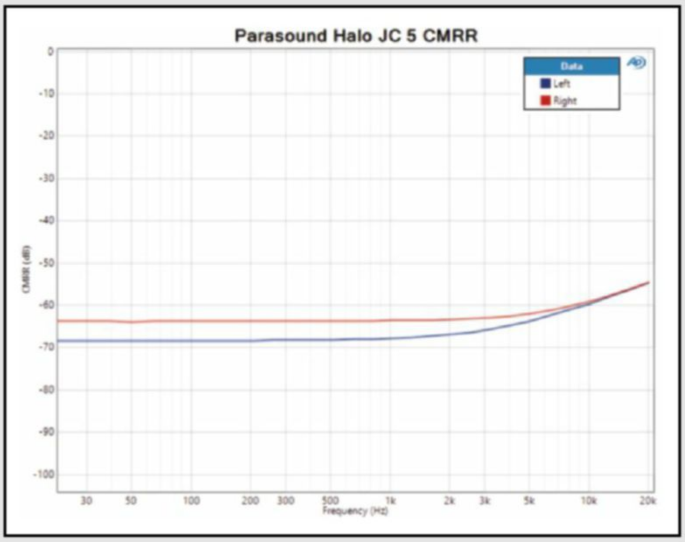


Figure 2: In balanced mode, the JC 5's common mode rejection (CMR) is very good and holds up well across the audio band.

a Vicnic 1kHz oscillator as a source, and a passive twin-T notch filter between the DUT output and the AP's input.

The JC 5 ran cooler than I expected, about 40°C on the heatsinks after a long idle or low power use, and climbing to about 53°C after a few hours of pushing it hard. The FTC preconditioning (one-third power sine wave into an 8R load for 1 hour) got the heatsink temperature up to almost 60°C, which is pretty hot but not enough to leave a mark. Because of the thermal mass of the metal used in construction, the front panel barely warmed.

I began with some baseline measurements, starting with voltage gain at zero attenuation, which was 28.8dB and 28.9dB from the left and right channel, respectively, in balanced mode. Unbalanced gain was barely different. Input impedance is specified at 66kΩ for balanced and 33kΩ for unbalanced, and measured almost exactly that.

Output impedance was below my reliable lower test limit (10mΩ) across the entire audio band. This implies that the frequency response will not be altered much with real-world speaker loads having large impedance variations with frequency. This is evident in the frequency response measurements (**Figure 1**). The change from an 8R resistive load to a loudspeaker only perturbed the response by 0.02dB. The response starts rolling off at the top of the audio band, down 0.1dB at 20kHz, which suggests sensible filtering.

One of the key advantages to balanced connections is rejection of common-mode noise. This, of course, is only the case if the source is truly balanced (i.e., has identical impedances on each leg) and the receiver has a high common mode rejection (CMR). Although CMR is generally given as a single number, typically at 60Hz or 120Hz, it will usually vary with frequency. Since common-mode noise can have many sources other than mains pickup, a well-designed balanced input stage will have high CMR across the entire audible range. **Figure 2** shows the JC 5's response (input-referred) to a common-mode input as a function of frequency; at mains frequencies, the CMR is 68dB and 64dB for the left and right channel, respectively, declining slightly to 55dB at 20kHz. This is very good noise rejection performance.

It's often interesting to look at distortion and noise residuals, especially at low power where artifacts like crossover distortion can become audible issues and where an amplifier spends most of its time when playing most music. **Figure 3a** shows the JC 5 output when playing a 2.83V sine wave at 1kHz into an 8R load (1W). Superimposed is the distortion residual magnified by 10,000 (80dB); the residual is almost entirely dominated by noise, with no visible signs of discontinuity at the zero crossing. Increasing the output voltage to 28.3V (100W) gives the waveform and residual shown in **Figure 3b**. It is dominated by the second harmonic at a very low level (0.03%). Clearly, the bias control of this amp is quite effective.

The spectrum of a 1kHz/1W/8R sine wave is shown in **Figure 4** for both balanced and unbalanced inputs. The predominant noise component is from the power supply at 120Hz, with a level of -88dB in both modes. To put that in perspective, if you have extremely sensitive speakers (e.g., horns) and you put your ear right up to them, you may be able

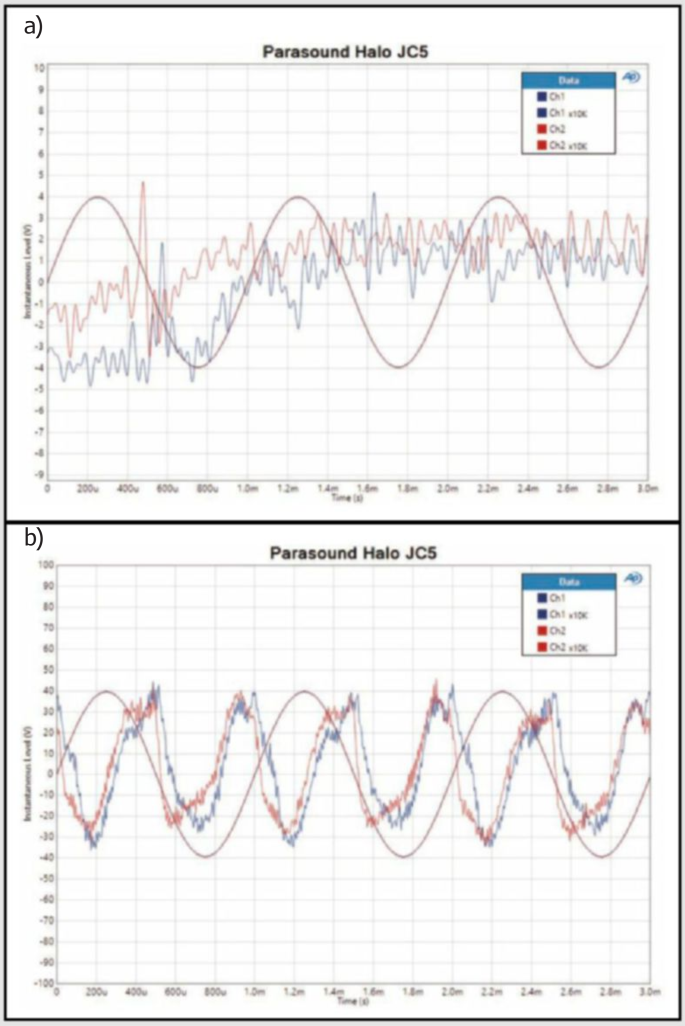


Figure 3: These graphs show the output waveform of the JC 5 playing a 1kHz sine wave at 1W (a) and 100W (b) into 8R, and the distortion residual magnified 10,000 times.

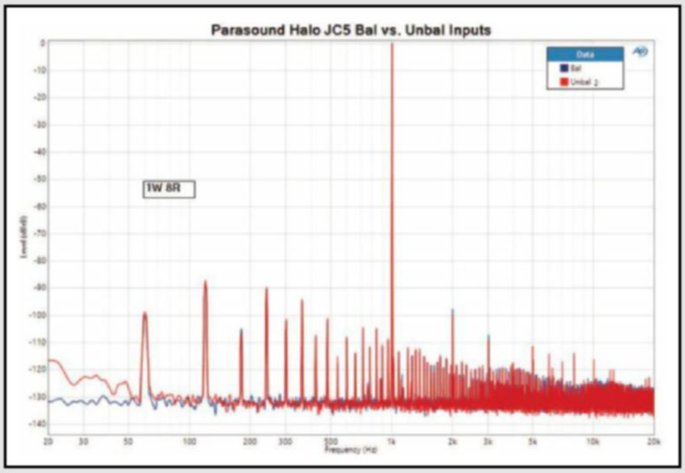


Figure 4: The spectrum of the JC 5 at 1kHz is dominated by very low levels of power supply noise rather than distortion.

to barely detect it—but if you have speakers that sensitive, you're unlikely to be pairing them with a 400W amplifier! Likewise, distortion is essentially identical in both modes and reasonably low, with the second harmonic being predominant at -100dB. Cranking it up to 100W gives the spectrum in **Figure 5**, with the second harmonic increasing to -82dB.

Next, I attempted to measure the THD vs. power level at 1kHz; using an 8R load, I obtained the curve shown in **Figure 6**. This was a bit surprising, with the amp showing its overload knee at 320W, with a 1% THD reached by 350W, a good deal of power but short of the specified 400W. With the load reduced to 4R, the amp comfortably met its 600W specification (**Figure 7**).

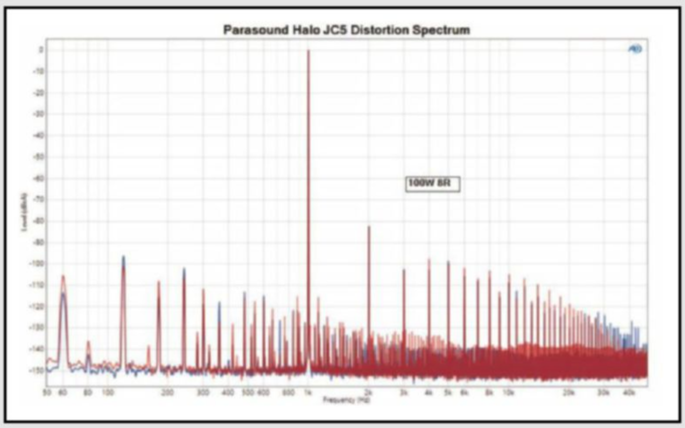


Figure 5: At 100W output, the distortion remains under 0.01%.

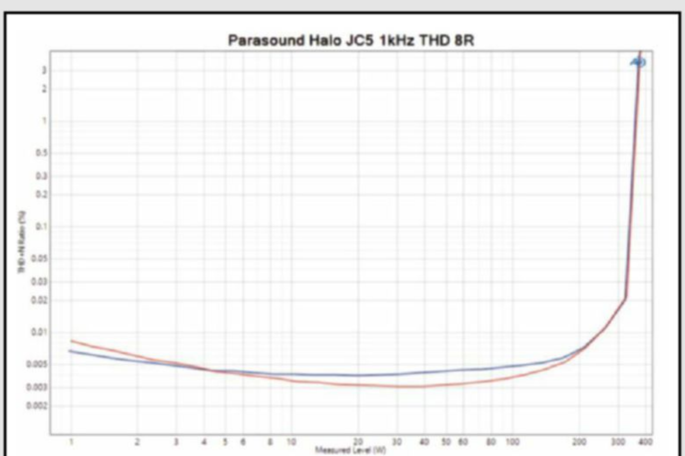
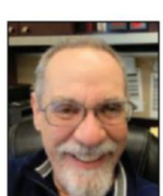


Figure 6: The power before clipping with an 8R load was limited by the ability of my house wiring to supply the JC 5, not by the amplifier!

About the Author



Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently runs a technology consulting agency in western New York. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.

On investigation, I discovered that despite being on a separate circuit for my lab/listening room, the JC 5 drew enough current to cause the line voltage to sag from a nominal 120V to 112V. This was the limiting factor for the 8R continuous measurement, and applying a correction indicated that the amp would make slightly over 400W with a stiffer electrical supply. Moral: If you're running equipment with this kind of power capability, you may want to put in a dedicated 25A power circuit!

My house wiring problem notwithstanding, I also tested the dynamic power using a series of tone-bursts, which is a short enough stimulus to avoid sag in the supply voltages both from the mains and from the amplifier power supply. With an 8R load, peak power was about 530W, which is 1.25dB of headroom above the rated continuous power.

I then looked at the dependence of THD+N on frequency at low (1W) and moderate (25W) power levels, with the results shown in **Figure 8**. At low power, the noise part predominated up to about 5kHz, at which point the distortion starts to rise, reflecting the inevitable drop in open-loop gain. At 25W, the

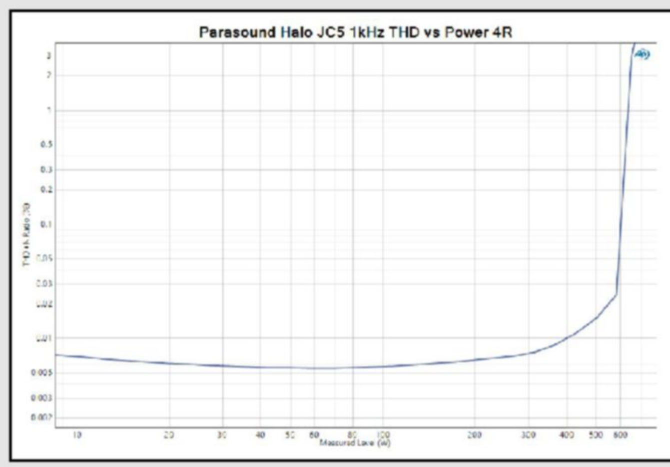


Figure 7: With a 4R load, the JC 5 reached over 600W output before clipping.

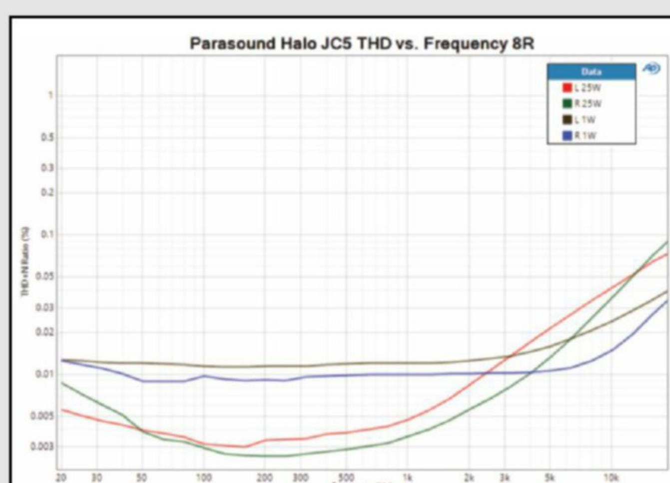


Figure 8: The distortion vs. frequency at two power levels remained low across the audio band.

noise floor drops below the distortion at mid-band, with the same rise in distortion in the top two octaves. Nonetheless, the distortion is reasonably low, and certainly well below any auditory threshold.

One interesting phenomenon I observed: as I swept through the audible spectrum at the higher power, the amp began to mechanically "sing" in the 2kHz to 5kHz range. I did not trace which part was emitting sound, but it was noticeable from a couple feet away at the test bench. With music playing, it would likely be unnoticeable, but it was a bit unexpected.

Moving up the ladder of signal complexity, the two tone (10/11kHz) intermodulation distortion spectrum driving an 8R load, with each tone set at 2.83VRMS (0dB on this scale), is shown in **Figure 9**. The first-order product (1kHz) is under -93dB from the reference level, with sidebands at -90dB. The largest distortion components were actually the second harmonics of the test tones at roughly -70dB (0.03%). Stepping on the gas, with each tone now at 100W (+20dB ref 2.83V), which is a rather punishing signal, the largest distortion component in the graph of **Figure 10** is still -65dB below 100W (0.06%). While this isn't a state-of-the-art number, it's still far below any audible threshold.

And finally, the multitone spectrum at 35V peak output into a 4R load is shown in **Figure 11**. As expected, at lower frequencies, the main offender is mains and power supply frequencies (multiples of 60Hz). At higher frequencies, things get a bit messier due to reduced feedback, but the level of distortion is still quite low and unlikely to be of sonic consequence.

To Summarize

The Parasound Halo JC 5 is a real powerhouse. The measurements fall short of state of the art, but are still orders of magnitude below audibility, and used in my system, I could not detect a sonic difference between the JC 5 and the two top quality Class D amps that I usually use. The I/O options and look and feel of the controls are superb. In balanced mode, the suppression of common mode noise is surprisingly good. This is a very well-engineered "classic" linear amplifier.

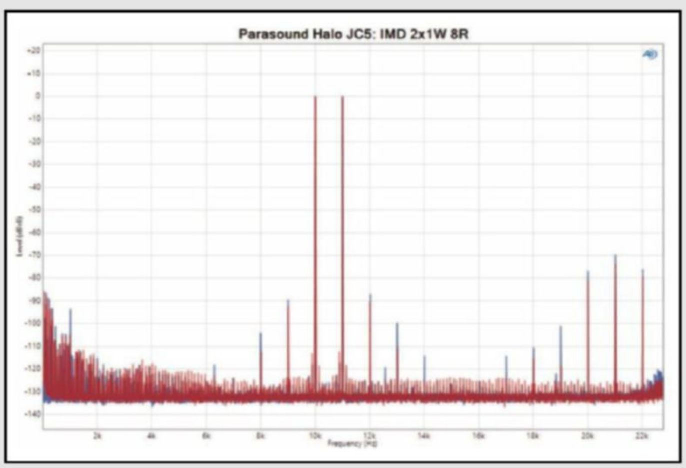


Figure 9: The high-frequency intermodulation distortion spectrum of the JC 5 shows very low levels of sidebands and difference products and is dominated by the second harmonics of the test tones.

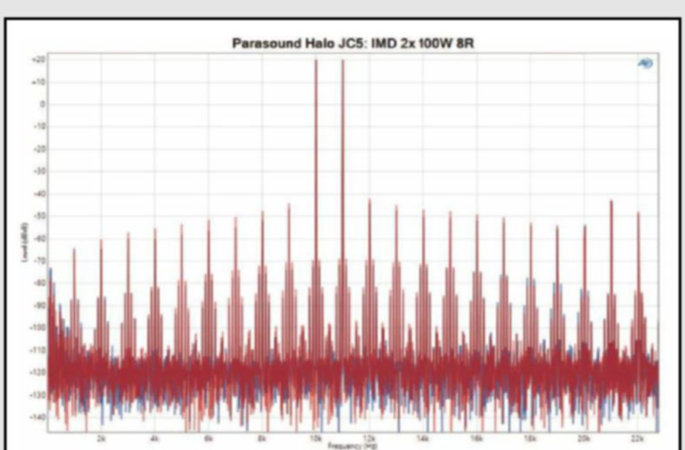


Figure 10: At higher power, the JC 5's high-frequency intermodulation (IM) distortion remains low.

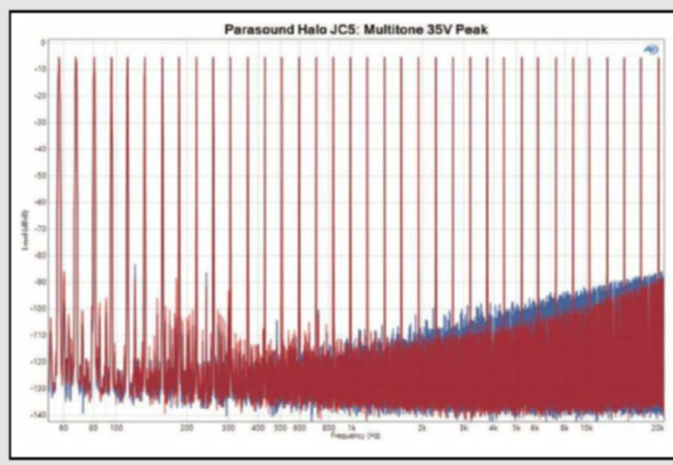


Figure 11: The multitone spectrum of the JC 5 at 35V peak into a 4R load shows low levels of noise and distortion.

Personal Seals of Approval

Noted designer John Curl mentioned at a seminar that he uses a JC 5. "The JC 5 is not a JC 1 exactly. There are differences, but I'm using a JC 5 pretty happily, and I've got $30,000 speakers and it works really well. Better than I expected really... given the trade-offs compared to the JC 1."

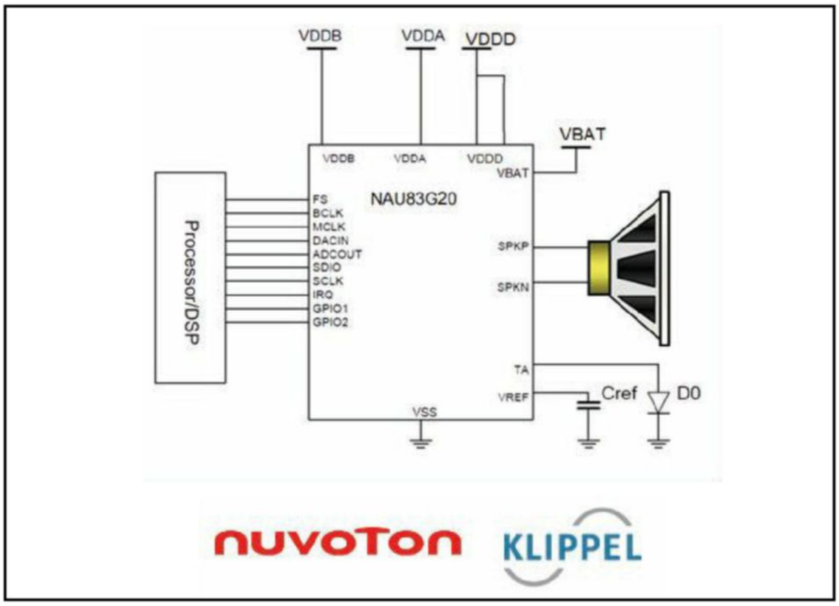
Vance Dickason, editor of *Voice Coil* magazine (a sister magazine of *audioXpress*), told me he's been a big fan of Richard Schram and his company Parasound for a number of years. "I have used HCA-800s as a measurement amp for my analyzers for at least the last 15 years. As far as my Halo P6 and A 21, they are absolutely outstanding. Also a big fan of John Curl. I listen to the equipment daily on some speakers I designed years ago for the Audax Vance Dickason Signature Series. The sound is very open, warm and musical."

New Ways to Formulate Speakers

This article looks at new companies, products, and inventions that are pushing speaker design to new levels. From innovative speaker designs and new transducer topologies to feed-forward protection circuits, this is an overview of innovation happening right in front of us.

By
Mike Klasco
(Menlo Scientific, Ltd.)

Perhaps both the easiest and also the most difficult topic to cover in audio is what's new with speakers. There are those who feel the existing products are just fine, Thiele-Small (T-S) parameters cover the engineering, a proper wooden speaker box should only have two wires going to the amplifier—and you have it all covered. But others feel that a speaker design should feature new diaphragm and enclosure materials, be designed with FEA simulation software, be laser scanned, and intimately interface with the latest DSP processing.



Klippel Controlled Sound (KCS) technology, integrated in a Nuvoton audio amplifier chip, creates a versatile solution to improve speaker performance and sound quality by compensating for nonlinear speaker responses.

The industry has seen a massive shift over the last decade with widespread adoption of mass-market audio systems, from smart speakers to soundbars. An honest assessment is that the sound quality of most home audio is not much more than a multi-channel table radio with a cheap remote woofer. Aside from pricing in a race to the bottom, visual rather than acoustic aesthetics now requires sleeker more compact industrial designs. In reviewing these innovations, I realized every single one results in a more shallow and compact solution.

The development that I see having the most impact on speaker designs this year is not even a transducer but a processing technique that will change how most of us design not just integrated speaker systems, but the drivers themselves. This is Dr. Wolfgang Klippel's Klippel Controlled Sound (KCS) in chip form from Nuvoton, which we mentioned previously here in *audioXpress* (see Resources). For my recent speaker projects, this is just what I needed. Essentially a dynamic pre-distortion circuit that is calibrated to the speaker and enclosure.

You might consider smart-amps the caterpillar and KCS as the butterfly, emerging as the complete form. A smart-amp, typically designed with the smartphone as its intended home, is predominately a feed-forward protection circuit specifically tuned for the limits of the speaker, both displacement and

Dinaburg Technology was issued its first US patent (10,812,912 B2) for an enhanced electro-dynamic loudspeaker design that consists of an active speaker constructed concentrically with a stabilizing ring radiator. This passive ring is compliantly held in place by surrounds on both the inner and outer periphery.

thermal. I have consulted for a couple of leading smart-amp chip vendors over the years and we consistently ended up rediscovering Dr. Klippel's patents.

The challenge was to increase the maximum acoustic output without creating new failure modes by inadvertently crossing the line with fatigue failure issues by dancing at the edge (which might annoy Apple, Samsung, etc.) or worse, driving to the edge on ring tones and speakerphone functions might inadvertently stumble into long-term speaker failure modes. It has been a few years now and NXP, Maxim, Texas Instruments, Cirrus Logic, Infineon, and Qualcomm have all succeeded with their smart-amps, all of which are a couple of watts—more than enough for smartphones considering battery drain and what the microspeaker can handle. At this point speakers can play as loud as possible without rattling buzzing, or failing—allowing everyone in the smart-amp business to sleep soundly.

But what else could speaker system designers want? I have always pressed the product managers at the smart-amp companies to consider a version of this dynamic protection processing in a form that could be used from soundbars to subwoofers to concert sound systems. For functionality, aside from saving the speakers from damage, how about we drop distortion, and for applications where there is full duplex with acoustic echo cancelers, along with painless barge-in, provide a significant margin before echoes.

Another intriguing aspect is what Dr. Klippel has defined as "green speaker design." Given the materials' budget, you can design an underhung voice coil with a huge magnetic structure and achieve high linearity—at a cost in weight. Or you can draw up a less extravagant design and use dynamic pre-distortion to keep your driver on its best linear behavior.

This is what Klippel's KCS promises, and as I found with my recent projects, it actually delivers. I cannot talk about the current product development that so impressed me (it won't be shipping until well after this is published) but let's just say I am a believer. KCS changes what a system designer can demand from smart speakers, Bluetooth speakers, soundbars, conferencing or huddle room voice lift, and even concert sound speakers.

Aside from this initial "excursion" into speaker processing, the rest of our focus remains on the electro-dynamic domain.

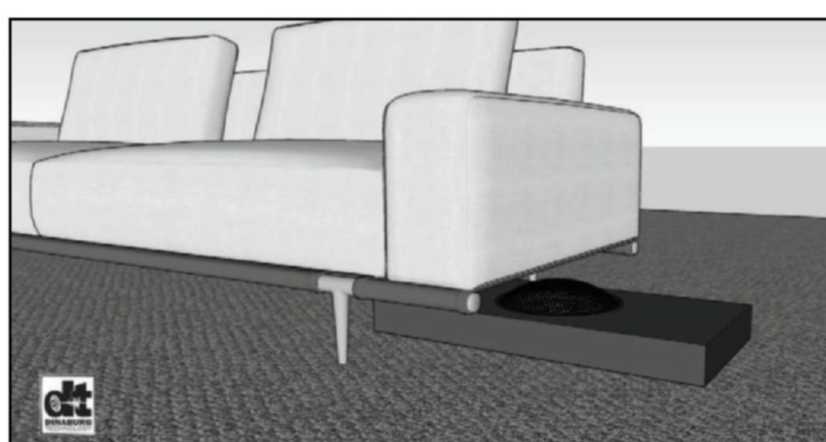
A New Coaxial Design

Dinaburg Technology has developed a speaker constructed concentrically with a peripheral passive ring radiator. This flat ring is compliantly held in place by surrounds on both the inner and outer periphery. There are a number of positive aspects with the ring configuration beyond the obvious benefits of a conventional vent-substitute design.

The design techniques enable lower distortion, extended frequency range, higher efficiency, and wider and more consistent beamwidth (dispersion). The invention has wide applications and can be used for near-field studio monitors, autosound, ceiling speakers, in-walls, and more uniquely to under-couch subwoofers.

Bass reflex, as provided by T-S simulations and supported by the acceptance of the audio engineering community, confers a combination of reduced cone excursion for a given acoustic output (in the range of the vent or passive radiator turning), higher sensitivity in this range, and/or extended low-end response.

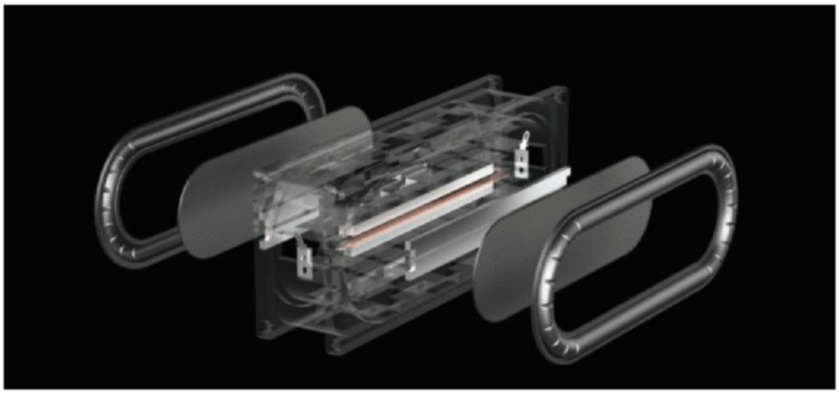
The passive radiator's diaphragm (compared to a simple vent) blocks midrange sound energy that is



A Dinaburg 8″ flat diaphragm active subwoofer in a 10″ frame with an integrated ring radiator in an enclosure height of about 3″ offers a practical under-couch subwoofer.

otherwise emitted from a port, and the passive ring configuration also provides for tighter constructive coupling to the active speaker (and to the room) compared to an open bass reflex port or a non-concentric passive radiator, which might have to be located on a different side of the enclosure.

One of the biggest challenges that low-profile speaker technologies have faced in the past, and still face today, is the reproduction of low frequencies. Resonado Labs solved this problem by engineering FCS technology to have a modular motor. By having multiple voice coil and magnet arrays under one diaphragm, the company can design powerful subwoofers that are able to maintain a thin structure yet have a large diaphragm necessary for low-frequency reproduction. FCS DualCore technology represents the FCS speaker driver structure comprised of dual voice coils bonded to one diaphragm.



FCS Bidirectional is the latest addition to the FCS suite. The patented design allows for two diaphragms to be attached and driven by a single voice coil, a concept that offers extremely wide acoustic coverage and is especially well suited for dipole-type loudspeaker systems.

About the Author

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.

From the measured data, it can be seen that the Dinaburg topology results in more output than what would be predicted by the more basic speaker box modeling simulations. Actually, these simulations assume the on-axis response in the range of where the passive ring provides a larger effective radiating area and tighter coupling of the bass to the room. This is in the bottom end response with the added benefit of avoiding beaming in the midrange that would have resulted from an active speaker of the same overall size as the outer diameter of the ring radiator. On the other end, the passive ring radiator maintains the speaker's pattern control down to a bit lower inflection point than the active driver has if just mounted on a baffle, (in all cases from seal, vented or with non-concentric passive).

The Dinaburg approach enables reduced depth with increased and extended bass response with higher sensitivity. The aspect ratio is that of a small active speaker but the radiating pumping power is that of a larger and deeper speaker. Additionally, a self-contained module is achieved with the combined frame/back chamber.

A couple of applications come to mind. Let's consider a standard 8″ ceiling speaker "high-hat" housing but with a 4″ active driver with a coax mounted tweeter and passive ring radiator. This design would offer wider coverage yet retain pattern control throughout the lower voice range, enabling wider spacing of the ceiling speakers, reducing both equipment costs as well as providing faster system installation. There are similar benefits for autosound in car doors or scaled-down drivers mounted into the front seat headrests facing the rear occupants.

One Dinaburg application that I am enthused about are shallow under-couch subs. Soundbars have dominated the audio-video market as they have the ideal form-factor to fit flat-screen televisions—except for the bulky sub. Design teams have wished for a practical under-couch subwoofer. A Dinaburg 8″ flat diaphragm active subwoofer in a 12″ frame with an integrated ring radiator is the same depth as a shallow 8″ woofer. With an enclosure height maximum of about 3″ and the driver/ring passive facing upward, the impact is immersive as the couch will provide both the bass and tactile bass.

Dinaburg Technology is currently focused on inviting collaboration with brands to offer its design innovations and technical support for a wide range of applications.

Thinner, Flatter, Better

Another driver technology that has been receiving attention is Resonado's Flat Core Speaker

(FCS) technology. FCS technology was introduced to provide the form-factor the advantages of racetrack drivers with the acoustic performance of conventional circular drivers. The motor structure of FCS is the key differentiator of the technology as the flat voice coil is able to run along the entire length of the diaphragm and apply uniform force. This enables a larger bandwidth of pistonic behavior for a high-aspect ratio, low-depth driver far superior to that of a conventional racetrack driver. FCS drivers can have the frequency range and efficiency of comparable cylindrical drivers (on a basis of diaphragm area).

Trulli Audio developed more compact iterations of its patented ThinDriver technology, measuring only 1.5″, and it is still able to deliver clear and powerful dynamics beyond its physical footprint.

Fibona Acoustics Enclosure Magnet Coaxial Transducer (EMCT) speakers comprise a transducer with wide range and coaxial driver in an enclosed box.

Earthquake Sound Corp. is the original inventor of the Shallow Woofer System (SWS) that produces a lot of bass from a very thin driver with an excursion of 2.5″.

The company's latest FCS DualCore was designed to address woofer and subwoofer driver types. To maintain pistonic force as the aspect ratio decreases and diaphragm area increases, FCS DualCore utilizes a second voice coil suspended in between a second pair of bag magnets, all underneath one flat diaphragm. This enables a larger cone and greater surface area to push more air for the reproduction of lower frequencies.

Resonado Labs licenses FCS technology to OEMs/ODMs and is currently licensing partners with Asian OEM/ODMs Zylux Acoustic and SoundLab.

Trulli ThinDriver Technology is another pioneering effort, and has been in development since 2008, under the Prescient brand. The evolved Trulli thin speakers can be made for anything from smartphones to laptops.

The TD-12 was characterized in a very good review by Vance Dickason, in the August 2015 edition of *Voice Coil*, but by the time the product was launched, the remains of the autosound aftermarket evaporated. The technology was relaunched with a new focus and the company is now Trulli Audio.

The Trulli TD38S 2″ (JAM5) micro-array speaker is a promising solution for portable applications, autosound near-field arrays, automotive personal sound zones with ANC, and quite a few other applications, due to its combination of high thermal power handling while maintaining a small footprint and shallow depth.

The topology is a flat diaphragm, by repositioning and expanding the voice coil to the juncture of the diaphragm periphery and the juncture of the surround. This 2″ speaker boasts a 1.3″ voice coil. This configuration also enables more spider corrugations so both huge gains in excursion and thermal power handling are practical. Volume velocity (pumping power = piston x excursion) through a square diaphragm for over 20% more piston area than a round speaker.

Also in the late stage of development is the TD200. This is a 10″ woofer with a 7″ diameter voice coil with less than 3″ depth (which includes clearance in front of the speaker for excursion). Ideal for in-wall subwoofers, underfloor autosound, and truck mounts (or even in the spare tire), and other applications.

Trulli is looking for development partners and licensees, and will soon be launching a consumer audio product. Len Foxman, Trulli Audio's President, will be happy to provide more information.

Moving Air Efficiently

Fibona Acoustics, based in Copenhagen, Denmark, boasts 30 years of experience, and is offering new transducer technology and licensing a package of patents and development support. Fibona uses a racetrack voice coil that is reminiscent to many smartphone microspeakers. Its Enclosure Magnet Coaxial Transducer (EMCT) technology is composed of a transducer with wide range and coaxial driver without the size constraints of traditional electrodynamic speakers.

The patented EMCT design features a shallow coaxial neodymium magnet design with a compact, integrated enclosure. The Fibonacci spiral, was the inspiration for the design. Also known as the Golden Spiral, this is a form used to calculate and convey the patterns of the Fibonacci numbers. Nature uses this number sequence for everything that we know—from the structure a flower to the galaxy and stars in the sky. Fibona has now recreated it on the EMCT coaxial diaphragm to provide incredible strength in a shallow profile.

Earthquake Sound's Shallow Woofer System (SWS) core technology was pioneered by noted speaker innovator Joseph Sahyoun and is protected by multiple patents. The SWS design inverts the cone over the magnetic structure and moves the flexible suspension components (cone and spider) outside of the magnet, allowing the motor to be elevated. This reduces the mounting depth of the speaker while enhancing excursion capability. Versions are in production for auto sound, in-wall subwoofers, and similar applications where very high output, extended bass, and shallow packages are required.

The patented ribbed cone greatly minimizes cone break-up distortion while maintaining a light moving mass, enabling more flexible installation enclosure options. The aluminum voice coil is welded to an aluminum dust cap thus turning it into a heatsink. This heat exchanger uses convection/conduction to transfer the heat out of the box avoiding power compression. Earthquake Sound also has extensive licensees with both OEM auto sound design wins and many aftermarket users of their patents.

Another speaker technology company about to make an entrance is Mayht. The company from the Netherlands has developed a highly compact, powerful speaker driver technology, with the promise of increased speaker bass performance, without compromises in quality, size, price point, or form factor. Targeted applications are flat TVs, portable speakers, soundbars, and portable PA and musical instrument systems.

Mayht is now licensing its fully balanced dual-membrane driver technology allowing for extremely low THD at high output—even in small enclosures.

The technology inside the Mayht driver include a new motor, suspension, and membrane architecture. After hundreds of simulations, thousands of prototypes, and a lot of testing, the portfolio currently consists of six patents. The Mayht dual-driver design uses multiple motors, symmetrically distributed across the membrane, allowing for a shallow design.

The Mayht driver is a single driver with two membranes moving in opposing directions. Mayht feels this is the most efficient way to increase air displacement capability and prevent mechanical resonance of an enclosure.

Resources

V. Dickason, "Klippel Controlled Sound (KCS) - Controlled Sound Technology for Nonlinear Compensation of Loudspeakers," *Voice Coil*, July 2019. https://audioxpress.com/article/klippel-controlled-sound-kcs-controlled-sound-technology-for-nonlinear-compensation-of-loudspeakers

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Fibona Acoustics | www.fibona-acoustics.com

Klippel GmbH | www.klippel.de

Mayht Technologies | www.mayht.com

Nuvoton Technology Corp. (NTC) | www.nuvoton.com

Resonado Labs | www.resonado.com

Trulli Audio | www.trulliaudio.com/early-access

The design provides all the benefits of the dual opposing principal, without increasing depth by having to mount two drivers back-to-back. Each driver enclosure uses what they call Negative Compliance, in order to avoid the effect of air pressure working against the movement of the membranes.

At maximum excursion, the membranes almost touch each other. Because of that, there isn't any space for a conventional secondary suspension and the Mayht driver contains a new distributed suspension technology.

With this approach, Mayht states its drivers are able to reach a 1:1.5 excursion-to-depth ratio because of its unique patented motor and membrane technology. One Mayht driver with two membranes fits in the size of a conventional driver, which only has one membrane. And with two membranes, the total Sd is doubled, further increasing the maximum air displacement.

Powersoft

New M-Force 301P02 Moving Magnet Linear Transducer

Ever since its release in 2013, Powersoft's M-Force patented moving magnet linear transducer has redefined what was possible in the realm of infra-bass, reaching well below the conventional frequency range. Now, the Italian company has announced a new and improved M-Force driver and free Reference Designs, enabling sound design professionals, speaker builders, and anyone interested in speaker engineering to benefit from the M-Force technology.

Despite its continued use and relevance at the highest echelons of live and installed sound, Powersoft has sought to push the M-Force technology's boundaries even further with the introduction of the new and improved M-Force 301P02 and its associated reference designs, which have been created to ensure that everyone can experiment with and explore the powerful subwoofer technology.

The patented M-Force moving magnet linear motor structure allows extraordinary levels of efficiency and reliability, making it a powerful choice for high efficiency, high SPL low-frequency sound reinforcement applications. And the implementation of the M-Force technology is now easier, thanks to the 301P02—a complete solution that enables all levels of speaker builders to incorporate the M-Force into any design with absolute confidence, both in self-powered and passive configurations.

The motor is factory matched to a purposely built diaphragm, through a newly designed coupling and a lightweight chassis that also improves thermal dissipation, while the addition of a front spider further stabilizes the design.

The new reference designs have been created by Powersoft to help users increase the SPL capabilities of their systems, whether they are speaker builders, sound system owners, or installation professionals. All of the designs are free and come along with a cut list and presets that can be managed using the company's ArmoníaPlus software.

www.powersoft.com

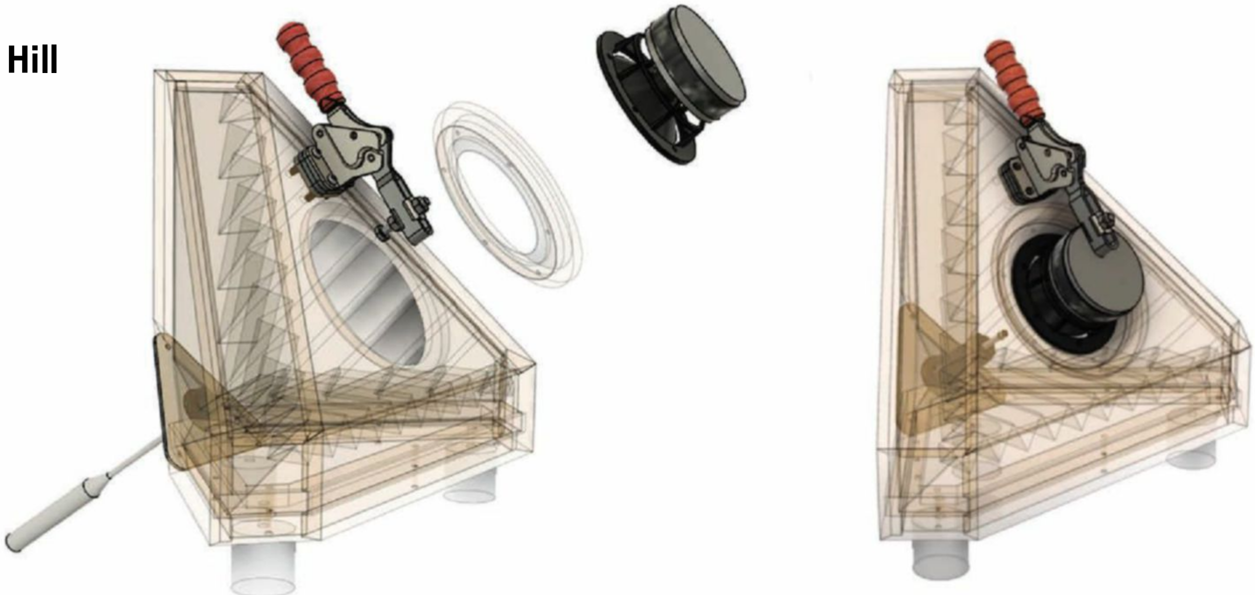
A subwoofer bandpass design featuring two M-Force 301P02 transducers, optimized for producing ultra-high SPL down to 18Hz.

SPEAKER FOCUS

The Tetrahedral Test Chamber Story (Part 1)

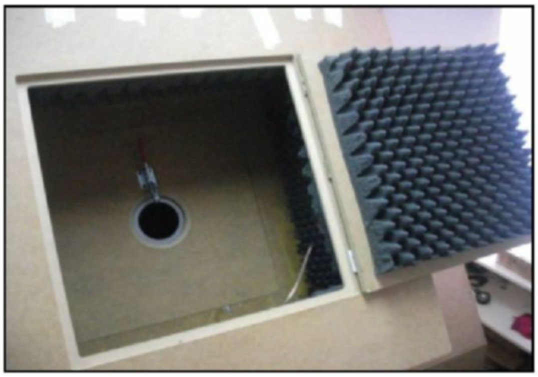
Recognizing the Need for Consistent Measurements

By
Geoff Hill



This article revisits the story of the development of the Tetrahedral Test Chamber (TTC) in an effort to solve the longstanding problem of making consistent loudspeaker measurements. This innovative loudspeaker test chamber, scalable to any size of loudspeaker driver, is now defined in international standards and in use throughout the world.

The roots of the Tetrahedral Test Chamber (TTC) goes back years, stemming from when I worked as a development engineer for Goodmans Loudspeakers, a respected British loudspeaker manufacturer, back in the mid-1980s. It was at Goodmans that I first designed and subsequently needed to measure loudspeaker drive units professionally.



An example of a test box that I designed back in 2010. This box already had some of the ideas that would be incorporated in the TTC, but it was still a huge box with a size of 2m × 2m x 2m.

My Early Days Using Anechoic Chambers

I had just joined Goodmans as a development engineer and left my position at Bowers & Wilkins (B&W), another highly respected British loudspeaker manufacturer, where I had been an electronics test engineer. At that time Goodmans' offices and factory moved to an industrial area in Hampshire, England. There, a new anechoic chamber was constructed, a massive steel monstrosity, 7m on a side with huge fiberglass wedges—some of which came from their old Downley Road chamber in Havant, Hampshire.

Access to the new chamber was via a stairway, which turned through 90 degrees before ascending roughly one-third up the structure of the chamber. So, it was a real pain to get anything in and out. Difficult and awkward does not even come close to describing it.

Entry to the Goodman's chamber was through a hermetically sealed 12" thick steel door. The two doors were constructed so that it was impossible to get out of the chamber if the doors were closed! As a result, it was company policy that no one person could ever work alone in the chamber, without another person always stationed outside.

The inner chamber door formed part of a 2PI baffle, and led onto a perforated steel plate catwalk, which caused severe acoustic reflections back to the measurement microphone. The catwalk allowed entry onto the steel mesh, which was strung side to side and front to back over the entire working area of the anechoic chamber. The mesh floor would bounce up and down as you walked around the chamber. This impacted the repeatability of the microphone and the loudspeaker measurement positions, rendering them unstable and inaccurate in X, Y, and Z planes!

However, these problems paled compared to the challenge of calibrating the chamber. Lots of care had been taken to ensure an adequate air flow rate so that someone could survive in the chamber should they get locked inside, I always thought it a pity that they did not consider that this same ventilation system made an ideal 15Hz organ pipe, resonating away at 85dB to 90dB SPL re 20uPa, of course it produced this level throughout the entire chamber!

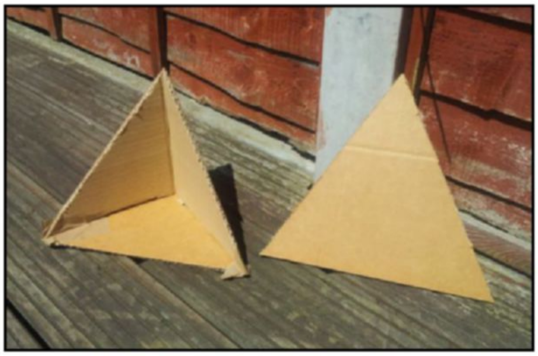
It was necessary to provide a reference loudspeaker to calibrate the chamber, so I selected one with as flat and as uniform a response as possible. We had this independently verified at great expense at the National Physical Laboratories in Teddington, so I was surprised to learn that the standard method for qualifying the chamber was just to feed pink noise into this speaker set up roughly equidistant of the center of the chamber—measuring the SPL in third octaves at 1m and then progressively increasing the distance along a single measurement line. Anechoic behavior being confirmed by measuring less than ±1.5dB deviation of one-third octave data, the low-frequency cut-off was similarly determined by the increasing error in this chamber once this exceeded 3dB.

The -3dB cut-off frequency was around 100Hz. This was at a cost of £250,000 and the chamber took up the space of a small house and was only able to measure down to 100Hz. Let us look at why a 100Hz low-frequency cut-off is a practical limitation that cannot be overcome using a conventional anechoic chamber!

We know the speed of sound in air ~343m/s, the wavelength is speed of sound/frequency (e.g., 343/1000 = 0.343m for 1kHz):

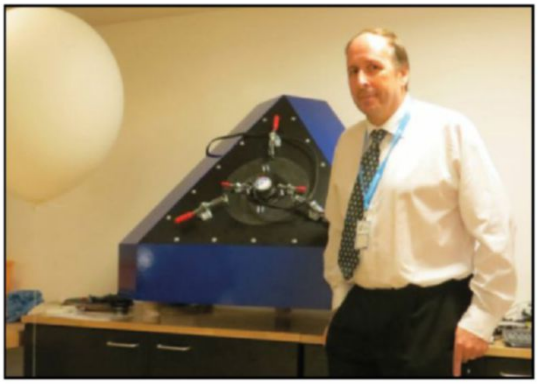
343/100 = 3.43m for 100Hz
343/10 = 34.3m for 10Hz

This JIS test box pictured without the IEC baffle in place, was created as an attempt to deliver repeatable loudspeaker measurements but it was still plagued with the problems associated with using an anechoic chamber. It can be seen here—one of the problems was a grated steel floor reflecting sound from the speaker back to the measurement microphone and corrupting the measurement results. The TTC solves this and many other problems.



Although I had visualized the concept of the TTC, I initially built it out of cardboard so that I could physically play with the idea.

A TTC test chamber capable of measuring drivers up to 12″ was packed into the back of a VW hatchback on its way to the Audio Engineering Society (AES) 51st Loudspeaker and Headphone Conference in Helsinki, Finland.



Geoff Hill is standing next to his game-changing TTC chamber at the AES 51st Loudspeaker and Headphone Conference in Helsinki, Finland. (Alan Slaughter photo)

In order to fully absorb a wavelength, the absorption material needs to be over a half wavelength long, so as the wedges in this and most chambers are less than 2m long this sets the cut-off frequency for most chambers at around 100Hz.

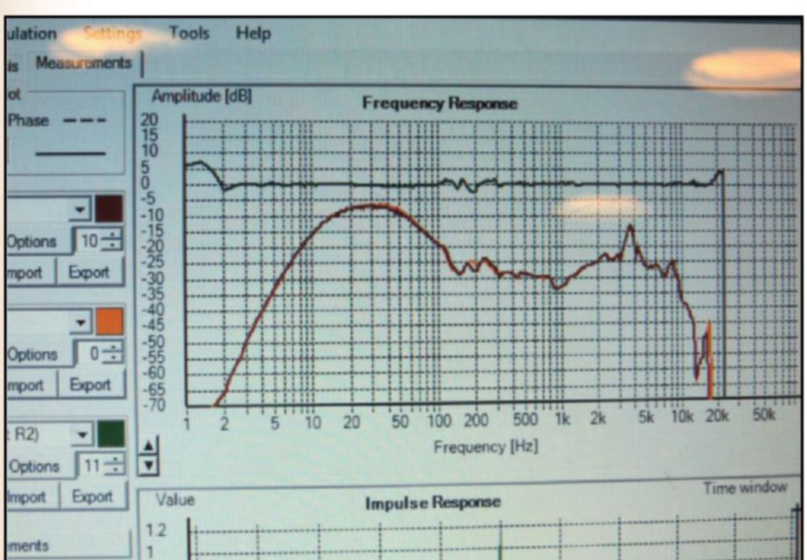
I was shocked when I realized that we had built a chamber the size of a small house so that we could measure 5″ (125mm) and 6.5″ (170mm) loudspeaker drive units that went down to 70Hz, which was below the low-frequency cut-off of the chamber.

End-of-Line Testing vs. Anechoic Chambers

It was also at Goodmans that I first became aware of just how inadequate the end-of-line and QC test stations were in comparison with the anechoic chamber, even with the problems that I already explained. It was shocking to see how little resemblance there was between the measurements made in the anechoic chamber and the end-of-line tests.

I well remember being called upon to "put limit lines" on a test for a loudspeaker driver being tested by an Ortofon P400 acoustic analyzer; this stored the curves in a plug-in EPROM and had to be manually increased or decreased around a "Golden Standard" drive unit that I had previously approved in the engineering laboratory's anechoic chamber.

The major problem was that not only was there a significant rise at low frequencies, but also the response was peaking up and down rapidly by ±20dB or more as well!



These are actual frequency response measurements made in Helsinki, including the difference curve of a series of measurements taken over three days, illustrating the consistency of the results using the TTC chamber.

The result was inevitable—the slightest problem or discrepancy, and "Engineering" (meaning me) was then called to the production line, to arbitrate as to whether there was a fault or not. And if so, what the cause of the problem was!

The bottom line was that the test chamber was seriously influencing the measurement—something that I would solve when designing my Tetrahedral Test Chamber (TTC).

All the line test boxes or chambers that I used or designed myself, until the TTCs, were universally inadequate and so it continued throughout my career whether at a hi-fi company, a mobile phone manufacturer, or a PA/music company.

When it all went wrong, I had to make sense of measurements sent to me by a driver supplier or a customer for approval. Often you could not even recognize the expected response of one type of loudspeaker over another. This was despite all the measurements being made by instruments able to measure better than ±0.1dB!

Giving Loudspeaker Measurements the Attention They Deserve

I must confess that at that time, early 1980s through 1990s and into the 2000s, I just accepted it as the way things were. That loudspeaker measurements were inherently unreliable and inconsistent; a view that I had held for many years! I was not alone as even today loudspeaker measurements are not given the attention that they deserve.

The rise in modern modeling and simulation has resulted in design engineers relying on the modeled response, while having little regard for the actual loudspeaker's response, then relying on DSP as a crutch to correct suspected problems rather than working from known accurate measurements. And to make those measurements, a chamber that did not require the space and the large investment of an anechoic chamber was needed.

In my experience, after designing many anechoic chambers and test chambers, the goal is to be able to make consistent measurements. Measurement errors are caused by the physical setup, mechanical issues, variability in connections, things not being in correct alignment, and the measurement methodologies in use. The anechoic chamber does not address any of these issues. So, we should forget about the debate around marginal improvements in cut-off frequency or the lowest noise floor of the chamber and focus on our goal of making consistent measurements.

In 2011, there was a lively discussion going on within the Audio Engineering Society (AES)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Standards Committee working group SC 04-03. The debate focused on the improvement of the measurements of loudspeakers in use in OEM and automotive applications. I was actively involved in these discussions. From these discussions I started to formulate and collect ideas for a book on the subject.

The model shown here is a TTC350, used for drivers up to 4″ (100mm).

About the Author

Geoff Hill, loudspeaker engineer, author, and 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 co-founded Impulse Acoustics in 1979, his first foray into the world of professional audio design and manufacturing, focusing on horn loudspeakers and amplifier design. In 1985 he moved to Bowers & Wilkins as Production Test Engineer and shortly after to Goodmans Loudspeakers where he was Development Engineer for 13 years, designing primarily hi-fi loudspeaker drivers and complete systems. After a short stint as Senior Acoustic Engineer for Wharfedale, Geoff was hired by Ericsson as Senior Acoustic Engineer. From there he moved to Motorola, to design, build, and implement a multi-user Acoustic Measurement Laboratory and large anechoic chamber for telecommunication products. In 2005, he returned to the world of speakers as Senior Acoustic Engineer for Monitor Audio Ltd. where for nearly six years he modernized the test and measurement facilities and equipment, designed and installed an anechoic chamber and led research efforts. In 2013, after working as technical director for other loudspeaker companies and continuing to offer technical support on the design and testing of loudspeakers, he founded his own consultancy business, Hill Acoustics. Geoff 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 the book *Loudspeaker Modelling and Design: A Practical Introduction* (Routledge 2018).

In 2012, I began writing my first book. As I wrote on the subject, it became increasingly obvious that designing a loudspeaker was only a part of the challenge, and a significant part was the verification of the design through measurements. Up until now, there was no acceptable way of doing this, short of the anechoic chamber. Around the same time, I was approached by Neat Acoustics here in the UK, to assist with its loudspeaker drive unit measurements. This is when I sat down and realized that I needed to resolve this problem once and for all.

The Deep Dive into Analyzing the Problems

I took all the knowledge and experience that I had gained over the course of many years and asked myself what were the best or should I say the most consistent loudspeaker measurements I had produced and why?

I quickly concluded that my most consistent measurements were made with the speaker firing toward or away from a corner of the anechoic chamber rather than using the standard baffle methodology.

Nearly all the standardized loudspeaker driver measurements required the use of a baffle, this compromises the measurement because most loudspeaker drive units are designed to be used in some form of enclosure or box. In addition, in none of the standards are the setup conditions of the baffle or the microphone defined apart from vague statements like:

"Measurements should be made in free-field conditions at such distance that the transducer/loudspeaker is operating in the far field," and that "care should be taken to ensure consistency in setup and repeatability."

These statements sound great in theory but truthfully, comments like this leave much unsaid and with so many different variables. For instance—How should the baffle be set up? Where should the baffle be in the chamber? Where should the test box be in the chamber? How should the microphone(s) be set up, at what angle and azimuth, at what distance(s)? With all these options, measurements become increasingly difficult to pin down precisely. This is understandable as in theory an anechoic environment should be totally free from reflected sound but in practice this is often not the case. The real problem is statements like "Care" and "Should" are hopelessly inadequate.

Anechoic measurements using a baffle are extremely unwieldy, requiring large and expensive structures. The anechoic chamber was originally designed as a general-purpose tool, which when combined with the use of baffles brings into question this whole methodology.

So how do we go about designing a test chamber that solves these issues? We must start by listing the criteria of a chamber that overcomes these problems:

- Footprint: The footprint should be a fraction of the size of an anechoic chamber. Acquisition, installation, and maintenance costs to be a fraction of those of an anechoic chamber.
- Repeatability: Must be capable of consistently stable measurements. This requires a defined geometry between the microphone and the loudspeaker under test.
- Accuracy: Must represent the tested loudspeaker's true response and capable of measuring to the low-frequency limit of the driver. Has effective absorption for high frequencies. Have minimal modal behavior. Capable of precise and simple calibration.
- Transferability: Able to obtain comparable results between chambers. Practical and straightforward implementation. Provide a path toward future standards.
- Simplicity: Be easy to set up and operate. To have a baffle and sub baffle allowing quick changeover of loudspeakers. To have the flexibility to operate with different sizes of microphones.

The Birth of the Core Concept

As I said before, many of the best measurements I have made have been in a corner. So, this formed the core of the physical design. I distrust complex solutions and prefer simple ones.

So, let us put the microphone in the corner and the speaker under test on a triangular baffle, firing sound toward the microphone. This satisfies the first criteria of keeping the structure relatively small and the second criteria with defined geometry.

Thus, the core concept of the Tetrahedral Test Chamber (TTC) was born. The name is derived from a regular Tetrahedron, which is made from four identical equilateral triangles.

My first thought was "can this concept really work?"

So, I reached out to a few esteemed colleagues and trusted advisors in the field of loudspeaker test and measurement and after considering my concept for a few minutes they ALL replied, "Well, I don't know why I did not think of this."

Wolfgang Klippel told me that I should publish this and get it out to the world. And then fortuitously, Neat Acoustics needed a measurement chamber and I had my chance to put the concept to the test. A few phone calls and meetings later I had an agreement for Neat to purchase the initial chamber, which has now been successfully in use for nearly 10 years.

Next I started the process of defining the scope of the project: First, Neat needed to measure five or six main chassis types from 5" through to 12" plus various size tweeters. So, this dictated the size of the chamber and I began sketching 2D CAD drawings.

Next Month

In Part 2 of this article, we describe the initial launch through the refinement phase, and peer review of the concept, proving the chamber's measurement repeatability, and its inclusion into the international standards. Then onto using the chambers in practice throughout the world.

Resources

AES73id - AES information document for acoustics — Loudspeaker driver comparison chambers

G. Hill, *Loudspeaker Modelling and Design a Practical Introduction*, Routledge/Focal Press, 2018. www.routledge.com/Loudspeaker-Modelling-and-Design-A-Practical-Introduction/Hill/p/book/9780815361336

G. Hill, "A Guide for Building Your Own Test Chamber," *audioXpress*, September 2020. https://audioxpress.com/article/practical-test-measurement-loudspeaker-measurements-at-home-a-guide-for-building-your-own-test-chamber

IEC60268-21:2018—Sound system equipment - Part 21: Acoustical (output-based) measurements

IEC60268-22:2020 Sound system equipment - Part 22: Electrical and mechanical measurements on transducers


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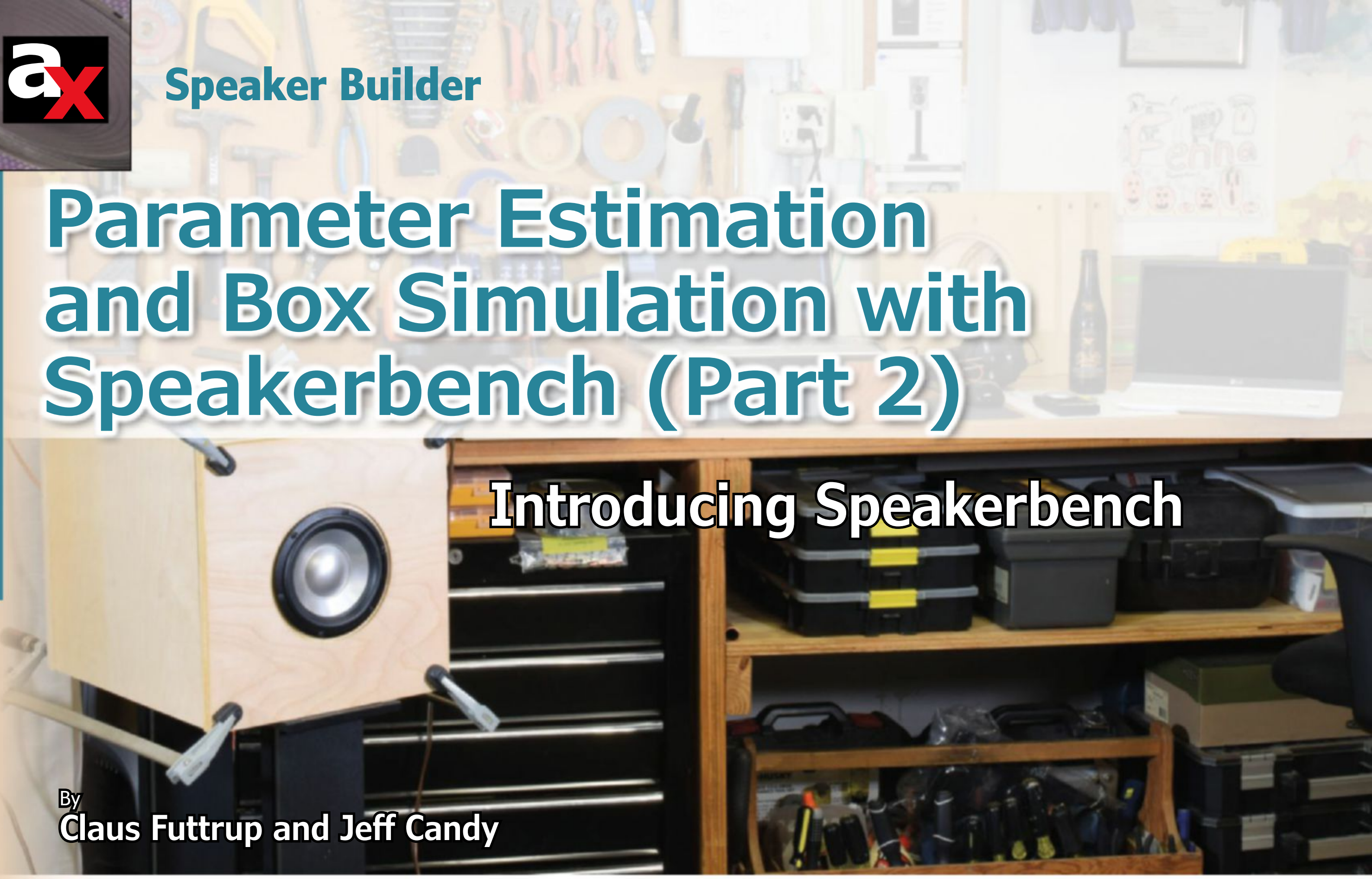

Parameter Estimation and Box Simulation with Speakerbench (Part 2)

Introducing Speakerbench

By
Claus Futtrup and Jeff Candy

In Part 1 of this two-part article, the authors discussed the difficulties in estimating cabinet losses using the classical Thiele-Small framework, and a solution using an advanced transducer model. In this second article, the authors present the online speaker design tool Speakerbench as an accurate and practical approach to this problem.

Although transducer manufacturers almost certainly have Klippel analyzers in their laboratories, they rarely publish viscoelastic data. Measurement files from the Klippel analyzer are not readily available for download, and moreover, we do not know of any available software that can process this data and use it for box simulation. One of the authors (CF) initially thought that it would take a few years for some in the industry to adopt advanced parameters once the advantage was understood. Alas, 10 years after the presentation [1] at an Audio Engineering Society (AES) Convention in London, England (May 2011)—and although the work has caught the attention of professionals in the industry

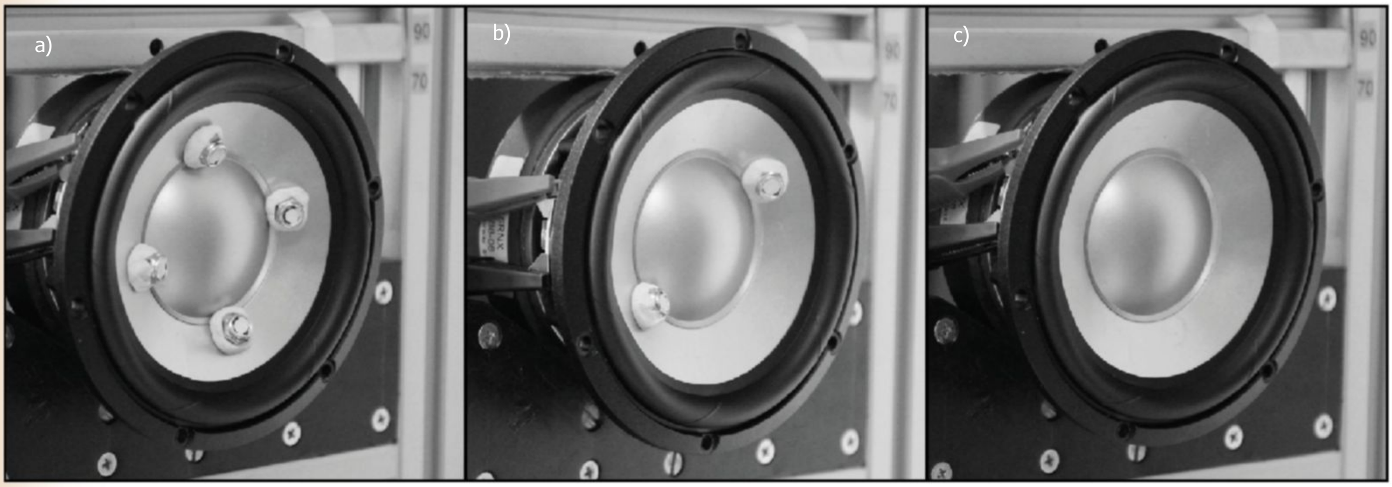

Photo 1: Three impedance measurements are performed in free air, with the driver secured in a suitable jig/workbench.

as well as DIYers—few people do box simulations using the advanced model.

The reason is clearly that it's quite a jump to throw the Thiele-Small (T-S) parameters overboard and at the same time advance your skills in box modeling sufficiently to actually achieve an improvement. The authors have advocated for some years to improve the simulation of loudspeakers with advanced parameters. We have published the relevant theoretical details and overcome certain obstacles; most notably, how to measure the viscoelasticity without expensive equipment, and how to work with the advanced model in the time domain.

Speakerbench: A Free Web-App for the Community

The first step in our proposed modeling workflow is to compute the advanced model parameter set, and for this purpose we created the Dual-Added-Mass procedure. The details of this procedure are described in a recent AES article [2]. The approach is verified to be capable of determining Bl within 1.0% and M_{MS} within 2%. In particular, the viscoelastic beta parameter, as well as the advanced electrical parameters, are extracted from the measured data.

The procedure offers a formal sanity-check for the measurement quality, unlike the classical added-mass method for which seasoned engineers would normally rely on experience and intuition (i.e., a gut feeling) to assess the quality of the extracted model parameters. Although the dual-added-mass procedure has been well-documented, it is non-trivial to design an algorithm to robustly carry out the parameter fitting and make an objective error estimate.

Thus, to facilitate use and adoption of the advanced model, we have created a web-application named Speakerbench, available online at speakerbench.com. The user must only perform three impedance measurements with suitable equipment, and Speakerbench does the rest. The transducer data from the app is provided in a compact JSON (JavaScript Object Notation), a lightweight data-interchange format, and can be downloaded and shared across the globe. Unlike DLC Dumax, LinearX, and Klippel data, the results from Speakerbench are valid only in the small-signal domain, but the data is of high quality and suitable for box design. After the advanced parameters are computed, the user can carry out box simulations within Speakerbench, or export to dedicated design software (e.g., VituixCAD).

Steps in Creating an Advanced Transducer Model and Datasheet

The key requirements for a good result with Speakerbench are that you have a sufficiently accurate setup for impedance measurements, and a quality scale for measuring the added masses. Measuring impedance should either be executed with a stepped-sine where each measurement is allowed to stabilize, or alternatively in a slow sweep (say minimum 10 seconds and preferably with a sweep profile that gives you more time at the lower frequencies). Measuring with 0.1 gram precision is mandatory for normal transducers with M_{MS} in the 10-25 grams range.

Let's walk through the process. Assemble four masses (using, for instance, a combination of Blu Tack and nuts) to be added to the diaphragm of the transducer. We recommend that total mass, which we denote by m_2, be close to the moving mass of the driver. This is just a guideline and not essential.

1: Attach the four masses, distributing them evenly near the voice coil (or just outside the area where the dust cap is attached) to ensure the load is balanced evenly avoiding rocking modes, as illustrated by the **Photo 1a**.

2: Measure the impedance of the transducer in free air, with all four pieces attached to the cone (Photo 1a). This is the m_2 impedance measurement.

3: Gently remove two diagonal pieces such that about half the added mass remains on the cone (**Photo 1b**). Remeasure the impedance with the remaining mass (m_1) on the cone. For reference, weigh the removed mass (m_2 - m_1) on a precision scale. When you remove the masses, avoid moving the cone (it should remain in its rest position).

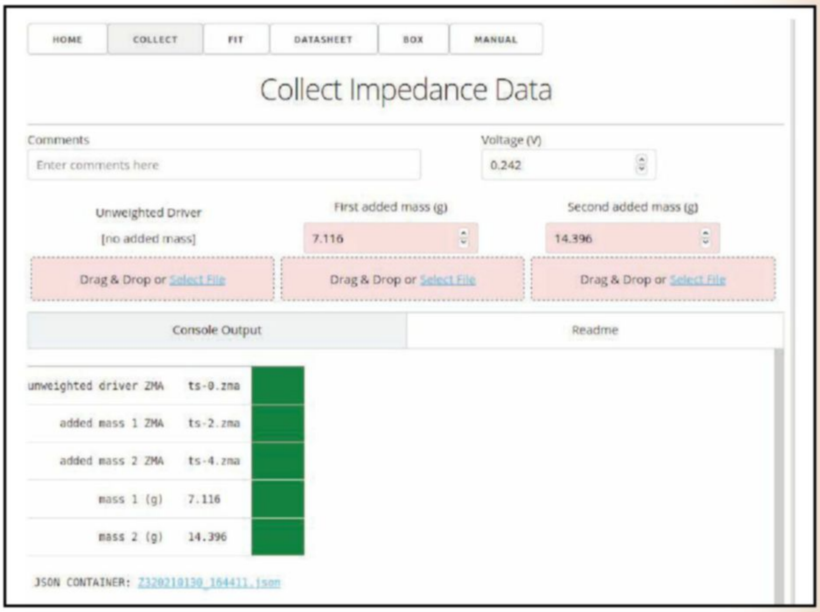

Figure 1: Illustration of the COLLECT app. Three impedance measurements, along with the two added mass values, m_2 and m_1, are uploaded and combined into a single Z-file.

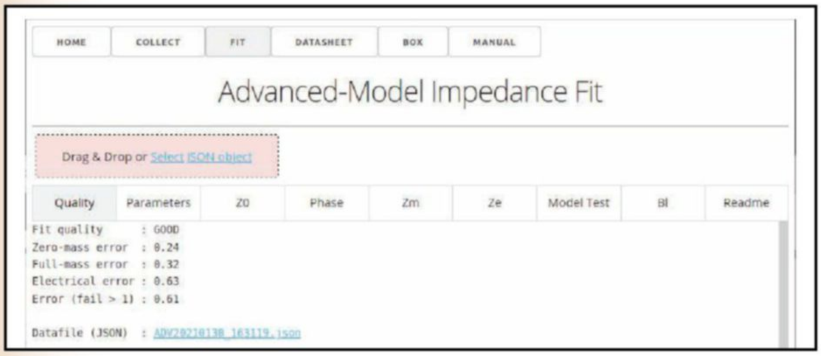

Figure 2: Illustration of the FIT app. Using an uploaded Z-file, this app computes the advanced fit and makes the results available in a ADV-file. A quality assessment is also given.

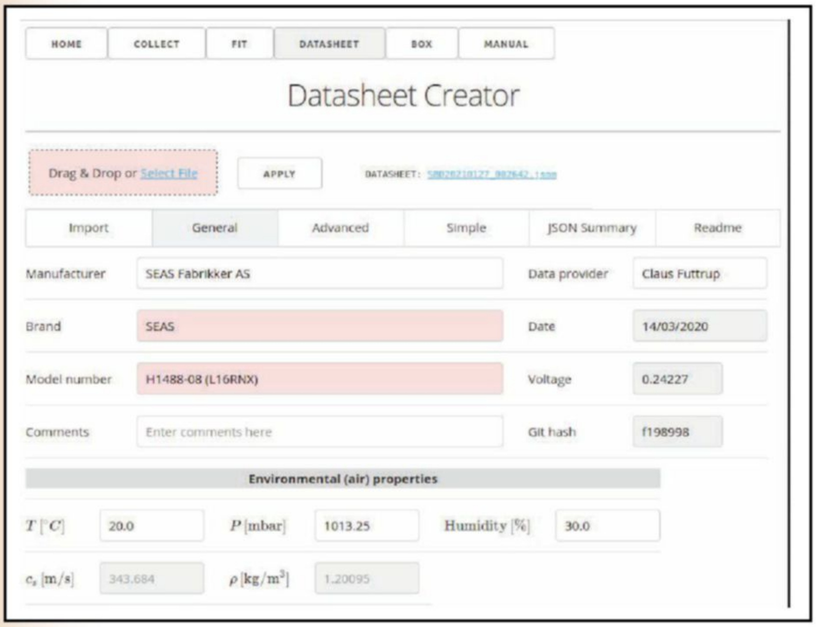

Figure 3: Illustration of the DATASHEET app. In the General tab you specify a description of the driver, etc. In the Advanced tab you specify S_D and in the Simple (= Legacy T-S) tab you may specify X_{MAX}. The datasheet is made available for download with a hyperlink (in blue) to the JSON SBD-file (date and time stamped).

4: Gently remove the remaining two masses (**Photo 1c**) and remeasure impedance. This is the unweighted measurement. Weigh the removed mass (m_1) on a precision scale. Sum the two weight measurements to calculate the total mass, m_2.

5: Upload the three impedance measurements and the weight of the masses (m_2 and m_1) into the COLLECT app (**Figure 1**), then download the resulting JSON Z-file.

6: Upload the Z-file from Step 5 into the FIT app (**Figure 2**). Speakerbench will then compute the advanced model parameters, as well as a quality-of-fit metric that can be assessed in the resulting plot windows. The advanced parameters are provided in a downloadable JSON ADV-file.

7: Upload the ADV-file into the DATASHEET app (**Figure 3**). Here you simply add a description of the driver you've measured and a few important parameters, like S_D and maybe X_{MAX}, and the datasheet is then complete, provided in a downloadable JSON SBD-file.

Quality Assessment

By performing three impedance measurements instead of two (as in the classic added-mass method), the pure motional impedance can be extracted with a minimum of model assumptions. From this pure motional impedance, the viscoelastic beta parameter is then determined. The remaining blocked-electrical impedance can be derived by a relatively simple yet powerful set of algebraic operations.

As part of the analysis, the algorithm attempts to extract a stable value of Bl over a region of about one octave around the resonant frequency. This gives a useful indication of the robustness of the overall fit.

After the fitting procedure is complete (Step 6) the fit quality is automatically assessed.

About the Authors

Claus Futtrup received his MSc in ME from Aalborg University in 1997 and has worked in the loudspeaker industry ever since. From 1997 to 2006, he worked at Dynaudio A/S first as an R&D Engineer, designing loudspeaker boxes and later as a System Engineer for automotive. From 2006 through 2007, Claus was employed as a Transducer Design Engineer at Tymphany in Denmark and in 2008-2013 he was R&D Manager at Scan-Speak. From 2013 to 2015 he worked as Technical Sales Manager for SEAS, Norway, and in 2015 he was promoted to Chief Technical Officer. From August 2021, Claus moved back to Denmark and started working at DALI as manager of the acoustics team.

Jeff Candy was born in Edmonton, Canada in 1966. He received his Ph.D. in Physics from the University of California, San Diego in 1994, and is currently Director of the Theory and Computational Sciences Group at General Atomics in San Diego, CA. He works primarily on theoretical and computational plasma physics, with focus on plasma kinetic theory and turbulence. In the field of audio, he is interested in the application of theoretical acoustics to practical situations. Jeff is a member of the Audio Engineering Society (AES) and a fellow of the American Physical Society (APS). In 2003, he received the Rosenbluth Award for fusion theory, and was 2008 Jubileum Professor at Chalmers University.

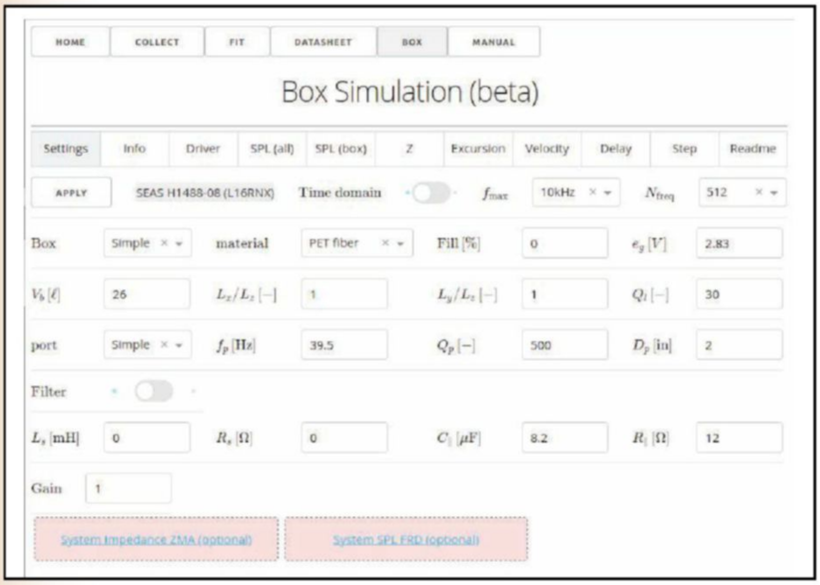

Figure 4: Illustration of the BOX app. In the box simulator you may specify the usual box parameters, including damping material, input voltage, box volume, port model, tuning frequency and even a simple passive filter. At the bottom, you can upload measurements and compare with the simulation.

In the Quality tab of the FIT app, a fit rating is provided (Figure 2). This means you don't have to be an expert in analyzing the output from the fitter. Possible ratings are (1) EXCELLENT, (2) GOOD, (3) FAIR, and (4) FAIL. Beyond these simple ratings, the user can also study the graphs in the succeeding tabs, where the app provides plots of the motional impedance (Z_M), the blocked-electrical impedance (Z_E), a model test that essentially compares your three measurements with the simulations based on the established model parameters, and Bl over the fit range. To get an excellent rating probably requires some experimentation and building experience with the added mass approach. Once you can achieve an excellent rating, it's worth celebrating (with a beer or other beverage). The good rating is also sufficient for reasonably accurate box simulations. The Speakerbench article in the 2020 *Loudspeaker Industry Sourcebook* [3] provides more information.

References

[1] C. Futtrup, "Losses in Loudspeaker Enclosures," Audio Engineering Society (AES) Convention: 130 (May 2011), Paper Number: 8324, Publication Date: May 13, 2011. www.aes.org/e-lib/browse.cfm?elib=15791

[2] J. Candy and C. Futtrup, "An Added-Mass Measurement Technique for Transducer Parameter Estimation," *Journal of the Audio Engineering Society* (JAES), Volume 65, Issue 12, pp. 1005-1016; December 2017. www.aes.org/e-lib/browse.cfm?elib=19366

[3] C. Futtrup and J. Candy, "Speakerbench," *Loudspeaker Industry Sourcebook* (LIS) 2020, page 124-131. www.loudspeakerindustrysourcebook.com/articles/speakerbench

Box Simulations with Speakerbench

Speakerbench wasn't initially intended to do box simulations, but there were numerous motivations to offer this functionality. Once an SBD file is created or uploaded, the transducer parameters are automatically forwarded into the BOX app (**Figure 4**), and an initial (probably bad) proposal for a box is provided. Just push the BOX button at the top. You can then explore the Settings to find an optimal configuration.

JSON Format and Portability

Speakerbench makes use of JSON as a standard container for calculation and datasheet output. This way you can easily share datasheets and simulations with fellow speaker designers. An online database of datasheets (SBD files) could easily be deployed, if desired. If someone sends you a Z-file or ADV-file or SBD-file, you can download it to your smartphone and then drop it into Speakerbench running on a browser.

Why Speakerbench?

What makes Speakerbench unique is that high-quality transducer data can be ensured even for the DIY speaker builder. The improvement in accuracy with the advanced model can be readily achieved, and should prove useful particularly when quantifying the various box losses. The user should be able to easily and accurately determine a suitable box volume. Speaker designers (DIYers or professionals) rarely build more than one box to determine the optimal volume for a loudspeaker system. With that one prototype box, you just have to make things work. This explains why it is so important to determine the best box volume from the very beginning.

Moreover, Speakerbench is a web-app and works on any computer, tablet, or smartphone with a browser without any software installation required. The online version will always be up-to-date with the latest one we have made available. Speakerbench collects the research and the proposed methods into an easily accessible software suite. We do the software development, as well as the underlying research and modeling, to help the industry move forward and improve their methods.

It is our hope that Speakerbench will facilitate the jump from legacy Thiele-Small parameters to more advanced modeling, and that people will feel like giving it a chance. A brief manual is available on the https://speakerbench.com website, and in addition we have created a series of short tutorial videos available on YouTube (a link is provided in the online manual).

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Tribute

Roger Russell (1935–2021)

A Lifetime of Passion for Sound

A few weeks ago we received notice about Roger Russell's passing. Since I've been working with *audioXpress* I only had received one direct message from Roger Russell, inquiring if our requirements for submitting a manuscript had changed. Approximately a year later, we received a submission from him.

In his original message, Roger mentioned that he wrote several articles for *The Audio Amateur*, *Speaker Builder,* and *audioXpress* over the years. Since that exchange, I gradually revisited his articles and was surprised with the range of topics addressed. The most recent articles focused on his work designing large vertical array speakers.

His latest article submission was basically a recap of his work with vertical columns. After we learned about his passing I decided to publish it in this Speaker Focus edition (p. 44).

Photo 1: Roger Russell is shown with his speaker lab testing equipment at McIntosh in 1969. (Image courtesy of Roger Russell's website)

After graduating in 1959 from Rensselaer Polytechnic Institute in Troy, NY with a degree in Electrical Engineering, Roger Russell was employed at Sonotone, from 1959 to 1967. The highlight of Roger's career was being Director of Acoustic Research at McIntosh Laboratory, Inc., and being the originator of McIntosh Loudspeakers, which were mainly vertical arrays (**Photo 1**). From 1967 to 1992, Roger created the C26 preamplifier and numerous electronics for McIntosh, before focusing his attention fully on speakers. His work in this field was notably early and he remained uninterested in the research that took place subsequently in vertical and line arrays, particularly for professional audio applications.

In the article he submitted to *audioXpress,* there was a note that stated: "Column directionality has been very useful in commercial sound installations for many years. It can beam sound further out into the audience and improve clarity. It can be located high up and have less attenuation with distance. The length, width, and bandwidth can be chosen depending on the application. In addition, compensating delays can be used to present the same sound at the same time at different distances."

After leaving McIntosh, Roger remained focused on his purpose to achieve the ideal stereo loudspeaker system for home listening and the result of his research with Carl Van Gelder was the IDS-25 single wide-range column system. Describing the system Roger wrote "I never cease to be amazed at the captivating sound, even in the first few seconds of listening. This system outperforms all of my earlier designs, whether it is a column or conventional multi-way type."

When researching for friends who have might have worked and met Roger Russell I emailed Don Keele, for the obvious connection with his work on line array designs. Don admits that their paths never crossed, and he regrets they never did. Don was particularly curious about Roger's favorite system, the IDS-25 and he sent me this note, which I think is of value. "This is a straight floor-standing array housing 25 small wide-range drivers all equally driven in a series-parallel connection scheme. For straight line arrays (e.g., the IDS-25) to work properly, your ears must be in the nearfield of the array. This is taken care of by making the straight-line array very tall (say 6′ to 8′ tall) insuring that your ears are in the nearfield of the array for average listening distances up to 10′ away and most listening heights."

Roger's work with McIntosh and subsequently with his own IDS brand and products, are still highly praised in hi-fi and DIY circles. On his personal website, Roger wrote extensively about his designs and their history and we highly recommend reading the many interesting articles contained there. In fact, I believe that entire website should be preserved as a book—just hoping that the many photos and illustrations can be found in better quality.

The article we publish in this tribute is a personal review of his career as a loudspeaker designer and his proudest achievements.

—J. Martins

*****

Roger used to say "everything in life has compromises" and, of course, loudspeaker design was included (**Photo 2**). Although he enjoyed loudspeaker design, Roger had a wide range of interests including family, his own McIntosh history website, a special pencil lead, a special type of clock, art, a glacial conglomerate, fulgurite, general semantics, philosophy, science fiction books, music, Bob and Ray, and so much more.

Photo 2: Roger Russell and Carl van Gelder pose with McIntosh speakers in 1983. (Image courtesy of Roger Russell's website)

I remember a McIntosh employee once came to Roger and started their conversation with "Roger, I know you to be a very particular man." Roger smiled seemingly with a combination of amusement and appreciation. The employee had come to Roger because his attention to detail was well known and respected by that employee.

One time from Arizona, Frank McIntosh called Roger with a question. Near the end of the conversation, Mr. McIntosh told Roger "if you are ever in Arizona then come see me." By pure chance, Roger was already booked to fly to Arizona the next day on an unrelated issue…Roger did not tell Mr. McIntosh this…imagine his surprise to see Roger the very next day. Did Roger have a sense of humor…sure bet he did.

Roger also had an emotional side—he wept at the funeral of McIntosh's president Gordon Gow.

And, Roger was patient in his teachings about loudspeakers and acoustics. His teachings were the foundation upon my understanding that has carried my career and thoughts even now. Long after Roger left McIntosh and moved to Florida, we continued with a deep friendship that started more than a decade earlier as a simple boss-employee relationship. Although there will be no more emails between us, I will cherish and appreciate the times we had together and remember Roger as the wonderfully outstanding man that he surely was.

—Carl Van Gelder

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Roger Russell was an unsung hero of loudspeaker design and development. He started the McIntosh Loudspeaker Division with the ML1C and continued all the way through the XR290s, developing not only an environmental equalizer/analyzer that matched to speakers in the room, but also speakers utilizing a bass equalizer that made a subwoofer unnecessary. He may have been the first loudspeaker engineer to bring a line source speaker to the home. Each speaker design was a logical extension of the past—always improving frequency response, phase cohesion, and dispersion while lowering distortion.

I first met Roger in the early 1970s when he was the Chief Loudspeaker Engineer at McIntosh. At that age he seemed like a god to me. I loved building and modifying loudspeakers and he patiently answered my questions about various projects. His designs were always well-thought-out acoustically, electronically, and mechanically. Many of the speakers he designed are worth as much today as they were new, an astonishing achievement in the speaker industry.

Roger could be described as kind, intelligent, soft spoken, humble, and knowledgeable. If you asked Roger about the differences in speaker cone materials, or cast vs. stamped metal baskets, or metal vs. paper, or soft domes, Roger would patiently explain the answers in layman's terms. If you wanted scientific back-up, he could provide it. He readily answered my questions up until the end, and I'll greatly miss him.

—Steve Rowell, President,
www.AudioClassics.com

*****

I became an audio aficionado at a very young age. Of course, speakers were a top priority then, as they are now. As I made my progression through various speakers, the designs of Roger Russell, the acoustic mastermind at McIntosh Labs, struck me as the purest. So much so that over the course of time I became the proud owner of three different R.R. speaker systems (**Photo 3**), including the XRT20, the XR250, and the XR290 systems.

My relationship with McIntosh was such that, although I was a consumer, I had a relationship with a number of people at the company, which included Roger. When I first heard his columnar design, I loved the sound and was the first person in Michigan to own XRT20s.

After Roger retired, I visited him at his home a few times in Florida and during one of my visits he demonstrated for me his new design, the IDS-25. I was stunned by the clarity and told him it was the best speaker I had ever heard. I just had to have these in my home and currently own two pairs. A single column design, of identical wide-range drivers where the entire frequency range radiates from the same line source product exceptional sound. IDS-25s are an extension of two of Roger's patents, and they are still in production to this day.

Working with Roger brought me a depth of knowledge few others can share. I will always treasure my friendship with Roger and enjoy his special gift at reproducing sound.

—Ken Haig

*****

In 1967 my family moved to Binghamton, NY where my father, Roger Russell, accepted a position at McIntosh Laboratory as Director of Acoustic Research. During his tenure there he designed and developed its line of loudspeaker products. As far back as I can remember we always had prototype speakers at the house, some of radical design such as multi-step, angular, and even curved. As McIntosh invested more in precision test equipment and a large anechoic chamber, my dad's designs became more refined and of course had to keep up with the legendary high-powered amplifiers that the company produced.

Dad was always in search of the best recorded music to play on the systems and growing up we were exposed to Kraftwerk, Mike Oldfield, and many others. In one instance I recall him ordering a just released copy of Pink Floyd's *The Wall* and had it shipped by bus from New York City. Over the years he held several patents and turned toward column design as evidenced in the McIntosh XRT-18 and the massive XR-290 systems.

After he left McIntosh, dad retired to Florida with his wife Joan and he kept active by writing magazine articles and he ultimately developed the IDS-25 column speaker systems. The "IDS" stands for Imaging, Depth, and Spaciousness—three concepts that he felt made the best representation of recorded music. The "25" represents the number of full-range drivers in each column. He upheld that the column arrangement was the best design possible for sound accuracy.

He maintained a website, www.roger-russell.com, where he published content about McIntosh's history and the many other hobbies in which he was interested. The website serves as a reference for many fans of his work and will continue to be maintained going forward.

Roger Russell was a pioneer in his field and passionate about loudspeakers—their design and accuracy of reproducing sound. He will be greatly missed but his contributions to the audio world will live on in perpetuity.

—Dan Russell

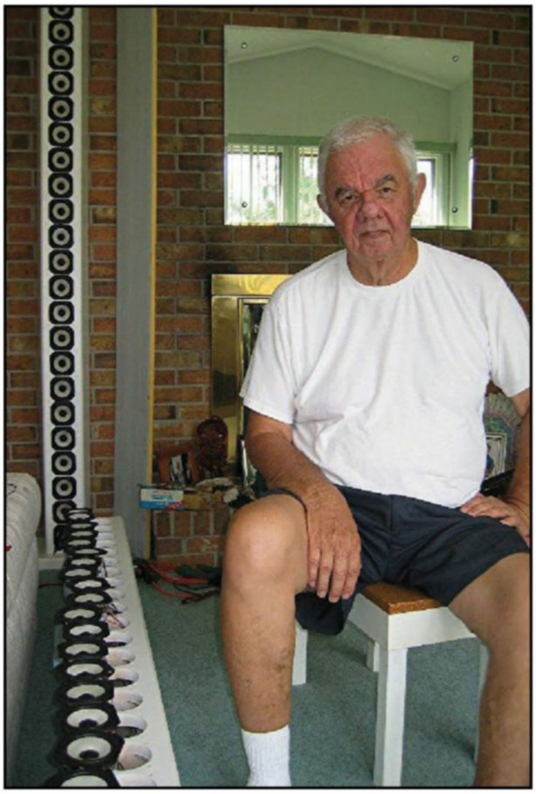

Photo 3: Roger Russell is shown with one of his RR speaker systems. (Photo courtesy of Ken Haig)

The Sound of a Column

By Roger Russell

A column of tweeters can solve a tweeter power handling problem and can also lead you into other benefits as well, including systems made with three columns or only a single wide-range column. Several different systems specifically for the home are described using the advantages of the column concept. The Haas effect explains why the speaker column and human ear work so well together. This effect has also become a vital part of the recording industry. The column concept describes many aspects of sound quality compared to ordinary systems. It can affect such features as increased power handling, lower distortion, improved transients, useful directionality, improved imaging, depth and spaciousness. Some listeners find the unique qualities of columns to be fascinating. All of these features combined together can help to present a significant attraction to the sound of a column.

To present a more accurate reproduction of what we might hear at a concert, more details have been needed about the original instruments and the sound they can create. Several years ago, Gordon Gow, the Executive Vice President of McIntosh, felt the truth could be found directly from the "Father of Stereo Sound," Dr. Harvey Fletcher at Brigham Young University in Provo, Utah. Gordon arranged for Sidney Corderman and me to visit with Dr. Fletcher.

Photo 1: The McIntosh XRT20 was the first McIntosh column system to be sold.

In 1933, Leopold Stokowski cooperated with Dr. Fletcher in a successful live demonstration of stereo sound via telephone line transmission. This includes a performance of the Philadelphia Orchestra in Philadelphia to Constitution Hall in Washington D.C. where the sound was separated into three channels to achieve realism. Dr. Fletcher showed us the speakers that were used to play at Washington Hall.

He also described his earlier research on the spectrum of real musical instruments. His measurements show that peaks for mid and high frequencies from some instruments can be just as pronounced as lows from others. This information indicates that speakers must be able handle high power peaks in order to be accurate. I verified this later with a modern Bruel & Kjaer real-time analyzer that has a peak hold feature.

I tested real orchestral instruments from bass drum to triangle several times. New ways of creating sound have put even more demands on reproducing sound faithfully. Only one tweeter is typically used to cover the high-frequency range but many identical tweeters can handle higher power with lower distortion. Arranging all of them in a single long column for the home is an ideal choice.

Dispersion left and right is the same as for a single tweeter. The vertical response of a floor-to-ceiling column is similar to a half-cylindrical radiator over most of its range and decreases by only 3dB at double the distance. This is definitely a plus compared to spherical radiation from a single tweeter that decreases at 6dB at double the distance. Distortion can be extremely low for 25 tweeters. However, sometimes controlled distortion is intentionally added for guitar or other amplified instruments to give a strong expression of emotion beyond limits of the equipment.

But that is not all. You will find that you hear the same sound at any listening height. This is explained by the precedence effect and means that if there are many, drivers radiating the same sound, the source will be localized to the driver that provides the earliest arrival. The other drivers will not be heard at all in some cases. This is further explained by the Haas effect [1]. Five milliseconds after the first arrival sound, it will be attenuated by the brain for about 50ms. It is this delay that prevents other sounds from interfering even if they are up to 10dB louder. However, for time delays above 50ms (for speech) or up to 100ms (for music) the delayed sound is then perceived as an echo. So, if you stand or sit, the nearest driver will be the one opposite to your listening height. In fact, this is true for any listening height and a long column. This ear/brain combination may all be part of our built-in heritage for survival in primitive times.

The McIntosh XRT20

The McIntosh XRT20 was the first McIntosh column system to be sold (**Photo 1**). It consists of a long column of 24 closely spaced 1″ soft dome tweeters and was an answer to the single tweeter power handling dilemma. The column is combined into a system called the XRT20. The credenza-style bass cabinet contains one 8″ midrange at the top in a sealed enclosure and two 12″ woofers are in line below. Conveniently, a series-parallel connection of 25 tweeters has the same 8Ω impedance as a single tweeter. However, the column is divided into two equal sections of 12 each and the 25th tweeter is an 8Ω resistor at the bottom. The sound is not audibly changed with the resistor and the power handling remains the same.

The tweeter column was designed to be mounted on a wall. For those who could not use wall-mounting, a stand was available. The column could be attached to either side of the bass cabinet. This way, the user could arrange the left and right systems to be a mirror image of each other and preserve stereo imaging. Floor and ceiling reflections for highs are greatly reduced with this column and is a distinct advantage. The column frequency range of 1,200Hz to 20,000Hz is significant to be able to reveal some of the illusion of depth and spaciousness in stereo recordings.

McIntosh XR19

For those who didn't have the space for the XRT20, the same bass cabinet could be used with a vertical column of 12 tweeters, the XR19 system. The XR19 system is made symmetrical to provide good imaging, and tweeters are placed in a semi-elliptical row. The furthest tweeter is then at 90 degrees from the front of the cabinet. However, this arrangement does not present the same sound from floor to ceiling like the XRT20 system and exhibits a limited illusion depth and spaciousness effect.

Felt strips are added above and below the tweeters to reduce reflections from the bass cabinet and tweeter grille frame. The system also included two 12″ woofers, and one 8″ mid. The crossover network conveniently fits under the tweeter frame. Like the XRT20, the bass grille covers three sides and is held to the cabinet with Velcro fasteners.

McIntosh XRT18

A short vertical column is a compromise for size, space, power handling, and cost. This column of 16 tweeters has vertical directional effects that begin past the ends where the output decreases. To preserve the performance, the height is selected to cover a normal listening area including standing and sitting positions. The column can be hung on the wall. An optional tweeter stand was available that places the top of the tweeter frame at about 88″ from the floor.

The flat tweeter grille is made of steel. Holes are made large enough for each tweeter not to affect radiation. Interestingly enough, the tweeter magnetic field holds it in place even with grille cloth wrapped around it. The 6″ midrange is mounted near the top of the bass cabinet in a sealed enclosure. A 12″ woofer completes the system. This orientation makes a symmetrical system and insures good imaging. In addition, it also enables the system to produce some of the illusion of depth and spaciousness.

McIntosh XRT22

You might wonder why McIntosh changed to the XRT22 (**Photo 2**) when the XRT20 was so popular. For one reason: the speaker lab was moved from

Photo 2: The McIntosh XRT202 was a single-piece veneered tweeter column that could be hung on a wall or on a separate CS22 stand on the floor.

plant 5 to a renovated bowling alley in front of the main plant. This included a full-sized anechoic room to make even more critical improvements in a reflection-free environment (**Photo 3**). It also included another reverberant room that was very important to measure the total radiation from all directions in a single measurement.

It also created an opportunity to introduce the newly patented McIntosh LD/HP woofer. This had a noticeable reduction in bass distortion, clearly audible with voice.

Photo 3: Roger Russell and Carl van Gelder were photographed in the anechoic chamber of the McIntosh speaker lab building.

References

[1] H. Haas, "The Influence of a single echo on the Audibility of Speech," *Journal of the Audio Engineering Society* (JAES), Volume 20, Issue 2 pp. 146-159, March 1972.

[2] R. Russell, "A Unique Stereo Column System," *audioXpress*, November 2005.

[3] R. Russell, "Use of Equalization to Compensate for the Natural Low Frequency Rolloff of a Loudspeaker System," US Patent # 3,715,591.

Resources

R. Russell, "Upgrading a Unique Stereo Column System," *audioXpress*, July 2006.

R. Russell, "Hunting the Elusive Transient," *audioXpress*, July 2020.

R. Russell, "Stereo Speaker System for Creating Stereo Images," US Patent# 4,267,405

Links of Interest:

Equalizers by Roger Russell | www.roger-russell.com/equalizers/equalizers.htm
IDS by RRussell | www.ids25.com
McIntosh History Site Map| www.roger-russell.com/sitemap.htm
McIntosh Loudspeaker Division Part 2: A History (1976 and up) | www.roger-russell.com/lsd2.htm
Roger Russell homepages | http://roger-russell.com

The single-piece veneered tweeter column can be hung on a wall or on a separate CS22 stand on the floor. Like the XRT18 tweeter column, magnetics hold the grille in place. The grille frame is a constant width to avoid changing response with listening height and losing some of the illusion of depth and spaciousness.

At the time, one dealer created a very unique application to keep the sound concentrated on the dance floor in a disco. The columns were suspended horizontally overhead about 9′ from the floor. The bass cabinet was located against the wall. By walking beyond the ends of the columns, all the highs could be heard to disappear. This directional feature could provide plenty of highs directly over the dance floor but not in other areas where they were not wanted. The columns could be made longer with sections to cover as much area as needed.

McIntosh XR290

The XR290 speaker system was the first McIntosh triple column system. A long column for the mid-range is added. It has greater power handling and lower distortion for all three columns. It replaced the earlier XRT22 and XRT22 systems.

The XR290 includes a column of four 12″ woofers, 12 5″-mids and 25 1″ soft dome tweeters. It takes up less floor space having no external tweeter column.

It was also a replacement for the earlier ML4, a system that had four 12″ woofers, four mid domes, an 8″ midrange, and two tweeters. It was a drastic improvement in sound with an overall power handling of 1200 peak watts. Although the prototype cabinet was all in one piece, the system was divided in two pieces for shipping and convenience in assembly. To preserve imaging accuracy right and left versions were made. The tweeters are normally located at the outside position. The rule for the first sound to arrive still applies and indicates the apparent sound source. The additional midrange column can bring out more of the depth and spaciousness in a wider frequency range and the sound stays the same at any listening height. This system definitely contributed to a more attractive sound.

IDS-20 Column Speaker System

After retiring, I continued to pursue my passion for column systems. This time, using a single vertical line of 20 4″ wide-range drivers. The IDS-20 enclosure is about 7′ tall. Each driver is identical and handles the full frequency range. No crossover is needed. The effective area of these small cones calculates out to be the equivalent of two 12″ woofers [2].

The area is related to the acoustic resistance [energy permanently imparted to the air]. While this resistance declines below resonance, the combined area is very large compared to that of a single driver.

Admittedly, claiming to cover the whole frequency range with common 4″ drivers is not realistic. In fact, the drivers barely make it to 10kHz, let alone down to 20Hz. To make the drivers work, the other part of the system was the analog equalizer. The low-frequency contour is the mirror image of the close microphone response. The result is a shocking 18dB boost increasing below 150Hz and down to 20Hz [3]. However, there is a 10dB acoustic gain at lower frequencies compared to a single driver. A slight high frequency increase helps to correct the highs.

This, plus the system overall efficiency, is another advantage for the single column. The system is still able to reproduce effortless bass at high listening levels and without any audible stress from the drivers. This was supported by listening as well as measurements. The system can handle a maximum of 200 peak watts from 20Hz to 10,000Hz. Despite the correction for extended low and high frequencies, the single column was an improvement for Imaging, Depth and Spaciousness. In fact, I later choose the letters IDS to represent the improvement that could be heard.

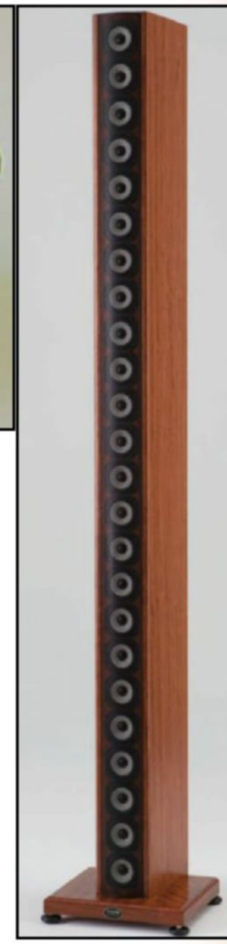

Photo 4: After retirement, Roger Russell pursued his passion for column systems, developing the IDS-20 and later the IDS-25 Column Speaker System shown at right. Above is the original Vifa 3.5″ wide-range driver selected by Roger Russell for the IDS-25 design.

IDS-25 Column Speaker System

Further research showed that a Vifa 3.5″ wide-range driver (Vifa TG9FD10-04) had better features, including a mid-range that was extremely flat (**Photo 4**). The high end has a rising response above 5kHz to a 10dB maximum at 18kHz on axis but flattens by 30 degrees off-axis. It also has a copper ring around the pole piece that can reduce distortion, a magnetic shield, a Qts of 0.89, a woven fiberglass cone, a long-excursion voice coil, and a surround that will not deteriorate. The moving parts of each driver weigh as little as 2.4 grams. Impedance is 8Ω, random noise power is rated by Vifa at 30W each. For a protection circuit, I included a solid state Polyswitch in series with the input.

The IDS-25 system does require a custom equalizer for the same reasons as the IDS-20 (**Photo 5**).

In the column there are 25 drivers making a total area equal to that of a 16″ woofer. The cabinet cross section is trapezoidal, allowing just enough space for a front grille.

The system can be placed 2″ or 3″ from the wall and has a footprint of about one square foot. It does not need to be toed-in. Near field decreases at only 3dB at double the distance making it excellent for small or larger rooms.

The IDS-25 system also provides more uniform bass. Improved transients can be found at all frequencies. Imaging is very accurate and easily reveals panning. Depth and spaciousness are outstanding. However, limited power handling is at 200W peaks for the lowest frequencies. Sensitivity is high at 92.5dB thanks to the acoustic gain at lower frequencies.

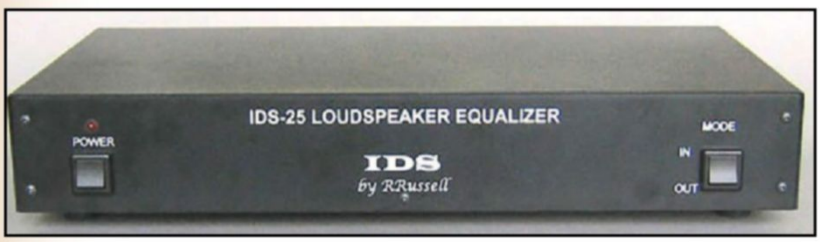

Photo 5: The IDS-25 system does require a custom equalizer

Conclusions

A column of drivers works well with your own ears and brain. It can affect such features as increased power handling, lower distortion, improved transients, useful directionality, improved imaging, depth and spaciousness. It makes the sound captivating compared to ordinary systems. This presents a significant attraction for the sound of a column.

Roger Russell (1935-2021)

"I have been referred to as a renaissance man perhaps due to curiosity about various things and learning more as the truth is often hard to find."

Author, Artist, Engineer, Inventor, Photographer, Collector, and formerly Director of Acoustic Research at McIntosh Laboratory, Inc. and the originator of McIntosh Loudspeakers.

Interest in sound, for Roger Russell, may have started as early as fourth grade when he was taking violin lessons. By ninth grade, he became a devoted listener to all kinds of classical music. In the early 1950s, good sound was hard to find and nothing seemed to accurately reproduce a violin.

There were all kinds of speakers available and they were all lacking in one quality or another. There had to be something better and he was determined to find a way to improve speaker performance.

His first exposure to stereo recordings was in the early 1950s when he heard Emory Cook's binaural demonstrations and the AM-FM stereo broadcasts from WQXR in New York City. He was soon using a Viking stereo tape deck and two Telefunken dynamic microphones to make stereo recordings of a Wurlitzer pipe organ at the Lowes Theater in New Rochelle, NY. This was later followed with a Magnecord professional recorder that he used to record a few concerts, thunderstorms, and even diesel trains at Harmon, NY.

After graduating in 1959 from Rensselaer Polytechnic Institute in Troy, NY with a degree in Electrical Engineering, he gained experience while working as a senior engineer in the Audio Products Section of the Sonotone Corp. in Elmsford, NY, a well-known manufacturer of hearing aids and other audio products. He spent 8 years there designing a new line of loudspeakers and microphones. Russell also wrote his first two magazine articles about a stereo volume expander-compressor and sealed speaker system design.

Following this, he was Director of Acoustic Research at McIntosh Laboratory for 25 years. While there, he designed the C26 preamplifier in 1967, a wide variety of speaker systems in the late seventies and eighties, and was awarded several patents.

After devoting much of his life to designing loudspeakers, crossover networks and cabinets, Roger Russell arrived at the simplest and yet most effective of all of his creations, the IDS-25 stereo loudspeaker system.

He wrote several articles about loudspeaker design, testing, and the hearing system. Several patents have been awarded for some of his designs. He was a member of the Audio Engineering Society and the International Society for General Semantics for many years. Besides his love of stereo sound and music, his other interests range from photography to mystery clocks and science fiction.

Roger Russell died at the age of 85 on April 30, 2021.

You Can DIY!

Dual Speaker Columns

By
Ken Bird

In this simple speaker project, our veteran speaker builder and longtime contributor to *audioXpress* proposes a versatile dual speaker column design. These affordable speakers are a versatile choice for home listening and matching with home entertainment systems and can be easily modified with different driver choices.

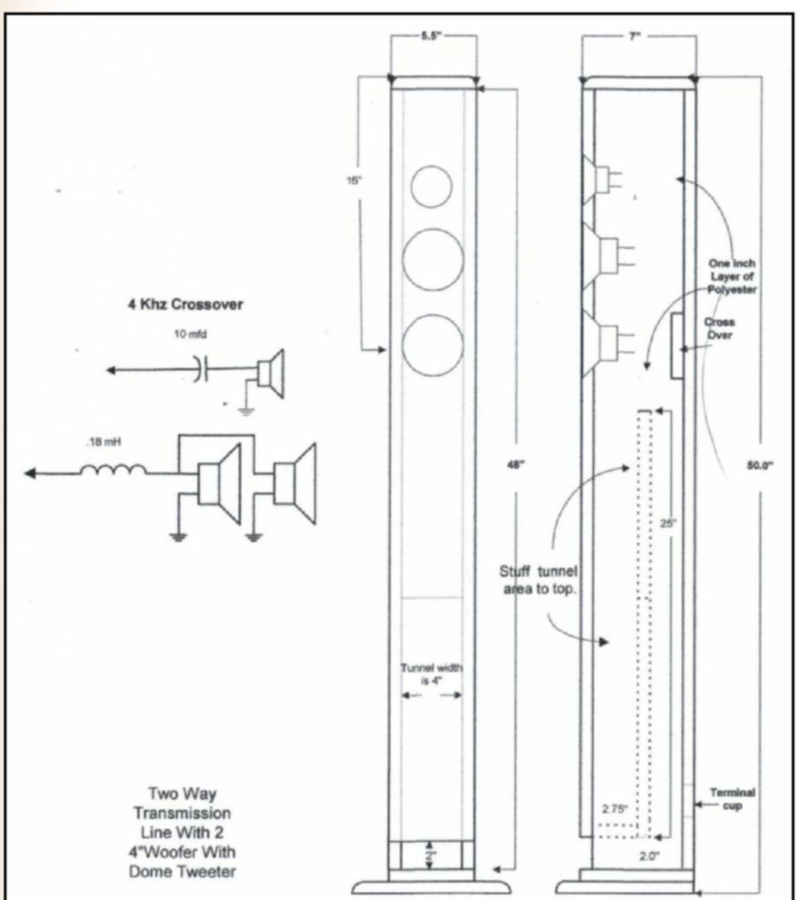

Figure 1: The crossover is a simple two-way design set at 3500Hz.

Column speakers have been popular since the early days of stereo. They usually provide great imaging, take up less space and can produce a good balance of sound. This system uses the principle of mass loading through a narrow duct to the port output. The walls of the enclosure are lined with 1/4″ indoor/outdoor carpet and small amount of polyester stuffing is employed for internal damping. The position of the drivers and the carpet lining reduces any multiple-order frequencies that might develop in the enclosure. Construction is simple, using dimensional lumber, and the drivers employed are not overly expensive. The narrow configuration and tunnel partition provide a solid cabinet with low resonance (**Figure 1**).

Required Drivers

The drivers that I chose for this project are the Hi-Vi 4″ aluminum woofers with a Dayton Audio 1″ soft dome tweeter (**Photo 1**). The woofers have a low Fs at 56Hz and have good response up to 5kHz. Crossing them in the vicinity of 3500Hz effectively matched the Dayton tweeter. They are stock items available from Parts Express and are detailed in the Parts List along with the crossover network parts. The crossover is a simple two-way design set at 3500Hz.

Construction

The speaker is constructed from 1′ x 6″ dimensional pine lumber (5.5″ x 0.75″) for the front panels and the side panels, and are assembled using 2″ finishing nails and wood glue. Other materials and assembly techniques can be used, just follow the same dimensions. Begin by cutting the sides and rear panels to a 48″ length. The front panel is cut to a 46″ length allowing 2″ for the port area. Cut two pieces from the 5.5″ stock to 7″ length for the top and bottom. The bottom stand section was cut 7.5″ × 9″ from a piece of 3/4″ plywood. The port tunnel is made from a piece of the 5.5″ stock ripped down to 4″ in width. The long section is 14.25″ and the short section is 2.75″ (**Photo 2**).

For this project, I cut the speaker mounting holes 3 and 5/8″ for the woofer and 2″ for the tweeters. I used a Jasper cutting jig with a router to make the cuts, but they can also be cut with a jig saw. The speaker terminal hole in the rear panel is 3″ in diameter (**Photo 3**).

Panel Assembly

Attach the "L" shaped internal tunnel to the front panel first, then staple the 1/4″ indoor/outdoor carpet to all the interior surfaces. Leave a gap on the back panel for the terminal cup wires and on the front panel for the speakers. For the front and rear panels, I made the carpet 3.5″ wide and the side panels 6.75″ wide. I used a carton knife and steel rule to make the cuts. Add 4 oz. of polyester fill to the bottom closed end of the enclosure (**Photo 4** and **Photo 5**).

Once the carpet is mounted, place wood glue or a bead of liquid nails on the edge of one of the sides and mount it to the front panel. Use 2″ finishing nails or 2″ brad nails in a nail gun. Corner clamps can be used to assure proper alignment for the panels. Run a bead of caulking down the inside seam

Parts List

Speaker Parts

4×	Parts Express Hi-Vi B4N 4″ 8Ω Woofers 297-429
2×	Parts Express Dayton Audio ND25FA-4″ 4Ω Tweeters 275-059
2×	Parts Express Terminal Cups 260-265
2×	Woofer crossover coil 255-208
2×	Tweeter capacitor 027-340

Lumber

4 each	7.25″ × 0.25 × 48″ sides
2 each	5.5″ × 0.75″ × 46″ fronts
2 each	5.5″ × 7.5″ × 48″ rears
4 each	5.5″ × 8.5″ tops and bottoms
2 each	4″ × 0.75 × 25″ tunnel parts
2 each	7.5″ × 9″ × 3/4″ plywood bottom stands

Miscellaneous

60″	18-gauge speaker zip wire
1/2 lb	2″ finishing nails
2×	Tubes Liquid Nails glue
2×	Tubes silicone caulk

Photo 1: The drivers used in the system include the two Hi-Vi woofers and a Dayton Audio 1″ dome tweeter. The crossover was hand wound and consist of a 0.18mH coil and a 10µF capacitor.

Photo 2: Shown are the wood parts for one enclosure.

Photo 3: Use a Jasper router jig to cut the speaker holes.

of each panel. With the front panel and one side attached, glue and install the 4″ panel, and nail it in place. Staple a 48″ length of speaker wire from the rear speaker terminal hole up to just behind the two speaker holes in the front panel (**Photo 6**).

Due to the enclosure's narrow interior, it will be difficult to caulk the seam of the rear panel. The best technique is to use a liberal amount of liquid nails instead of wood glue when attaching the panel. Before attaching the rear panel, place about 2 oz. of polyester pillow stuffing into the cavity formed by the tunnel and the front wall of the enclosure. Do not compress it tightly down, but let it loosely fill the gap. Do not place any stuffing in the tunnel outlet. Then, run a liberal bead of the liquid nails on both edges of the side panels and the tunnel wall. Place the rear panel in position, nailing it in place. A lot of the material will be squeezed out, but this can be removed with a putty knife and a damp cloth. Let the enclosure cure for a few hours, or better yet overnight, before adding the top and bottom pieces (**Photo 7**).

Once the top and bottom pieces are glued and nailed in place, fill in the nail holes with wood filler and sand the entire enclosure. The finish is your choice and can be either paint or a desired stain. For my project, I gave the enclosures two coats of primer, sanded them with fine paper, and then painted them a satin black.

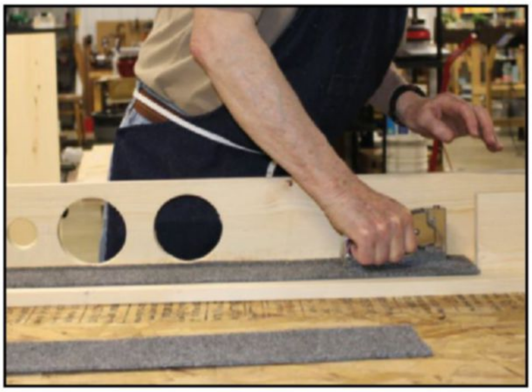

Photo 4: Add indoor/outdoor carpet to the enclosure sides.

Photo 5: Polyester stuffing is added to the lower cavity of the enclosure.

Crossover Network and Speaker Install

The crossover is a simple first-order type using a single coil and a capacitor. The woofer crossover coil has a value of 0.18µH and the tweeter 10µF. The crossover was built on 3″ × 4″ block of wood and I attached it to the rear wall just behind the lower woofer. I pre-wired it for the tweeter and woofer connections and then I attached with a 1.25″ drywall screw.

Next, wire the 8Ω woofers in parallel to make sure the positive and the negative connections are correct. Use a 1.5V flashlight battery to check for

Photo 6: Use corner clamps to hold the assembly while the walls are glued and nailed together.

Photo 7: The enclosures are shown assembled and ready for final finishing.

proper polarity at the speaker terminals. Both cones should move forward when the positive terminal of the battery is connected to the positive terminal of the terminal cup.

Listening Test

For the listening test, place the speakers about 8′ to 10′ apart and 12″ to 18″ from a back wall. Corner placement will enhance the bass response. Connect your receiver or amplifier, being careful to get the correct polarity on each tower speaker terminal. They will work with any amp from 10W and up. Play the type of music you like and adjust your bass and treble controls as you like. You will find the speakers provide great bass for the size with good imaging on vocals. If you want heart thumping bass, consider getting a subwoofer of 10″ or better to use with them, but for most applications, as general listening or as additions for your TV audio, they will be very satisfactory (**Photo 8**).

Photo 8: Here are my completed Dual Speaker Columns ready for use.

About the Author

Ken Bird has had a 40-year engineering and product management career focusing on consumer, industrial, and telecommunications electronics. Ken retired to his native Missouri and now pursues his interest in building speakers and tube amplifiers, along with model railroading, amateur radio (WOKLB), and digital photography.

Center-Channel Upmixer Delivers Stereo Nirvana

By
Peter H. Lehmann

The signal processing unit described in this article derives a center channel from standard stereo music recordings. Output of the unit is fed to a loudspeaker positioned laterally in the middle between left and right loudspeakers of a stereo system. Reproduction by this center loudspeaker supplements the quality and localization of the phantom center image.

Floyd Toole in Chapter 15 of his book *Sound Reproduction* describes the apparent spatial qualities of the center phantom image produced by standard (two-channel) stereo reproduction. This is for the listener situated at the ideal listening location (the sweet spot) and where the "featured artist" that he refers to has been recorded to be localized at center-front.

Toole wrote, "If one leans a little to the left or right, the featured artist flops into the left or right loudspeaker, and the soundstage distorts. When we sit straight up, the featured artist floats as a phantom image between the loudspeakers, often perceived to be a little too far back and with a sense of spaciousness that is different from the images in the left and right loudspeakers." [1]

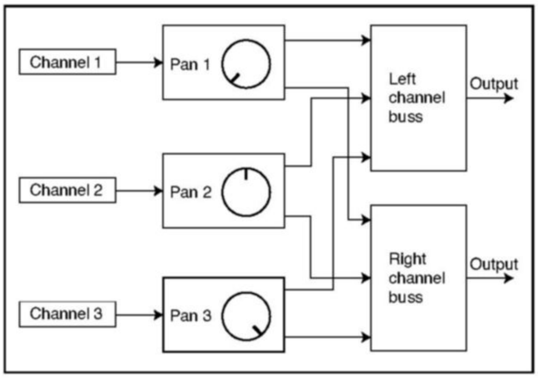

Figure 1: Derivation of the center signal by my upmixer assumes an amplitude-stereo recording. This type of recording determines lateral sound image localization by the settings of the knobs of panning potentiometers.

This quote agrees with my experience. The solution to this problem is to substitute a genuine center loudspeaker in place of the phantom center image. Thus, I set out to devise a signal-processing unit that would derive a center-channel signal from the two channels of standard stereo music recordings. The derived center-channel signal would then be fed to a loudspeaker situated laterally in the middle between the left and right loudspeakers.

Mixdown is the process of summing the content of multiple channels of recorded music into the two channels of a stereo recording. Thus, I'm calling my signal-processing unit an upmixer in the sense that it partially reverses a mixdown.

Amplitude-Stereo Recording

The deriving method of my upmixer assumes an amplitude-stereo recording. In this type of recording, the relative amplitude of the sources of the recording in the left and right channels determines where they are laterally localized upon the recording reproduced through stereo loudspeakers.

Figure 1 illustrates the amplitude-stereo recording technique. Channels 1, 2, and 3 are the isolated recordings of the performances of the different instruments of a composition. There is no "leakage" between the channels, that is, none of the sound recorded in a given channel is allowed to be recorded in any other channel.

The knob of pan pot 1 of Figure 1 set at 7 o'clock results in Channel 1 with full amplitude exclusively at input to the left channel buss. The knob of pan pot 2 set at 12 o'clock causes Channel 2 to be at input to both left and right channel busses in phase and with equal amplitude. Given that the knob of pan pot 3 is set at 5 o'clock, this causes all of the amplitude of Channel 3 to be only at input to the right channel buss.

Where the hypothetical recording of Figure 1 is reproduced by left and right channel stereo loudspeakers, the sound of Channels 1, 2, and 3 is localized respectively as coming from the left channel loudspeaker, in the middle laterally between the loudspeakers, and coming from the right channel loudspeaker. In this recording scenario, the task that I set for myself was to recover the sound of Channel 2 with as little cross talk from Channels 1 and 3 as possible.

Deriving the Center Signal

At the heart of my upmixer is the circuit schematic diagram, which is shown in **Figure 2**. I call this circuit a Lesser Amplitude Follower (LAF). The reason for this name will become apparent from my explanation of the operation of the circuit.

The processing of the circuit of Figure 2 is a modification of an AND gate effected by a circuit of diodes and resistors [2]. In the LAF circuit, there are two linear inverting half-wave rectifiers with positive output including, respectively, op-amps U1 and U3. The output of the rectifier including op-amp U1 is connected to the input of a buffering stage including op-amp U2. Likewise, the output of the second rectifier (including U3) is connected to a buffering stage (including U4).

The output stage of the LAF of Figure 2 includes the cathodes of diodes D5 and D6 connected respectively to the output terminal of the first and second buffering stages (including U2 and U4). The anodes of these diodes connect to one termination of resistor R5 and the opposite lead of R5 connects to the positive power supply voltage, +15V.

The two input terminals of the LAF shown at Figure 2 are the input terminals of the half-wave rectifiers. For the simulation of Figure 2, there are two sine wave sources of equal amplitude and frequency each connected to one of the input terminals of the LAF. The output terminal of the LAF is the common connection of the anodes of diodes D5 and D6.

In Figure 2, given diode D5 or D6 forward biased, then for that diode, the voltage taken at the anode

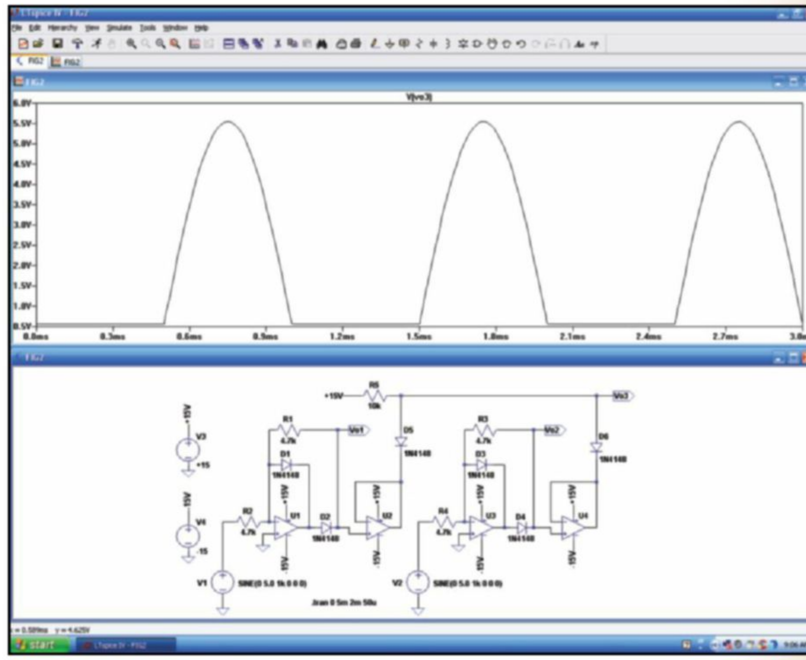

Figure 2a: In a simulation with the core circuit of my upmixer, which I have named a Lesser Amplitude Follower (LAF), the two voltages at input to the circuit are from sine wave sources V1 and V2 of equal amplitude and frequency. This figure shows the first part of the simulation with the sources in-phase.

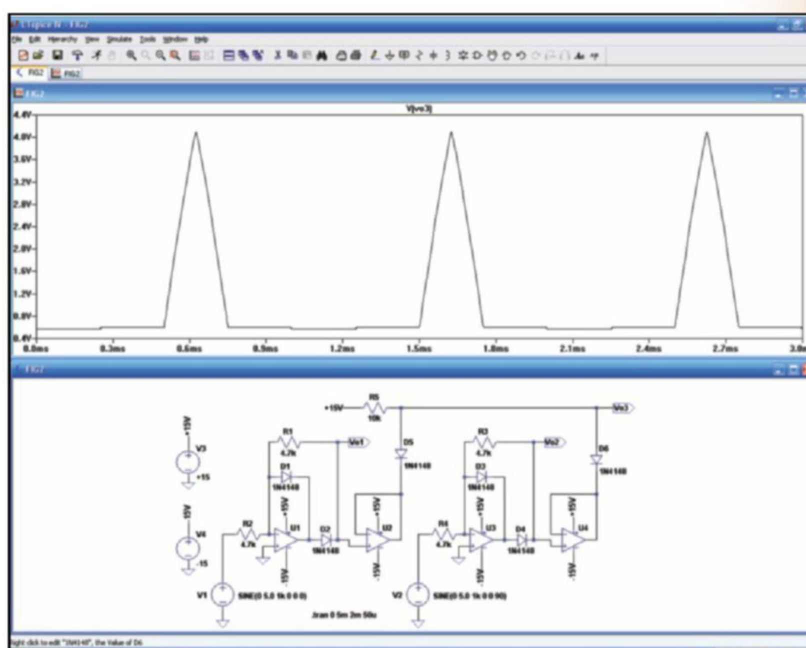

Figure 2b: In the second part of the simulation, relative phase of the two sine wave sources V1 and V2 is 90°. The change of relative phase results in a reduction of the RMS voltage of the output voltage Vout by 6dB.

of the diode must be equal to the voltage taken at the cathode of the diode plus a diode drop. Thus if the voltages taken at the output terminal of the buffering stages are identical, then the output voltage of the LAF equals that of the inverted half-wave of the voltages at input plus a diode drop.

However, if the instantaneous negative half-wave voltage of the two input AC signals are unequal at any time, then this causes one of diodes D5 and D6 to be reverse biased while the anode of the other diode that is forward biased resides at the absolute value lesser input voltage plus a diode drop. The

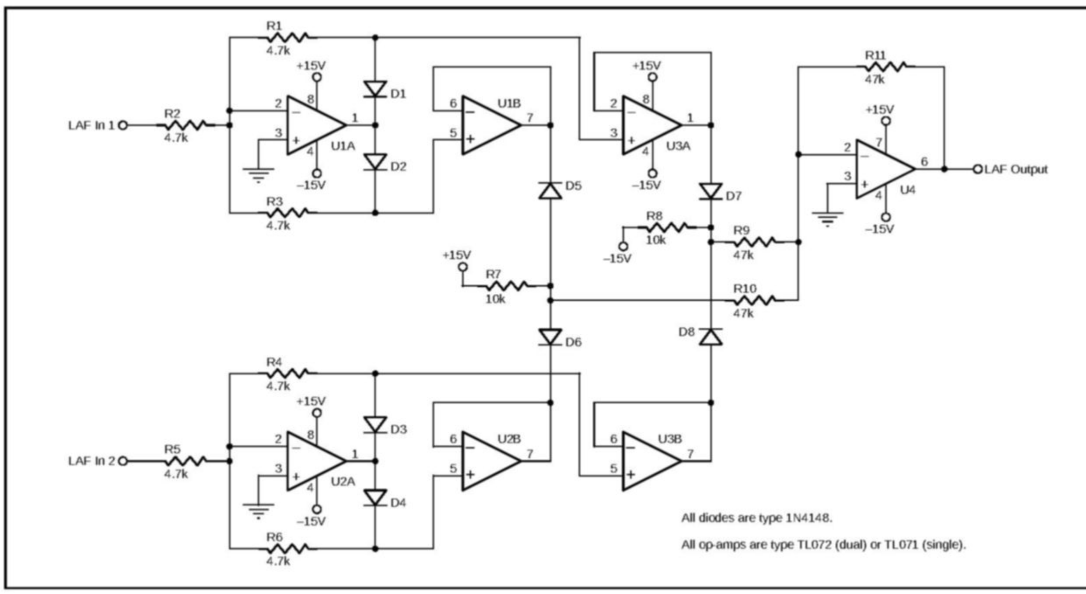

Figure 3: A practical Lesser Amplitude Follower (LAF) must restore the full-wave nature of audio signals at input. The practical LAF shown here includes a negative-output as well as positive-output LAF section. The outputs of the two sections are summed resulting in a full-wave output voltage taken at pin 6 of op-amp U4.

voltage of lesser absolute value becomes the output voltage of the LAF.

In the simulations of Figure 2a and Figure 2b, input sine wave sources V1 and V2 are respectively in-phase and 90° out of phase. The two sine waves generated are of equal amplitude and frequency. The two waves in phase and 90° out of phase represent, respectively, the center signal and other signals (cross talk) with a random phase relationship the level of which we wish to attenuate. The waveform shown in Figure 2a is the inverted half-wave rectification of the identical input sine waves while that of Figure 2b shows the output voltage of the LAF following the lesser amplitude of the inverted half-wave input of the sine waves 90° out of phase. Sine wave sources V1 and V2 initially in phase, and then set to be 90° out of phase, results in

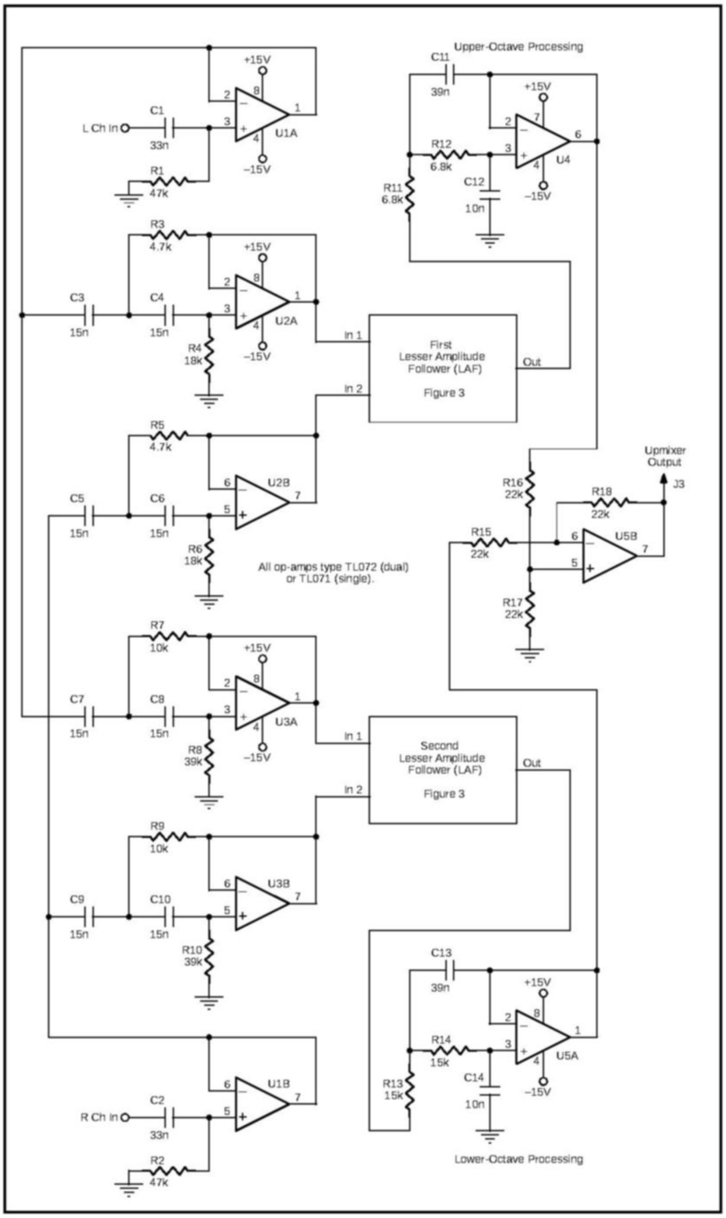

Figure 4: My upmixer provides high-pass filtering of the stereo signals at input to LAF processing and low-pass filtering of the output of the LAF. This is done in upper and lower octave processing sections. Two processing sections are required to produce an adequate output level of the upmixer.

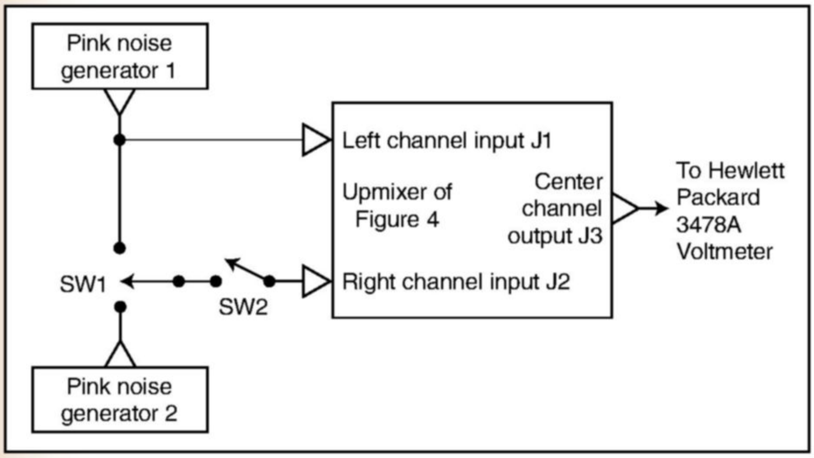

Figure 5: The block diagram shows bench testing of the upmixer circuit of Figure 4 with two independent pink noise generators as signal sources at input. A music signal panned to center is simulated given switch SW2 closed, and switch SW1 in the up position. Two non-identical music signals with a random phase relationship oppositely panned hard left or right is simulated by switch SW1 in the down position and switch SW2 remains closed.

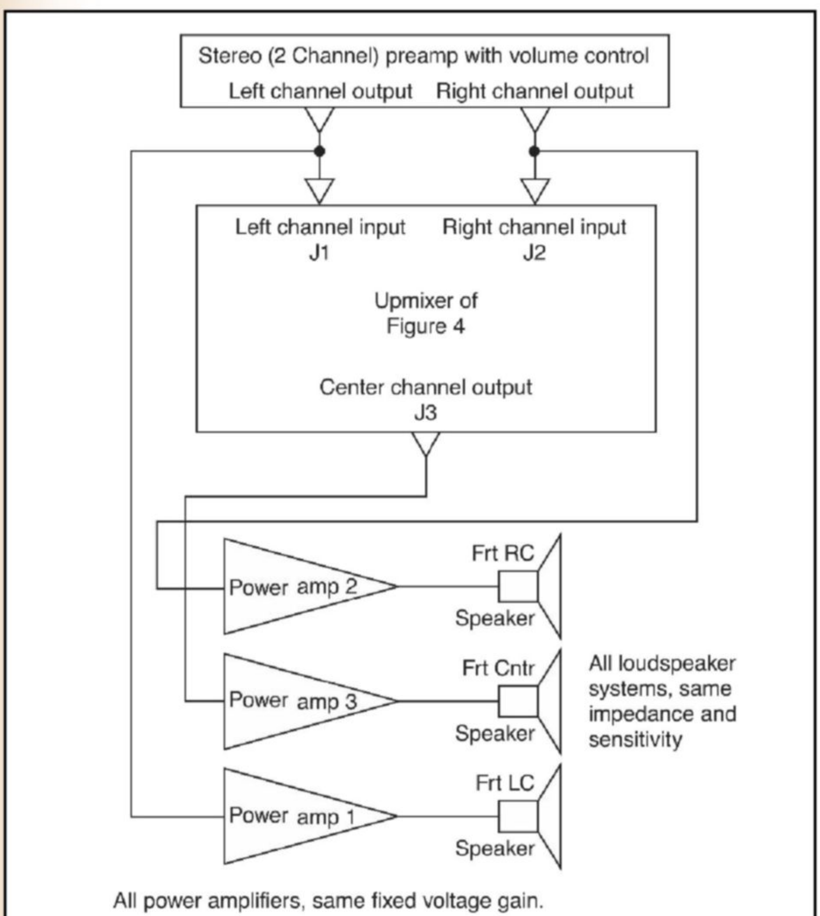

Figure 6: So the level of the derived center channel correctly matches that of the left and right channels at any listening level, the preamplifier must have a volume control. Equal voltage gain of the three power amplifiers and equal impedance and sensitivity of the three loudspeakers insures that the relative output level of the center loudspeaker relative to that of the stereo loudspeakers is correct.

a reduction of the RMS output voltage of the LAF by 6dB.

Figure 3 shows the schematic diagram of a practical LAF providing for full-wave rectification of the signals applied to input terminals LAF IN1 and LAF IN2. The two input terminals are inputs to signal polarity separators including op-amps, respectively U1A and U2A. The buffered positive outputs of the two separators connect to the cathode of respectively diodes D5 and D6, and the anodes of these diodes connect through resistor R7 to the positive power supply voltage (+15V). In a like manner, the buffered negative outputs of the separators connect respectively to the anodes of diodes D7 and D8. The cathodes of D7 and D8 connect through resistor R8 to the negative power supply voltage, -15V. The inverting adder including op-amp U4 recombines the half-wave outputs of the two LAFs by the connection of resistors R9 and R10 from the inverting input terminal of op-amp U4 to the junction of diodes D7 and D8, and the junction of diodes D5 and D6.

Applying the illustration of mixdown using the amplitude-stereo technique of Figure 1 to processing by the LAF of Figure 3, the output of the left and right channel busses are connected to respectively the In1 and In2 inputs of the LAF. The mixdown of Channel 2 results in the signal of that channel applied equally to the inputs of the LAF; thus the LAF passes the signal of Channel 2 with 0dB attenuation. The mixdown of Figure 1 results in Channels 1 and 3 hard panned to, respectively, the left and right channels.

The signal content of Channels 1 and 3 are unrelated and thus phase of the signal of Channel 1 in the output of the left buss is random with respect to that of the signal of Channel 2 in the output of the right buss. Thus, the summation of Channel 1 and Channel 3 by LAF processing attenuates and distorts that summation. As the content of Channel 2 has been made the center signal by the panning scenario of Figure 1, processing by the LAF of Figure 3 causes the level of the center signal to be higher than that of the summation of the other two channels 1 and 3 panned hard left and hard right.

Increasing Attenuation of Cross Talk

Those signals at the pair of inputs to the LAF processing of Figure 3 with a random phase relationship, that is, cross talk, become distorted by that processing. That distortion is primarily third-harmonic distortion. The level of the third-harmonic distortion can be attenuated by high-pass filtering the pair of input signals to the LAF of Figure 3, and low-pass filtering the output of the LAF, where high- and low-pass cut-off frequencies of the filters are one octave apart. Attenuating the third-harmonic distortion makes it less audible and also increases attenuation of the cross talk.

Figure 4 shows the schematic diagram of my upmixer and details the integration of high- and low-pass filters with a LAF (Figure 3) in upper and lower octave processing sections. Two octaves of processing are needed to produce an adequate output level of the upmixer.

At the top of the Figure 4 is the upper octave processing section of my upmixer with a pass band of 800Hz to 1600Hz. The lower octave processing section with a pass band of 400Hz to 800Hz is shown at the bottom of Figure 4.

Both sections of my upmixer of Figure 4 are configured in identical fashion with the inputs of two high-pass filters, respectively connected to buffered left and right channel line level inputs. The outputs of the high-pass filters of each section connect to respectively the two inputs of the LAF of the section. The output of the LAF of each section connects to the input of a low-pass filter. The outputs of the low-pass filters of each of the two processing sections are at input to op-amp U5B configured as a subtractor. Due to phase shift introduced by the filters of the processing sections, the subtractor sums the output voltages of the sections.

All of the filters of Figure 4 are second-order Sallen-Key following the equations given by Siegfried Linkwitz on his website [3]. However, rather than Q equal to 0.5 as advised, I have made Q of the filters equal to 1. The high-pass filters (including op-amps U2 A and B) and low-pass filters (including op-amp U4) of the upper octave processing section have a cut-off frequency equal to 1.41 times 800Hz, that is, 1131Hz. The high-pass filters (including op-amps U3 A and B) and low-pass filters (including op-amp U5A) of the lower octave processing section have a cut-off frequency equal to 1.41 times 400Hz, that is, 566Hz.

Bench Testing My Upmixer

The block diagram of **Figure 5** shows the testing that I did of my upmixer of Figure 4 to attempt to determine how successful the processor is at attenuating cross talk relative to the level of passing the center signal given standard stereo inputs.

The two generators of Figure 5 were independent, stand-alone generators of pink noise at the same output voltage level. The first generator is constantly connected to one input of the upmixer while the two switches give the options of no signal, or the pink noise of the first or second generator at input to the second input of the upmixer. Initially the pink noise of generator 1 is connected to both Jacks J1 and J2 of the upmixer by means of switch SW1 in the up position and switch SW2 closed. Where switch SW1 is subsequently switched to the down position, then the output RMS voltage taken at Jack J3 of the upmixer decreases by 9dB. Relative to the output of generator 1 initially applied to both inputs of the upmixer, if switch SW2 is opened, then the RMS voltage taken at output Jack J3 decreases by 28dB.

My conclusion from the above testing is that my upmixer, relative to the level of passing of the center signal, attenuates cross talk by a minimum of 9dB. Given that music is impulsive, I might conclude that on average the relative attenuation of cross talk is greater than 9dB and less than 28dB.

About the Author

Peter Lehmann received some basic instruction in electronics while in high school. However, he did not return to electronics until several years after graduating with a liberal arts degree in 1974. For several decades now, electronics has been a strong avocational interest of his, focusing on audio and analog circuitry. Nevertheless, he says he does have several books on his shelf dealing with digital electronics. Peter started writing for *audioXpress* in 2018. His "Reflective Way Two-Way System" article series was published in November and December 2018.

Stereo Reproduction and the Upmixer

The type of stereo system shown in **Figure 6** allows for most easily and effectively converting two-channel stereo to three-channel. At the top is a preamplifier with a volume control required to allow the output level of the new center channel to be adjusted simultaneously with that of the left and right (stereo) channels. Preferably the three power amplifiers of the system should have the same voltage gain. Similarly, the three loudspeakers should have the same impedance and sensitivity.

Listening Test

In my opinion, the addition of center-front reproduction using the processing of my upmixer results in a distinct improvement of the reproduction of standard two-channel stereo. Perhaps some people would consider the improvement a bit subtle and only worthwhile to those more preoccupied with music listening than most. But if not too costly, I think that most people would recognize the improvement and prefer to have the additional third channel.

An advantage of the modification of standard two-channel stereo reproduction that I have described here might be that it doesn't involve replacing the phantom center image, but rather supplements it. That is, my upmixer operates over the frequency range of 400Hz to 1600Hz (i.e., two octaves), while the range of music is about 8 octaves.

Also, my modification involves no alteration of the channel signals fed to the left and right loudspeakers. Floyd Toole, in Chapter 8 of his book *Sound Reproduction,* doesn't totally discount the phantom center image in favor of an actual center-front loudspeaker.

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[1] F. Toole, Sound Reproduction, Burlington: Focal Press, 2008, p. 277.

[2] T. Kuphaldt, Lessons in Electric Circuits, Volume III- Semiconductors, Section 3.10, pp.132, 133. www.allaboutcircuits.com /assets/pdf/semiconductors.pdf

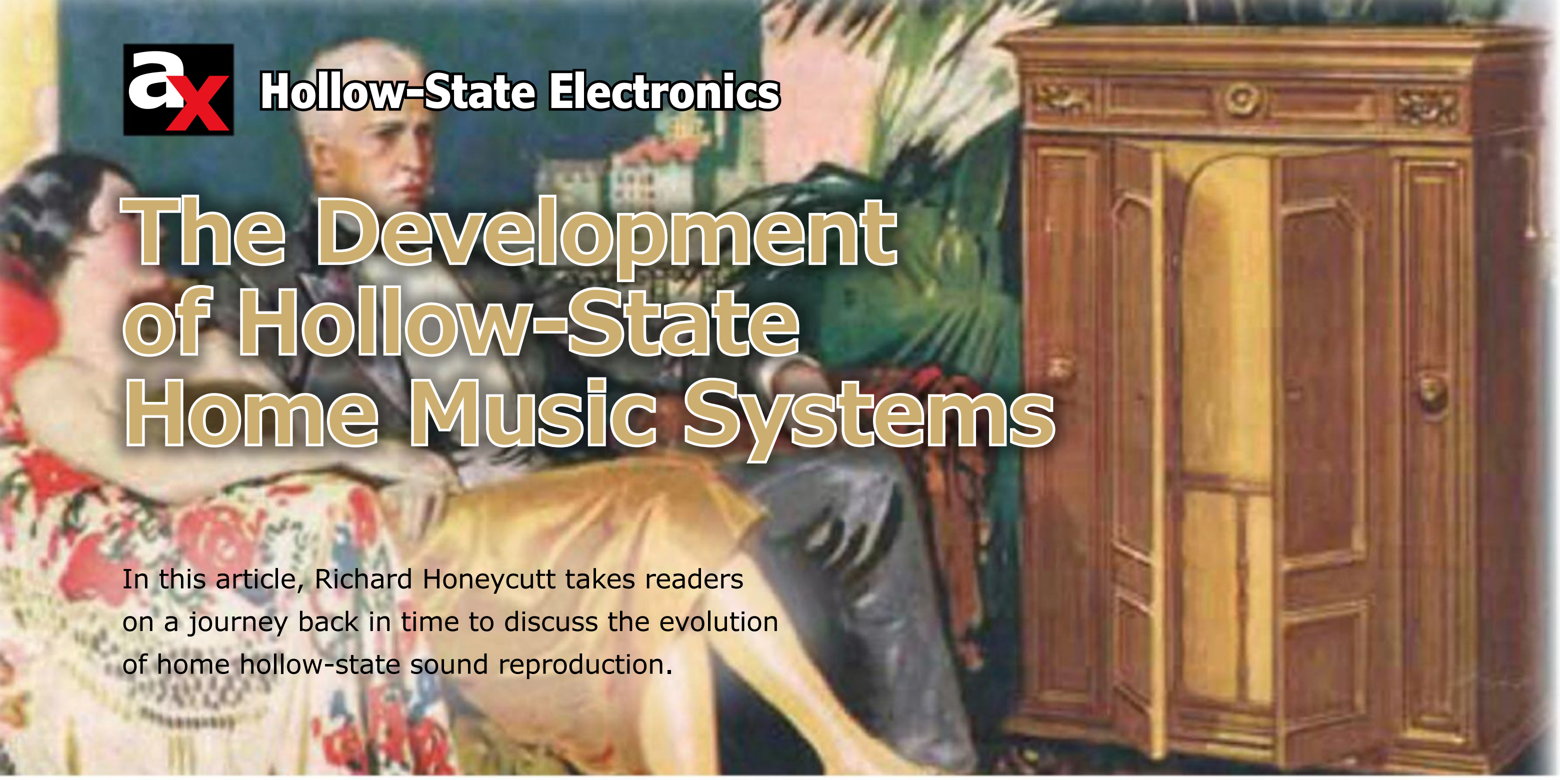
[3] S. Linkwitz, "Active Filters," Linkwitz Labs, www.linkwitzlab.com/filters.htm

Resource

P. Lehmann, "Reflective-Way, Two-Way System," *audioXpress*, November/December 2018.

The Development of Hollow-State Home Music Systems

In this article, Richard Honeycutt takes readers on a journey back in time to discuss the evolution of home hollow-state sound reproduction.



By
Richard Honeycutt

Photo 1: The Orthophonic Victrola was the best of the acoustical phonographs.

In many places today, it is common to see people listening to music on a portable radio, a cell phone, or a laptop computer. With the sound quality available from the best automotive audio systems today, it is easy to enjoy listening in car. As a lifelong musician, audiophile, and former broadcast radio and recording engineer, I find it baffling that anyone would choose to listen to music on a cell phone or laptop, unless excellent earbuds, earphones, or computer speakers are available. But as a fellow musician loves to point out, "there's no accounting for taste or lack thereof."

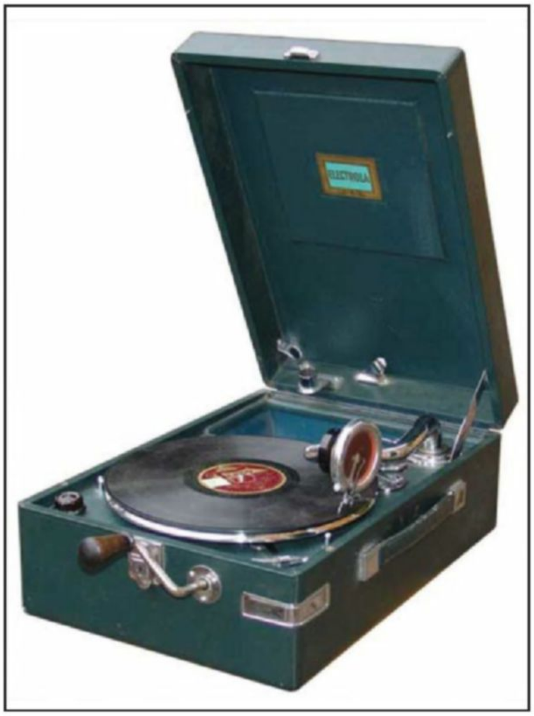


Photo 2: The Electrola electric phonograph introduced electronic amplification to home phonographs, and led to eliminating the need to "crank up" the spring motor before playing a record.

The Evolution of Sound Quality

Prior to the 20th century, music in the home was either homemade, or limited to wealthy homes whose owners could afford to hire musicians. Beginning in 1889, Edison cylinders were available to allow people to listen to prerecorded songs at home. Emile Berliner's introduction of the phonograph disc in the 1890s started the record industry as we knew it for most of the 20th century.

The first phonograph discs were recorded and played acoustically. In the early 1920s Western Electric introduced electrical recording, creating better records and allowing—for example—singers without very loud voices to be recorded. (It is not commonly known that one of Al Jolson's greatest strengths was not the quality of his voice, but his ability to sing loudly, thus producing records with a good—for the time—signal-to-noise ratio.)

The best of the acoustical phonographs was the Orthophonic Victrola (**Photo 1**), designed by Bell Labs and produced by the Victor Co. This device incorporated a folded horn to better match the acoustical impedance of the phonograph needle to the air of the room, producing louder volume and more bass sound. Upon hearing a recording of a band played upon this device in 1925, John Philip

Sousa commented, "[Gentlemen], that is a band. This is the first time I have ever heard music with any soul to it produced by a mechanical talking machine" ... "The new instrument is a feat of mathematics and physics. It is not the result of innumerable experiments, but was worked out on paper in advance of being built in the laboratory ... The new machine has a range of from 100 to 5,000 [cycles], or five and a half octaves ... The 'phonograph tone' is eliminated by the new recording and reproducing process." [1] The Orthophonic was priced from $95 to $300, depending upon the aesthetics of the cabinet. This amounted to five weeks' to a year's pay for a clerical worker, and the most expensive units cost more than a new Model T Ford.

Also in 1925, RCA initiated the era of electrical record players with the release of the Electrola (**Photo 2**), which had an electric motor. Prices ranged from $650 upward. As smaller "record players" with electric motors, hollow-state amplifiers, and loudspeakers became more commonly available, prices dropped, and soon they replaced acoustical players. **Figure 1** shows the schematic of a 1946 Electrola amplifier, which provided 12.5W output from a push-pull pair of 6F6s into a 2Ω electrodynamic speaker. As with virtually all equipment using electrodynamic speakers, the field coil doubled as a power-supply filter inductor.

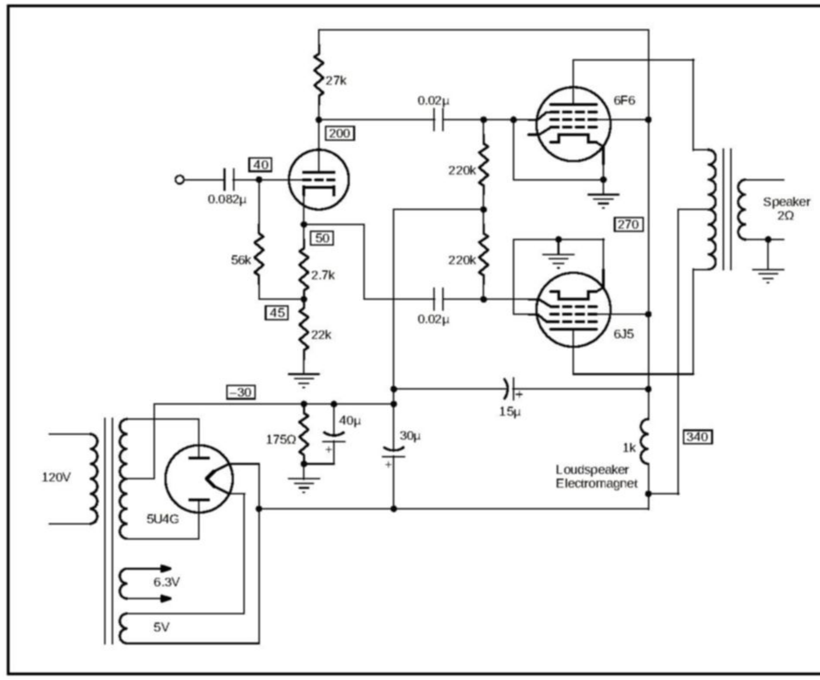


Figure 1: The 1946 version of the Electrola used a similar amplifier design to the first models.

Early Turntables

Early record players had a real problem with "turntable rumble"—noise conducted into the pickup from the motor bearings. With mechanical technology being incapable of solving the problem of rumble, engineers began to limit low-frequency response instead. With smaller record players having speakers that were 6" and under in diameter, this limitation occurred naturally. But often smaller coupling capacitors in the amplifiers were also used to limit low-frequency response.

Record players were by no means common in the 1920s. Radio receivers not only provided access to music in the home, but they also offered newscasts, political addresses, and weather forecasts. The Atwater-Kent Model 35 radio (**Photo 3**) of the mid-1920s used a single-ended triode stage to drive a headphone output, as you can see from the schematic in **Figure 2**. The unit shown in Photo 3 has a speaker that is essentially a headphone with a cone attached. That speaker has an impedance of 6.5kΩ at 500Hz, typical of magnetic headphones. Even though the speaker diameter was about 8", the frequency response did not extend much below 200Hz.

Home Radios

By 1937, home radios had evolved and the sound quality had improved in some ways. Whereas the Atwater-Kent Model 35 used a Tuned Radio Frequency (TRF) design, the 1937 Philco model 620B (**Photo 4**) used a superheterodyne ("superhet) circuit, which has a tunable LC "tank" circuit in each of the three RF stages. When these were properly adjusted, they worked very well, producing good sound. But since it is very difficult to keep all the RF stages adjusted so that they tune to the exact same frequency band, the total bandwidth of a TRF radio was often about 20kHz. With broadcast stations having carrier frequencies spaced 10kHz apart,

Photo 3: This Atwater-Kent Model 35 was the home music system for many people in the mid-1920s.

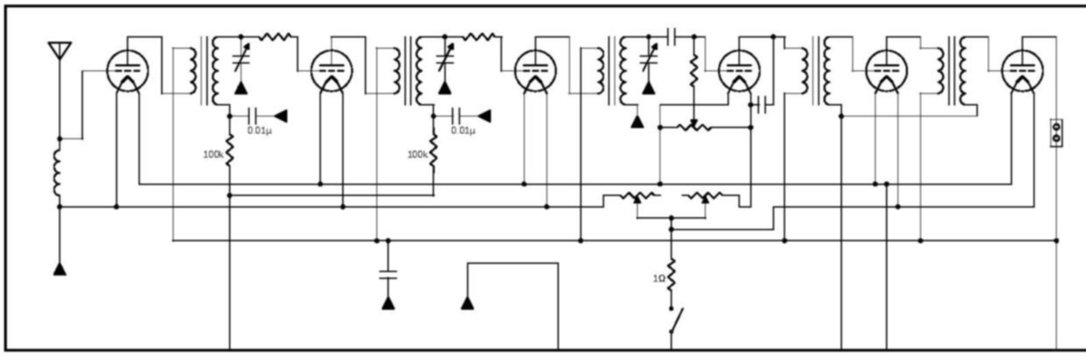


Figure 2: As this schematic shows, the Atwater-Kent Model 35 radio used six triodes.

Photo 4: This 1937 Philco Model 620B radio boasted an 8″ speaker and had really good sound for its time.

TRF receivers were often plagued with interference from adjacent stations.

The operation of a superhet radio is different: there may or may not be a bandlimiting filter in the RF stage—if there is an RF stage. Many AM radios feed the antenna signal directly into the converter stage with no RF amplification. But most of the bandwidth restriction occurs in the intermediate frequency (IF) stage(s), and that pretty much enforces a 5kHz high-frequency limit. In so doing, the superhet provides improved selectivity (ability to reject interference from adjacent stations). So the TRF radios excelled at high-frequency reproduction, but the superhets offered better selectivity. In both cases, low-frequency response was limited by the speakers usually being less than 8″ in diameter, and mounted in an open-back cabinet.

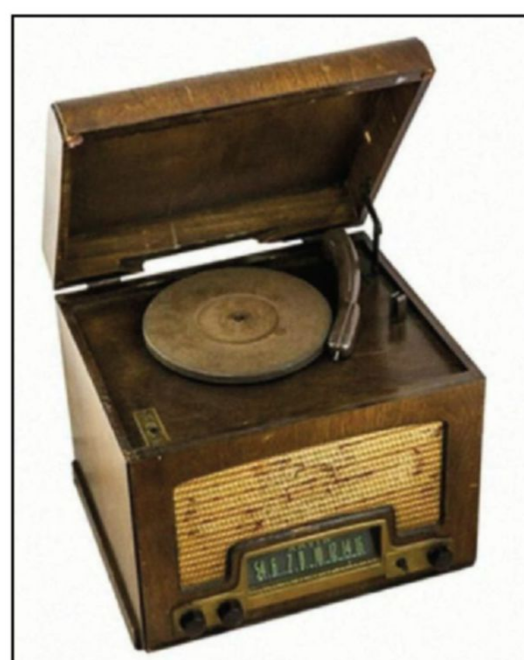


Photo 5: This 1940s-vintage Arvin radio-record player was convenient, but its relatively small speaker limited its fidelity.

AM Radio Signals

It has often been stated that the bandwidth of an AM radio signal is limited to 5kHz. The reason is that, say, a 1000kHz carrier modulated by a 5kHz signal will produce a transmitter output containing three frequencies: the carrier, the sum, and the difference. So for the 1000kHz carrier and 5kHz modulating signal, the bandwidth would cover 10kHz—from 995kHz to 1005kHz. However, the Federal Communications Commission (FCC) almost never assigns adjacent frequencies to stations in geographical proximity, so some overlap in the actual transmitted bandwidth is permissible, especially since the higher-frequency modulating signals are weaker. Therefore, many AM station engineers only limit the transmitted bandwidth to 20kHz. AM receivers, however, (especially modern ones) often limit the bandwidth to 3.5kHz or less.

The 1937 Philco Model 620B table radio shown in Photo 4 actually produced much better sound than the AM section of most of today's stereo or surround receivers, because most modern designers assume that AM will only be used to listen to sportscasts and other speech-only programs, and therefore make no effort to optimize AM sound quality. By contrast, the owner of the Philco 620B might well listen to broadcasts of the Metropolitan Opera on Saturday afternoons, and concert broadcasts from Radio Free Europe, the British Broadcasting Corp. (BBC), Deutsche Welle, or Radio Moscow (all on the short-wave bands) during evenings. This radio contained an RF stage, a detector/oscillator (would now be called a "converter"), two IF stages, an audio preamp stage and a single-ended audio output stage feeding a maximum of 4.8W to an electrodynamic speaker.

By this time, many experimenters had begun buying turntables and building "phono oscillators" (small low-power AM transmitters) so they could listen to records on their radios. Table radios such as the Philco 620B and its floor-standing cousins provided good sound for the time. In fact, in the early 1950s, it was not uncommon for TVs to have phono inputs. However, with their single-ended miniature pentode audio output tubes driving very small transformers feeding 5" or smaller speakers, the sound did not match that of the late-1930s radios.

Another alternative for playing records in the 1940s was the combination radio-record player such as the Arvin model shown in **Photo 5**, which was typical of the five-tube radios of the day.

The Start of the Hi-Fi Industry

Beginning in the late 1940s, serious home-music listeners became a large enough group that they birthed the high-fidelity "hi-fi" industry. Although brands such as Fisher, H. H. Scott, and Marantz (and later, Asian brands as well) are better known today, Pilot Electric Mfg. Co. was a major force in the hi-fi industry from 1952 until about 1970.

Photo 6 shows a Pilot Model AA-903 mono hi-fi amplifier that accepted inputs from phono, tape head, auxiliary, tuner, or tape-recorder amplifier sources. Its circuit design is fairly common for hollow-state hi-fi amps of the day.

The sound of Peter Jensen's 1915 outdoor sound system that covered a crowd of 50,000 people in Golden Gate Park was reported as being "completely natural." Also, listeners at the Bell Labs "live-versus recorded" demo in the 1930s could not distinguish the live sound from the reproduced sound of an orchestra. From the 1930s until recently, the frequency response of a telephone was only 200 Hz to 3,000Hz.

Noting the great advances in sound-system technology that we have enjoyed since then, we might be tempted to suppose that humans' critical listening abilities have evolved during the past century. Then we see people listening to music on cell phones, or our car is passed by a "boom car" emitting more rattles and buzzes than music, and we think, "Well, not so much!" However I hope this trip in the "Wayback Machine" has opened interesting chapters in the history of home hollow-state sound reproduction for you. Shall we talk again after another century?

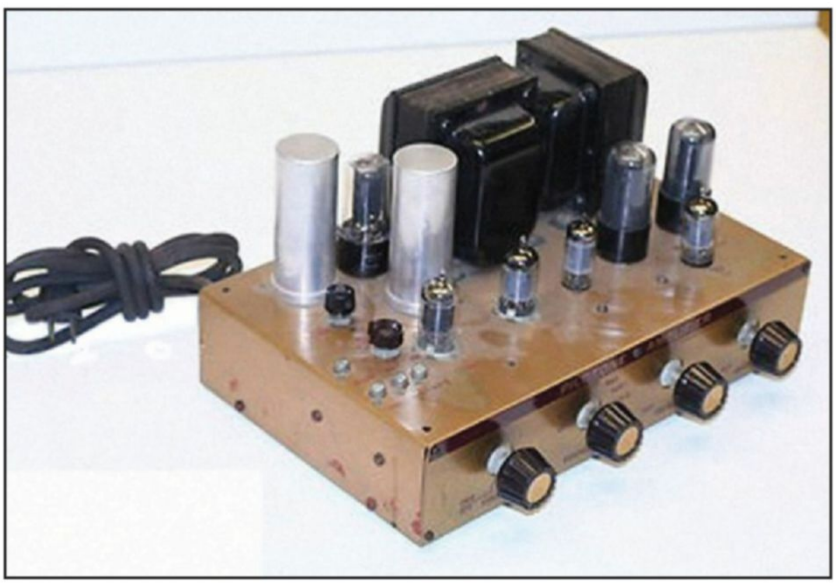


Photo 6: The Pilot Model AA-903 mono hi-fi amplifier was used as a control-room monitor amplifier in many broadcast studios, as well as in many homes.

Reference

[1] "New Music Machine Thrills all Hearers at First Test Here," *The New York Times*, (October 7, 1925). Archived May 10, 2013 at the Wayback Machine front page.

About the Author

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

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