

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Focus on Acoustics
Electroacoustic Enhancement
and Active Acoustics
By J. Martins

Simulation Techniques
SonicEdge Ultrasound Speaker
Modeling and Application
By René Christensen

Automotive Audio
Noise Cancellation in Automotive Audio
What's the Destination?
By Philipp Paul Klose

Show Report
AXPONA 2023
By Oliver A. Masciarotte

Speakers
Designing a True Midrange Driver
By Jørn Rune Kviserud

Speakers
Bass Reflex Performance
Envelope — Part 2
By Gary Gesellchen

You Can DIY!
Instrumentation Amplifier-Based
Preamplifier Line Stage
By Bruce Gillingham

Hollow-State Electronics
Visiting the Western Electric
Vacuum Tube Factory
By Richard Honeycutt



HD 400 PRO and dearVR MIX

Perfect partners for
headphone mixing



sennheiser.com/hd-400-pro
dear-reality.com/dearvr-mix

DEAR REALITY

 SENNHEISER



MANUFACTURING / DESIGNING TRANSFORMERS & ENCLOSURES SINCE 1917





August 2023 ISSN 1548-0628
www.audioxpress.com

audioXpress (US ISSN 1548-0628) is published monthly, at \$50 per year for the US, at \$65 per year for Canada, and at \$75 per year Foreign/ROW, by KCK Media Corp., P.O. Box 417, Chase City, VA 23924, US Periodical Postage paid at Chase City, VA, and additional offices.

Head Office:
KCK Media Corp.
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
Fax: 888-980-1303

Subscription Management:
audioXpress
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
E-mail: custserv@audioxpress.com
Internet: www.audioxpress.com

Postmaster: Send address changes to:
audioXpress
P.O. Box 417
Chase City, VA 23924, US

Advertising:
Strategic Media Marketing, Inc.
2 Main Street
Gloucester, MA 01930, US
Phone: 978-281-7708
Fax: 978-281-7706
E-mail: audioxpress@smmarketing.us

Advertising rates and terms available on request.

Editorial Inquiries:
Send editorial correspondence, suggestions, and contributed articles to:
audioXpress, Editorial Department
P.O. Box 417
Chase City, VA 23924, US
E-mail: editor@audioxpress.com

Legal Notice:
Each design published in audioXpress is the intellectual property of its author and is offered to readers for their personal use only.
Any commercial use of such ideas or designs without prior written permission is an infringement of the copyright protection of the work of each author.

© KCK Media Corp. 2023
Printed in the US



The Perception of Change

As I always like to say: I know that I know nothing, but I’m always learning. (Plato would agree if he had said it)

Unfortunately, in the life of a magazine editor I have long learned that there is never enough time to just study and learn. I strongly admire people who have devoted their whole career to study, research, and publish what they learned (many times combined with teaching). That’s what I always like about attending conferences, where here and there you come across people and work that shake the foundations of our knowledge and understanding.

That was the case recently, when I—together with our technical editor Jan Didden, and a good number of *audioXpress* authors—attended one of the opening sessions of the 154th Audio Engineering Society (AES) International Convention, held at Aalto University in Espoo, Helsinki, Finland, from May 13-15.

The panel session, titled “Towards an Objective Understanding of High-End Audio”—expanded in the actual event to Hans van Maanen, Milind N. Kunchur, Joshua Reiss, Menno van der Veen, Mike Turner, and Jaime Angus-Whiteoak. The first intervention by distinguished professor Milind N. Kunchur, from the University of South Carolina, basically set the tone when he attempted to summarize in the allotted time his recent work, “The Human Auditory System And Audio” (©Elsevier 2023) accepted manuscript. Preprint: <https://dx.doi.org/10.2139/ssrn.4437822>

Each sentence in his brief introduction exploring his extensive work in psychophysics, auditory neurophysiology, and high-fidelity sound reproduction was basically enough to merit a session in itself. He stated up front that evaluation of audio performance needs to take into account “the complexities and intricacies of the human auditory system” and “be based on a better biological foundation.” Statements about the human auditory system, phase, frequency, and time signal analysis in the time domain, or even brain centers and memory (backed by the extensive bibliography quoted in his research) all have profound implications on the audio industry as we characterize it today. In the audience, distinguished AES Fellows and members of the audio community could be seen jumping on their chairs with excitement!

As described in the conclusions of this work, Prof. Kunchur explained how musical training causes significant cortical changes (shifts the emphasis from the right to the left hemisphere of the brain) and how an extended multiple-pass (EMP) listening protocol facilitates forming a consolidated opinion in durable long-term memory. He further quoted research that found that our brain needs at least 15 seconds to replace the short-term memory during a listening test as the reason why standard blind tests employing short-segment comparison (SSC) of back-to-back brief stimuli often fail (Thus, judging sound quality takes time!)

In his “Implications for Audio” conclusions, he also explains why very high and even ultrasound frequencies may be audible “at high levels or as part of a complex tone due to mechanisms such as heterodyning,” (nonlinear mixing) and bone conduction.

In a remarkable Richard Heyser Memorial Lecture also focused on the human auditory system, titled “The Ear is not a Fourier Transformer,” Jamie Angus-Whiteoak, emeritus professor at the University of Salford, reinforced the message about the need to better understand “the unique ways the human auditory system allows us to hear the incredible complexity of the audio signal and how that might affect what we do in the future to improve audio.”

I had this extensive research in mind while working on this issue’s article about active acoustics and seeing how much academic and scientific work has contributed to the significant progress in this important technology. Among many other sources, I had to go through the extensive and remarkable work of Josh H. McDermotta and the Department of Brain and Cognitive Sciences, at the Massachusetts Institute of Technology, and the (related) extensive work of David Griesinger in this field.

There’s so much we could all benefit if we evolve our new designs and product developments always based on the latest scientific research.

My appreciation and thanks to Luigi Agostini, E-coustic Systems project division manager for the EMEA market, for the support in the meeting and interview with Steve Barbar, and coordinating our visit to the amazing DiMenna Center for Classical Music, featured in this issue. Enjoy!

J. Martins
Editor-in-Chief

The Team

Publisher:	KC Prescott	Associate Editor:	Shannon Becker
Controller:	Chuck Fellows	Graphics:	Tori Zienka
Editor-in-Chief:	João Martins	Advertising Coordinator:	Heather Childrey
Technical Editor:	Jan Didden		

Regular Contributors: Bruce Brown, Bill Christie, Bob Cordell, Joseph D’Appolito, Dimitri Danyuk, Vance Dickason, Jan Didden, Scott Dorsey, Gary Galo, Chuck Hansen, Richard Honeycutt, Charlie Hughes, Mike Klasco, David Logvin, Ward Maas, Oliver Masciarotte, Nelson Pass, Christopher Paul, Bill Reeve, Michael Steffes, Nora Wong, Stuart Yaniger, Ethan Winer

OUR NETWORK



SUPPORTING COMPANIES

Acoustic Geometry	33	Parts Express International, Inc.	68
Cine Gear Expo Atlanta 2023	47	PCB Piezotronics, Inc.	43
COMSOL, Inc.	9	Redco Audio	43
Hammond Manufacturing, Ltd.	3	SB Acoustics	41
HEAD acoustics, Inc.	13	Scantek, Inc.	39
KAB Electro-Acoustics	49	Sennheiser	2
Lenbrook International	17	Triad Magnetics	67
Menlo Scientific, Ltd.	63		

NOT A SUPPORTING COMPANY YET?

Contact Peter Wostrel (audioXpress@smmarketing.us, Phone: 978-281-7708, Fax: 978-281-7706) to reserve your own ad space in the next issue of the magazine.

The Audio Voice

Sign up now for our **FREE** weekly e-newsletter



Weekly Newsletter for the Audio Industry, Professional Designers and Audio Enthusiasts
Product Design | Audio Electronics | Acoustics | DIY | Innovations in Audio



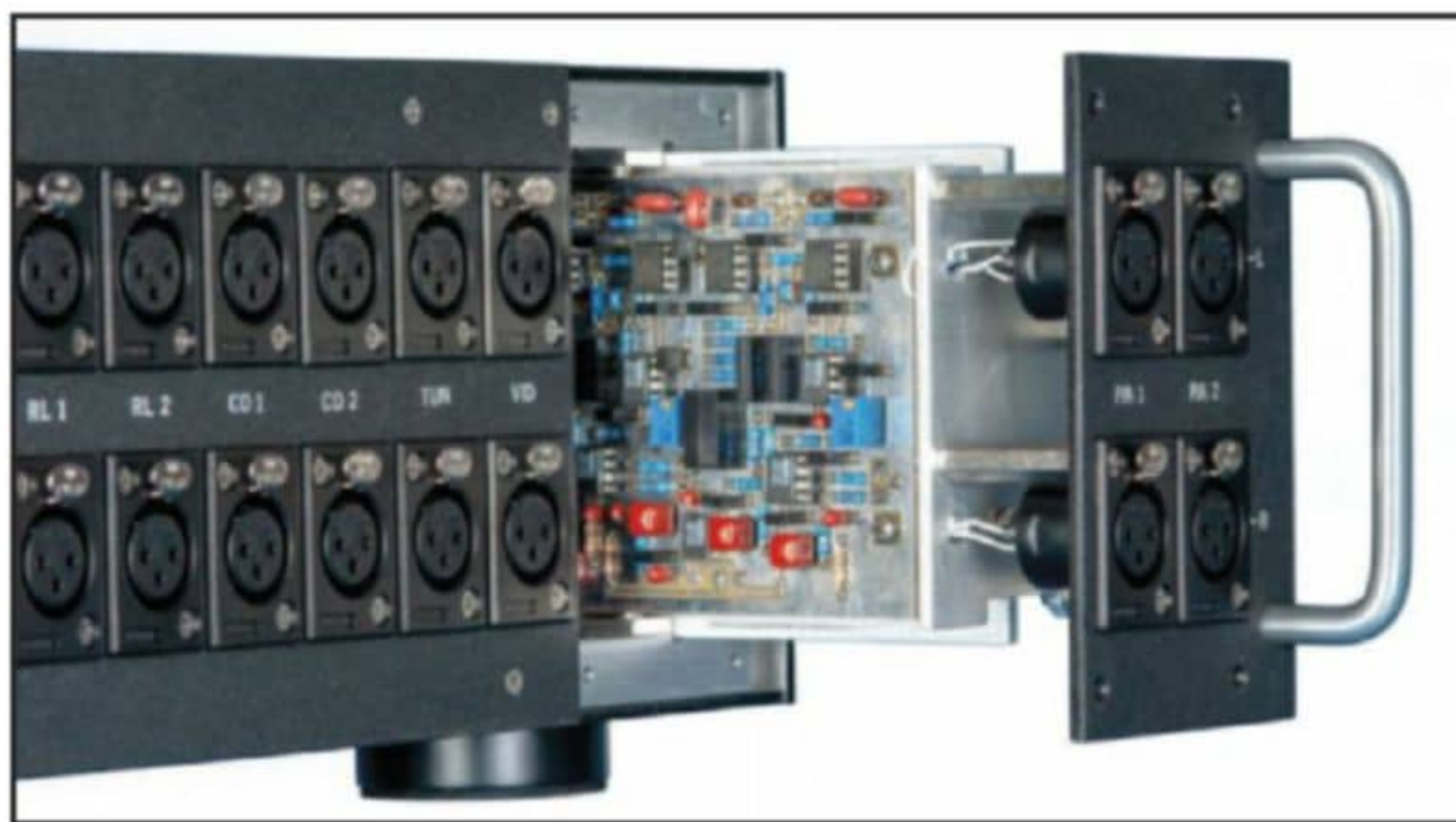
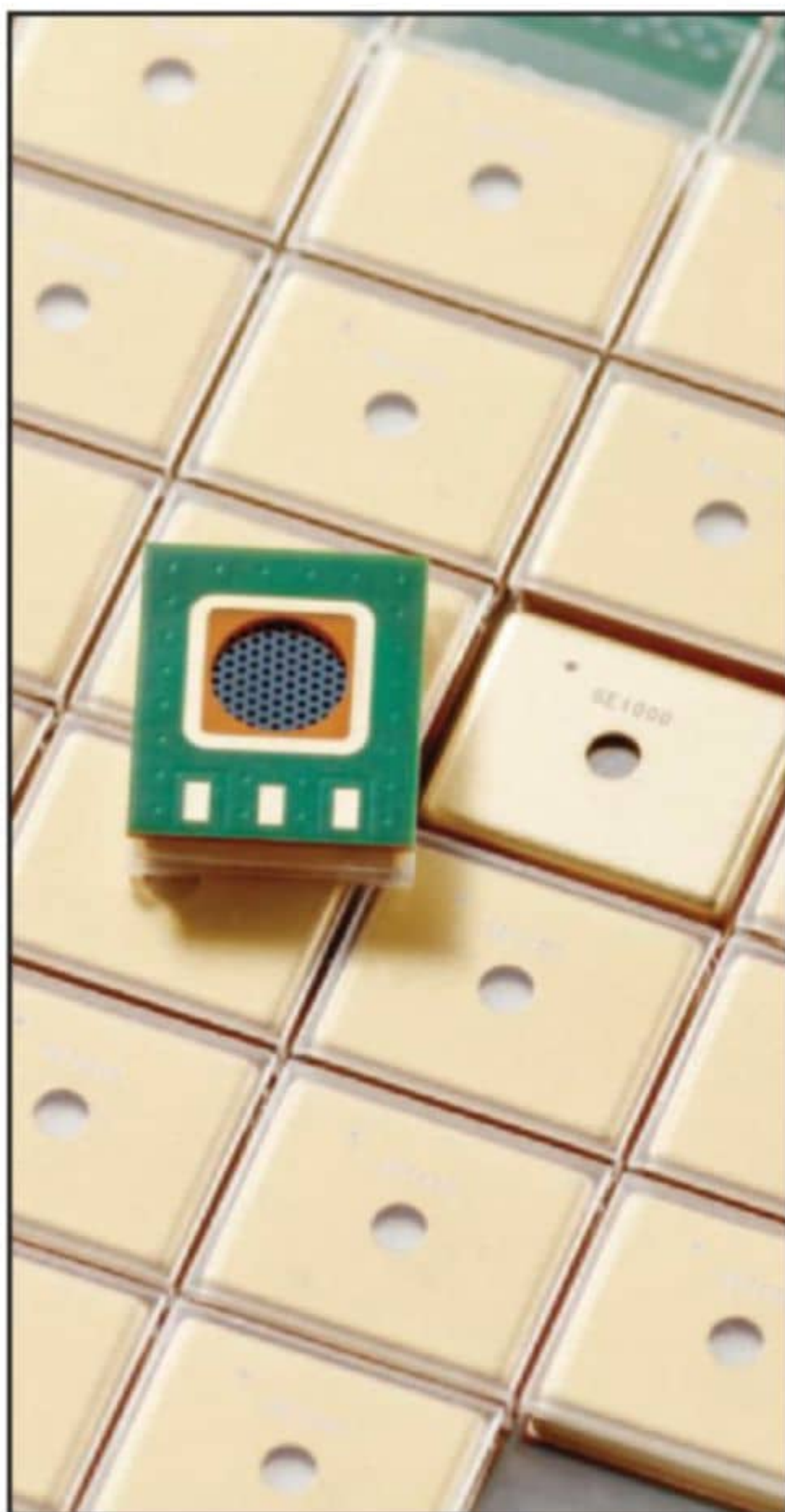


axCache

Every issue of audioXpress published since 2001, on USB.

Order at: cc-webshop.com





● Features

8 AXPONA 2023

Renewed Vitality on the Home Audio Front

By Oliver A. Masciarotte

The 2023 edition of AXPONA showcased some innovative products, plus it drew a more diverse and interesting crowd than in years past. Get an insider's look at the AXPONA show.

26 SonicEdge Ultrasound Speaker Modeling and Application

By René Christensen

Acculution has partnered with SonicEdge to design audio devices using SonicEdge's disruptive microspeakers for acoustic design. These speakers represent a paradigm shift in their operation and acoustic integration, and this article showcases some of the audio design work.

30 Noise Cancellation in Automotive Audio What's the Destination?

By Philipp Paul Klose

This article investigates how active noise cancellation (ANC) can be beneficial in an automotive application and looks at the challenges and technical implications in the development phase. It also provides an outlook for applications, products, and industry adoption.

36 Designing a True Midrange Driver

By Jørn Rune Kviserud

The mid frequencies in a multi-way speaker are often handled by a woofer. Here we detail many things that can be done to the driver to improve its properties.

44 Bass Reflex Performance Envelope Part 2 — System Analysis

By Gary Gesellchen

Part 2 of this novel approach to estimating the performance envelope from bass reflex loudspeakers shows how driver and port sizes determine how loud and how low a system will play, and how that relates to generally available drivers.

54 Instrumentation Amplifier-Based Preamplifier Line Stage

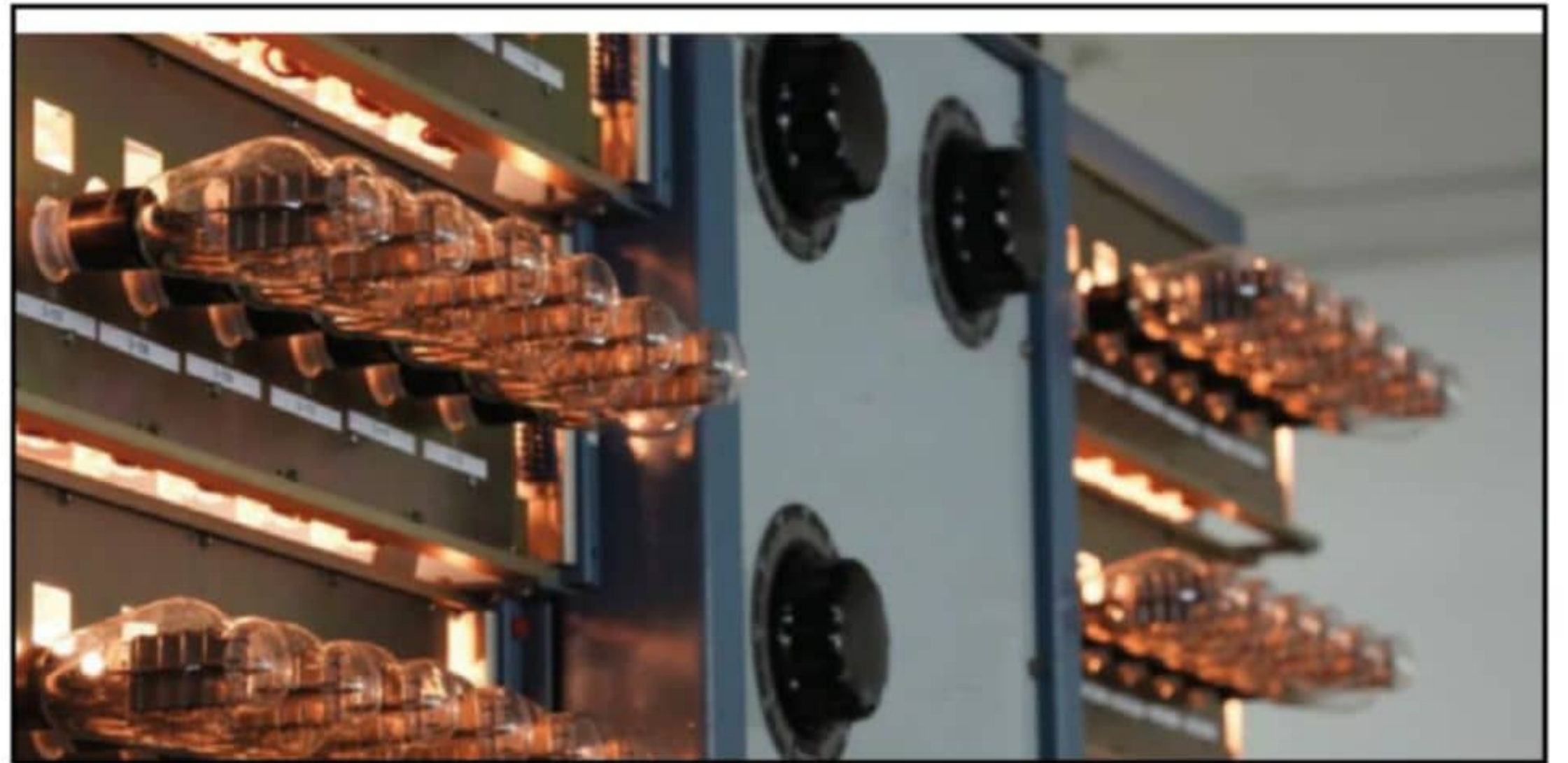
By Bruce Gillingham

This article describes a line stage preamplifier that also includes time-delay-matched, fully balanced (voltage and impedance) outputs, and adjustable gain.

60 Power Transformer Parameters, Selection, and Testing Part 10 — More Transformer Tests

By Chuck Hansen

In Part 10, we continue to test different types of transformers for leakage flux and external magnetic shielding, core loss sensitivity, and more.



● Acoustics

MARKET UPDATE

16 Electroacoustic Enhancement and Active Acoustics

By J. Martins

For this year's Acoustics special we are going to look at Active Acoustics and some of the companies offering solutions in this increasingly important field. And we start with a conversation and field experience with E-Cooustic Systems, the pioneers in this field.

● Departments

4 From the Editor's Desk

5 Supporting Companies

● Columns

HOLLOW-STATE ELECTRONICS

50 Visiting the Western Electric Vacuum Tube Factory

By Richard Honeycutt

Planned for some time, *audioXpress* was finally able to visit Western Electric in Rossville, GA. Officially established in 2018, this facility has been continuously producing 300B single-ended triode tubes since 2020 and is already expanding production to other vacuum tube models. Richard Honeycutt shares his guided tour.

● Websites

www.audioxpress.com

www.voicecoilmagazine.com

www.cc-webshop.com

www.loudspeakerindustrysourcebook.com

www.circuitcellar.com

www.linuxgizmos.com



@audioxp_editor



audioxpresscommunity



linkedin.com/company/audioxpress



audioxpress

AXPONA 2023

Renewed Vitality on the Home Audio Front

By
Oliver A. Masciarotte

With beautiful 80° weather following our drive to Chicago's western suburbs, I was excited to see old friends and immerse myself in quality stereo playback once again. The 2023 edition of AXPONA showcased some innovative products, plus it drew a more diverse and interesting crowd than in years past.

I started my mad sprint through the AXPONA show at AXISS Audio's suite (www.axissaudio.com), featuring Gauder Akustik's (www.gauderakustik.com) debut of its Arcona Mk II, Berlina Black Edition, Capello Mk II, and DARC Mk II loudspeakers, all with new drivers. Known for his very high-order crossovers, Dr. Roland Gauder was on hand to dote on his new DARCs on active demo. They employ a 12th-order passive crossover with a cleverly cascaded network that compensates for group delay. As supplied by AIR-TIGHT electronics (www.airtight-anm.com) fed from Yukiseimitsu Audio's (www.audio-yukiseimitsu.com) new \$50,000 AP-01 magnetically suspended turntable, the sound was even handed and engaging. In part, this was a result of the US debut of Nihon Onkyo Engineering's (www.noec.co.jp) consumer version of its all-wood Acoustic Grove System for diffusion in home listening environments.

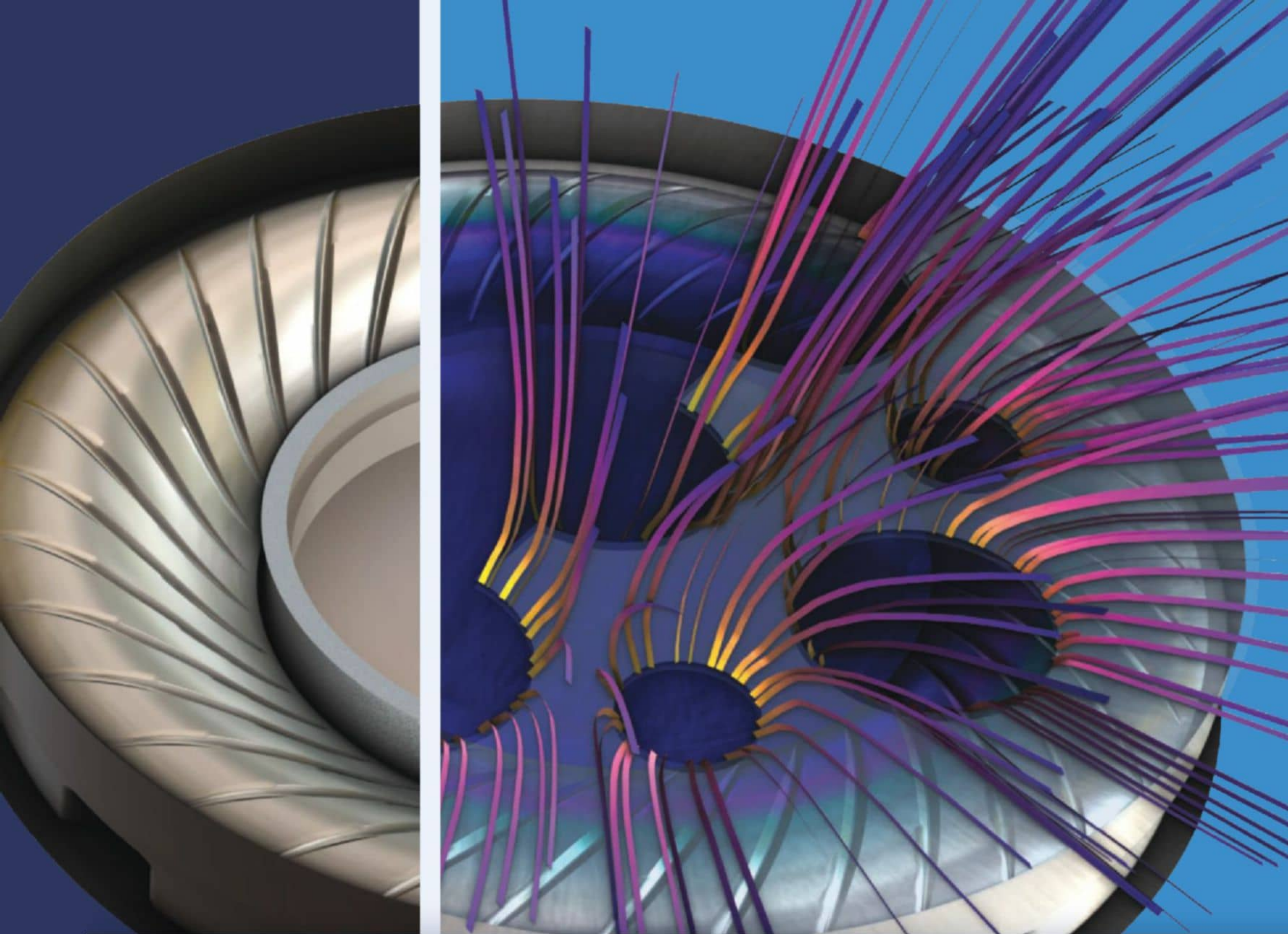
The Speaker Companies

One of the better-sounding, least cluttered, and unassuming demos at the show was Dutch & Dutch's room (www.dutchdutch.com), with a pair of its model 8c on custom Sound Anchor Comp ADJ stands (www.soundanchors.com). That's it. The 8c speakers were streaming Roon via Ethernet, using a MacBook Pro as the Roon Core. This was as bare bones as it could get, no treatment with

the speakers in the corners up against a window. I can imagine how much better the room would have sounded had they not been streaming from Roon but, as sales dude Gabriel Jones mentioned, they "...intentionally decided not to use any room treatment and placed the speakers in very non-conventional locations to better demonstrate their abilities." The 8c offers a cardioid midrange and controlled directivity at low frequencies. Bravo!

Next up, another nexus of good tunes, the Grimm Audio room (www.grimmaudio.com). I've known Eelco Grimm professionally for many decades, but this was the first time I'd heard an all-Grimm consumer setup. Another minimalist approach, their MU1 player drove the flagship \$25,300 LS1be speaker and SB1 subs. I loved it. With a wink, Eelco played me Billie Eilish's "Bad Guy" to great effect. Their upcoming MU2, on static display at AXPONA and live at High End Munich, will feature a discreet DAC, the same FPGA as in the MU1, a novel 1-bit noise modulator, and all analog outs.

An established name here in *audioXpress* is that of the late lamented Siegfried Linkwitz, who contributed volumes on loudspeaker theory and design. His speaker IP is now handled by EarFood GmbH (www.linkwitz.com), which was showing the new \$26,900 LINKWITZ LX521.4MG system, Panzerholz edition. The base model LX521 employs HDF and MDF baffles, while the additional



Take the Lead in Acoustics

with COMSOL Multiphysics®

Multiphysics simulation drives acoustics innovation by providing insight into every design aspect that influences product performance. The ability to account for coupled physics phenomena lets you predict, optimize, and virtually test a design under real-world conditions – even before a first prototype is built.

» comsol.com/feature/acoustics-innovation

\$3,000 adds the more inert Panzerholz phenolic-impregnated compressed wood laminate with walnut or aluminum fascia. Good looking and better performing...

As reported in these pages back in January of 2020, GoldenEar Technology (www.goldenear.com) is now under the AudioQuest umbrella. AXPONA 2023 saw the US premier of its newly refreshed, super-slim three-way T66 tower, arriving later this year for around \$7,000/pair. Five years earlier, in 2015, Uli Behringer's Music Tribe acquired the TC Group, of which Tannoy (www.tannoy.com) was a part. Soon to be 100 years old, Tannoy has been sleeping for a while but now that enterprise has been freed of its shackles. Tannoy is back! From on high, local control has been ceded to the hearty crew of hirsute Scots. The company claims that everything is again designed, engineered, and manufactured in the UK.

The precious metals used by CrystalCable, in part, dictate the cost.



The Dutch & Dutch 8c are not new but certainly worth hearing.



With cabinetry fabricated in England, everything else is again done in Scotland. The first new model in seven years, the Sterling III LZ (\$12,600/pair) was on static display. Hewing to the original 1967 design, the Sterling revival was driven by Japanese demand.

When I think of Tannoy, I think of improved versions from...Fyne Audio! At the show, Fyne (www.fyneaudio.com) had the US debut of its latest "Vintage Series," the \$35,000 Vintage Twelve. Manufactured entirely in Scotland, this two-way loudspeaker with retro looks is chock full of up-to-date technology including their signature 12" dual coaxial driver. Old pal John Bevier told me, "There's a 3" positive dome aluminum tweeter tucked behind their patented IsoFlare that places the tweeter into a recessed metal fluted waveguide that (aligns) the diaphragm" with the paper mid/bass driver. "The surround found on the mid/bass...shows a unique slight spiral with a deep channel molded into the material that helps eliminate distortions that traditional smooth lipped surrounds create." Whatever the tech, the result was lovely.

Companies Celebrate Their Longevity

Several anniversaries were discussed at the show, including this year's Linn Products Limited's 50th (www.linn.co.uk) and Luxman's 100th (www.luxmanamerica.com) coming up in 2026. Another old marquee that's now back on the scene is Mission (www.missionhifi.com), with a reimagined model 770, which originally came to market in 1978. As with Tannoy, the \$5,000/pair 770 "...heralds the return of UK-based manufacturing for



The cutest coax in the biz is now from Fyne Audio



Modern engineering drives Fyne's takes on the classics.



New to these pages, Geshelli Labs makes some of the most enjoyable gear at any price. This is a proto of the upcoming Zoofa.



The understated appearance of Grimm Audio's MU2 belies the sophisticated design.

the Mission brand," according to their marketers. In keeping with many other Brit brands, mineral-loaded polypropylene was chosen for the new 20cm mid/bass unit, which is mated to a 28mm coated, soft dome tweeter. The black/translucent drivers and white face with prominent logotype and purpose-built rectilinear stands combined to lend a mid-century modern air of sophisticated good looks, while the overall design offered a fat, warm, and punchy classic sound that made me smile. As provided by the folks at American Audio & Video (www.americaudiovideo.com), the 770s were driven by Audiolab 9000 Series electronics (www.audiolab.co.uk). Retro indeed! At the Just Audio booth (www.zushopjustaudio.com), Sales Manager Spencer Mills showed me another blast from the past, their magnificent wall of refurbished vintage Japanese CE receivers for resale... awesomeness!

Electronics

Speaking of electronics, there were a couple of products that caught my interest...First, Primare (www.primare.net) had a very flexible eight channel, Ncore NC500-based amp they were showing. The \$5,500 A35.8 is fully bridgeable to provide stereo



A top-shelf BOM makes Madisound's A50 a kit for the discriminating audiophile.

bi-amping or a single A35.8 can drive a 5.1, 7.1, or 5.1.2 channel setup.

Next, revenant (to the US) manufacturer Plinius Audio (www.pliniusaudio.com) previewed its Inspire 100 all-in-one streaming integrated. Built in New Zealand, as with all things Plinius, the \$4,400 model is due out mid summer. In another “How’s That For Convenience” category, the show saw the North American launch of Cambridge Audio’s MXN10 (www.cambridgeaudio.com), a network player that supports...wait for it...MPEG-DASH! Of course, Qobuz, Spotify Connect, and TIDAL streaming services are also in the mix. The MXN10 sends to USB, Bluetooth 5.0 A2DP/AVRCP supporting SBC and AAC, Airplay 2, and Chromecast. The GaN transistor guys at Orchard Audio had two new products for the show, the PecanPi+ streamer, starting at \$800, and the 500W per channel Starkrimson Stereo Ultra 2.0 amplifier, which starts at \$2650.

I had a nice talk with Gabi van der Kleij-Rijnveld about the North American debut of her new Diamond Series 2 cable family from Crystal Cable (www.crystalcable.com). After 12 years in development, the new cables (interconnects from \$540 to \$3,440) feature a novel silver with interstitial gold metallurgy, improved insulation, new construction, and upgraded connectors. At Linear Tube Audio (www.lineartubeaudio.com), its new baby was on static display, a pre-production prototype of their DAC, “... which will be released in a few months.” Meanwhile, there were some very nice sounds emanating from its \$7,650 Ultralinear+ integrated driving PureAudioProject’s \$6,500 Duet15 Prelude OB speakers (www.pureaudioproject.com) via an all-ANTICABLES (www.anticables.com) loom (\$215 to \$900).

Windermere, FL-based distributor Alex Sound Technology (www.alexsoundtech.com) gave me a very interesting demo of the new-to-me Diretta audio transport protocol that replaces UPnP and suppresses electrical current pulses resulting from the level shifting inherent in traditional audio signaling. Diretta addresses



The Acoustic Grove System (AGS), part of Nihon Onkyo Engineering’s room treatment solution



Phillips Design’s Oh! family of omnis would sit comfortably in a midcentury modern household.



The current newstalgia craze drives Mission 770 reissue, and I’m glad for it!



The 3,450 Plinius Koru phono pre atop the new M10 Reference preamp

ViBRIDGE

In-ear device testing – faithful to the bone

Bone conduction simulation – the game changer



REAL-LIFE-PROOF.

www.head-acoustics.de



Tannoy is back, and relying partially on its reputation to carry the day.



radiated noise, plus supply rail and ground plane modulation. Using a mix of its products that support the protocol and are new to the US, the \$32,000 Takatsuki TA-S01 SET amplifier being fed from the \$4,800 Sforzato DST-050EX transport, and \$3,6000 Pachanko Server Constellation MINI-SE with interconnects from Alex Sound's own True Power Lab (\$1,100 to \$3,300), the soundstage was better defined and the transient response clarified when compared to UPnP.

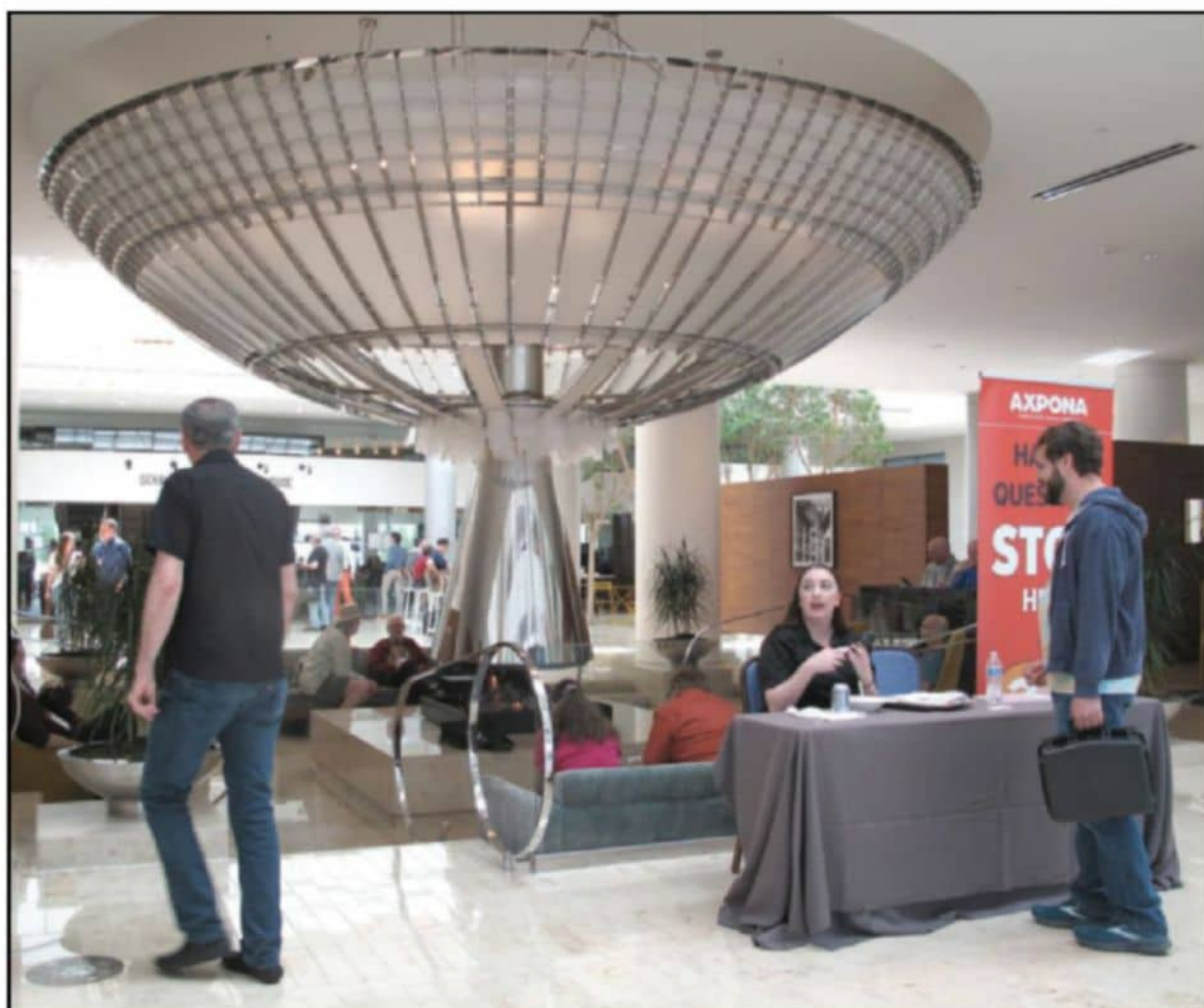
Output was delivered to the \$22,850 Blumenhofer Acoustics Genuin FS 2 Mk 2 bass reflex speakers with compression driver horn midrange. Being a digital interconnect method, both sender and receiver must support the protocol. In addition to Pachanko, Sforzato and several others, Fidata, Luxman and SPEC also support Diretta.

New in DIY Kits

DIY darling Madisound was showcasing a new Danish kit and new Norwegian drivers (www.madisoundspeakerstore.com). From Scan-Speak, the new A50 3.5-way kit was designed by Troels Gravesen. Top of the line Ellipticor transducers are utilized, including the new, "best in class" D8404/552000 dome midrange. Although the jury is still out, the ever-jovial Adam Johnson told me it's "...taking the crown from ATC as the world's best dome mid." At \$12,500/pair for drivers and premium crossover, no enclosure, it's the most expensive kit Madisound has ever offered.

From SEAS, they also had the new TPCD membrane drivers—the T29X tweeter and the W16NX woofer—to be released later this year. Johnson told me that "TPCD diaphragms are constructed using multiple thin layers of carbon fiber that are stacked and bonded together to form a lightweight and stiff structure. Each layer is engineered to cover the full cone or just some small areas, so that they combine (and) give stiffness to the areas we want stiffness and softer with more damping where that's needed. This construction technique results in a material that is much stiffer than traditional diaphragm materials such as paper or plastic, while still maintaining a low mass and good internal damping."

Speaking of ATC (www.atc.audio), its already exceptional products are now reduced in price by 25% due to optimized internal freight infrastructure. Still on a DIY kick, I was hanging out with the hi-fi handymen in the Parts Express room (www.parts-express.com) and learned about the \$675 Orian three-way bookshelf kit with flat pack enclosures. With a 5¼" coaxial mid/tweeter and a high excursion 8" woofer, the Orian makes for a



The Renaissance Schaumburg is easy to navigate and has plenty of spaces to take a break from the hustle and bustle of the show.

much-appreciated graduation gift or almost instant upgrade project for a second system.

Newcomers

Always lusted after MBL’s amazing loudspeakers (www.mbl.de) but can’t swing the monthly payments? Newcomer Phillips Design Company (www.phillips-design.com), based out of furniture capitol Grand Rapids, MI, was demonstrating its cylindrical, omnidirectional three-way priced from a relatively affordable \$4,500 to \$6,700. At 78 pounds each and sheathed in your choice of fabric or leather, these hefty Corian-topped omnis exhibited very broad uniform horizontal coverage with a relaxed and convincing voice.

Debuts

AXPONA saw the worldwide rollout of PAHMER enterprises’ line (www.pahmerenterprises.com) of four bass reflex Audio Craftsmen loudspeakers, driven by an Accuphase receiver. They range in price from \$4,000/pair for the stand-mounted two-way Laval to \$17,000 for a couple of 2.5-way floor-standing Halifax. The \$1,500 solid-wood Z stands are a perfect complement to the upscale and attractive \$9,000/pair Victorias.

Back in the October 2020 issue of *audioXpress* (www.audioxpress.com/files/attachment/2726), Kent Peterson and I characterized and reviewed Magnepan’s (www.magnepan.com) Little Ribbon Speaker or LRS. I found their ragged response and lack of low end enough to have sold my pair not long after purchase.

In the article, I did mention upgrades and, had I known about VPE at the time, I probably would have kept them. VPE Electrodynamic’s \$1,600/pair folded active dipole Little Dipole Woofer (LDW) Model 1 not only adheres to the TLA, but they are expressly designed to complement the LRS. In conjunction with VPE’s \$300/pair Magna Riser Airborne stands (www.vpeelectrodynamics.com), the wedge-shaped Model 1s sit comfortably below and behind the LRSs, very effectively augmenting the LRS’ anemic bottom three octaves. My notes on the Model 1 simply says, “Game changer!”

Overall Impressions

Showcasing a range of new and exciting products that catered to a range of different needs and budgets, the AXPONA show continues to lead with a friendly and easy-to-navigate venue, a galaxy of high-quality exhibitors, and educational programs that inform and entertain. I really can’t wait until this time next year for another dose of contrasting tech and crazy good sounds! 



Just Audio was kickin’ it old skool!



VPE Electrodynamic’s LDW Model 1 folded active dipole sub lend validity to Magnepan’s LRS.

About the Author

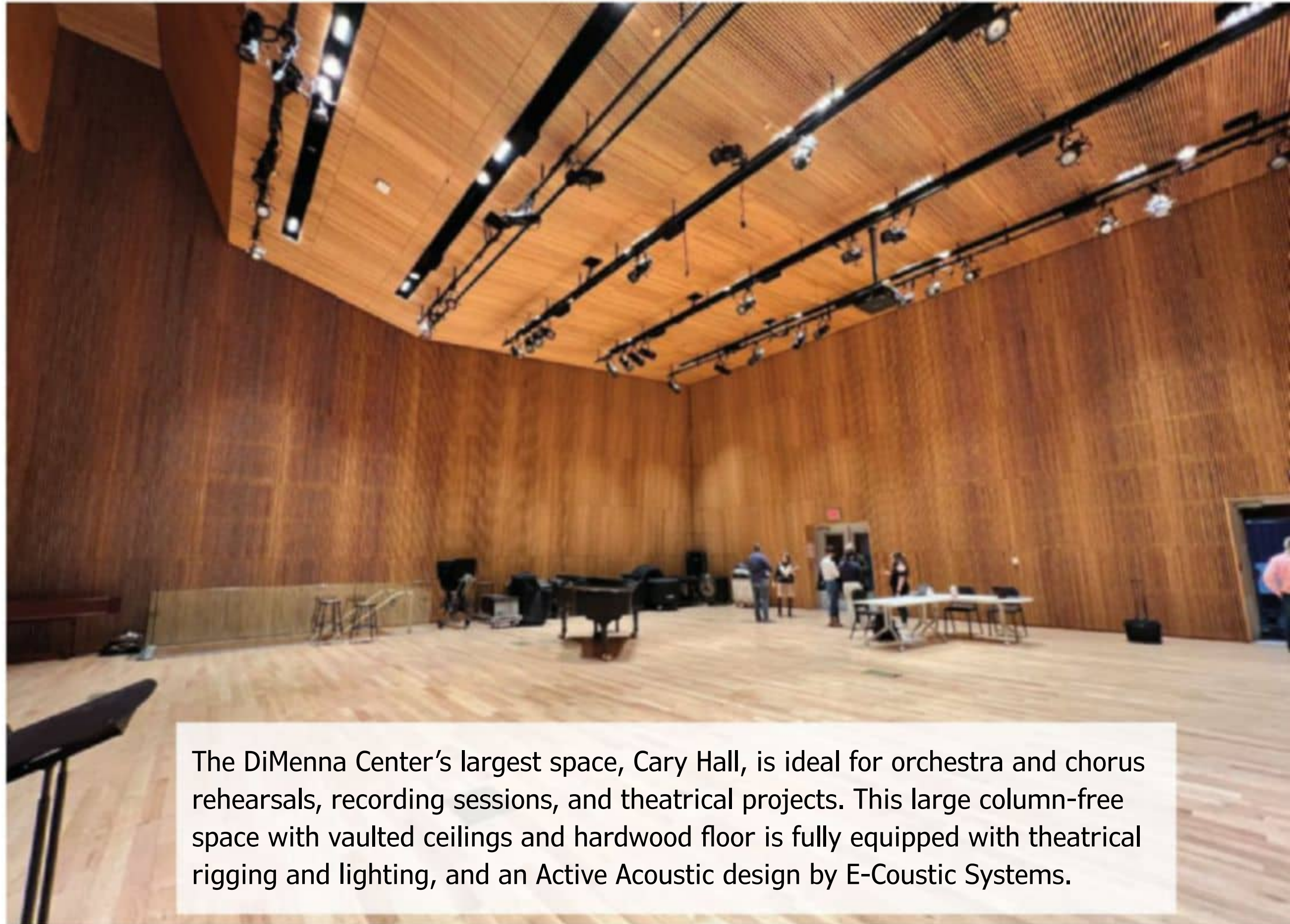
Oliver A. Masciarotte has spent more than 40 years immersed in the tech space, working on manufacturing, marketing, and product development for many pro and CE audio manufacturers including dbx, a/d/s, Lexicon, Sonic Solutions, and Minnetonka Audio. His client roster is as diverse as Apple, Harper Collins, 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.



Electroacoustic Enhancement and Active Acoustics

For this year's Acoustics special we are going to look at Active Acoustics and some of the companies offering solutions in this increasingly important field. And we start with a conversation and field experience with E-Coustic Systems, the pioneers in this field.

By
J. Martins



The DiMenna Center's largest space, Cary Hall, is ideal for orchestra and chorus rehearsals, recording sessions, and theatrical projects. This large column-free space with vaulted ceilings and hardwood floor is fully equipped with theatrical rigging and lighting, and an Active Acoustic design by E-Coustic Systems.

Noise curves and in particular Noise Criteria (NC) curves are an interesting metric in acoustics. In a simplistic way, these can be described as the curves immediately below the sound frequency spectrum created by the background noise in the room while the room is unoccupied.

These metrics are being constantly analyzed and revised, with the ANSI/ASA S12.2-2019 document "Criteria for Evaluating Room Noise" being an important recent update of the standard, expanding NC curves, and introducing a method for evaluating low-frequency fluctuating noise using room noise criterion (RNC) curves [1].

An NC value of 15-20 is normally recommended for performance spaces, as an example. Although Noise Rating (NR) is a metric also commonly used to define hearing safety, speech communication and annoyance thresholds, NC is still a common metric because it better defines noise "comfort."

Noise comfort? You know you appreciate good acoustics when you enter a space—and most likely, the low noise level is a key factor offering a contrast with any sound sources and the way they propagate in the room. As an example, in a very quiet room we can easily notice a hum or buzz originating from a power line or a light source—electromagnetically-induced acoustic noise. When that noise source is eliminated, we can better appreciate the characteristics of a musical instrument played in that space.

In acoustics, noise control of internal and external sources is a foundation that determines how much every other type of acoustical treatment will be effective for the intended purposes.

The DiMenna Center

We can appreciate different acoustics in a recording room, a cathedral, or a concert hall. And yet, each is very different. A quiet room, where we can "enjoy" the silence, provides an immediate sense of comfort.

That is what I experienced in The DiMenna Center for Classical Music, touted as "New York's only venue designed by musicians for musicians."

It's not easy to have a space of these acoustic qualities and enjoy absolute silence right in the core of New York, and yet the Mary Flagler Cary Hall in The DiMenna Center delivers just that [2].

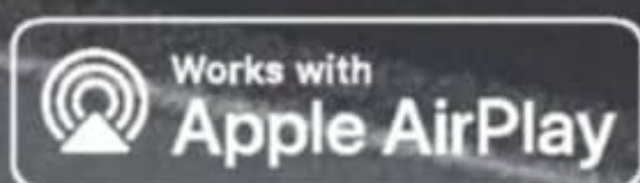
That's where I met with Steve Barbar, co-creator of the Lexicon Acoustic Reinforcement Enhancement System (LARES), currently president and co-founder of E-Coustic Systems.

The Orchestra of St. Luke's built The DiMenna Center for Classical Music in 2011 as New York City's only acoustically optimized rehearsal and recording space. Custom designed to accommodate musicians ranging from soloists to full symphony orchestras, the largest space, Cary Hall, has been in high demand also for castings, theatrical projects, and recording sessions (including film soundtracks). So much so, that the Center added a state-of-the-art Dolby-certified control room in the building.

Located close to the Javits Center, Hudson Yards, Penn Station, and Times Square in midtown-Manhattan, Cary Hall at The DiMenna Center is centrally located and is now an ideal environment for a wide range of activities, including concerts. The reason why it can accommodate so many different types of events is because the



BLUESOUND



Qualcomm[®]
aptX[™] HD



POWERNODE EDGE VERSATILE POWER

The POWERNODE EDGE is a two-channel BluOS[™] streaming amplifier that is more powerful than its size suggests. Versatile enough to fit into a 1U rack height, or mounted on a wall or placed on a table top, it offers HiFi sound for any smart home music system. Come see the POWERNODE EDGE at CEDIA, booth #3516.



CONNECTS WITH
Control4
CERTIFIED

ELAN
HOME SYSTEMS

U-C

RTI

CRESTRON

LUTRON

The panoramic view of The DiMenna Center's Cary Hall shows the conductor's podium position. Two rows of speakers are placed behind the wood walls.



room is equipped with active acoustics, designed and supervised by E-Coustic Systems. Of course, the room was carefully built and designed originally for orchestra rehearsals, but the acoustic results were never satisfactory. The room is equipped with curtain rails, because originally they thought the space would be reverberant, until they discovered that it was too dry. So, the first thing they did was use materials to increase the wall reflectivity, which never achieved the intended goals. Without E-Coustic Systems' intervention, it would have been impossible to accommodate all the multiple demands, which today expand to recording sessions for major TV and film productions, including for Netflix, HBO, and Disney.

Pioneers in Active Acoustics

Acoustic enhancement approaches have existed for many years, with passive, mechanical systems mostly applied to create variable acoustics. Using digital signal processing and loudspeakers to control the acoustics is a relatively recent approach. David Griesinger and Steve Barbar, the team behind the LARES systems, have been doing that continuously for more time than anyone else. Their work in this field started at Lexicon in 1988, and LARES originated as a separate company division in

1990. Since then, their approach has been used in thousands of temporary installations, and they have designed hundreds of permanent systems for concert halls, opera houses, performance venues, houses of worship, scientific research labs, and more.

Today, the company is called E-Coustic Systems and is the world's most experienced developer and integrator of "electro-acoustic enhancement products," as they still like to describe.

The Millennium Park in Chicago, IL, where the Jay Pritzker pavilion is located, and the Grant Park Music festival takes place, is probably one of the company's most emblematic installations. During the Summer festival, the active acoustic system extends into one of the longest and largest open areas, serving a very large audience.

The research work of David Griesinger leads the vast and fascinating evolution of acoustics and electroacoustics, and in recent years extended to 3D sound and even binaural reproduction, heading an industry trend.

Steve Barbar is the main force behind E-Coustic Systems activities in the field, working with a vast—and prestigious—number of acoustical design companies and professionals globally. Unique systems that improve the listening experience in a wide variety of environments and applications and provide "a cost-effective means of resolving sonic challenges in ways that traditional architecture can't easily accomplish."

In the early days with LARES, Griesinger and Barbar's work in concert halls, opera houses, and events, such as at the famous Vienna Philharmonic Orchestra outdoor concerts, relied heavily on a combination of art and science. Sometimes, it was an exercise in restraint and diplomatic skills to deal with extreme artistic temperaments and highly subjective criteria, not necessarily serving the audience's best interest. And when it worked to everyone's satisfaction... it was almost always kept a secret.

Their scientific and technically effective approach allowed them to face most of those challenges and gradually build a number of references that served to reinforce their own confidence in the active acoustics approach and helped them achieve the



Outside facade and cut-out of the building. The DiMenna Center for Classical Music rehearsal and recording facility in New York, NY, was built in 2011. The prestigious acoustical consultancy firm Akustiks provided the acoustical and audio video design, including fully isolated rehearsal and practice rooms that are extensively interconnected with audio and video infrastructure. The installations were damaged by a flood in 2018, and the refurbishment provided the opportunity for the installation of updated systems, including the new recording control room designed by Francis Manzella Design.

indispensable recognition among the key shareholders of these type of ventures and investments. From opera houses all over the world, to the Amsterdam Muziektheatre and The Netflix Corporate Auditorium in California, to internal research facilities for Apple, the company discreetly serves hundreds of satisfied customers.

And with time, their technology and systems also evolved to become more reliable, effective, and affordable. At least in respect to processing power, signal routing, and network infrastructure. While other key aspects such as loudspeakers and microphones, familiarity, repeatability, and serviceability proved to be the most important assets.

E-coustic Systems is today “the product of over 30 years research in the fields of digital audio, neuroscience, and acoustics—refined by more than 20+ years of successful installations in a wide variety of venues and applications.”

At the core, we find server-class processors (now in their third generation) that run patented signal processing techniques and algorithms. These can scale according to the installation and number of channels. An E-Performance system supports up to 512 audio channels, while an E-Architecture system meets the most demanding applications to accommodate venues with seating capacity in excess of 50,000, and the most challenging audio applications.

Unlike the approach seen in other active acoustics systems, microphones are used to pick up only the direct and reflected sound generated by the sound source and are kept to the bare minimum. E-coustic Systems’ patented processing uses independent time variant channels that de-correlate both the individual loudspeakers and the microphones, with the reduced number of microphones helping to increase gain before feedback, which also explains the stability and effectiveness of the company’s approach. Loudspeaker numbers—fairly traditional two-way designs—are as numerous as the installation demands, with a reduced number of subwoofers also used. Key in this approach is the complete absence of coloration or system instability, allowing separate adjustments of reverberation time and reverberation level throughout the venue.

E-Architecture MainFrame III acoustics processors are also equipped with independent analog inputs for surround sound and effects channels and can even accommodate multi-channel digital inputs from digital consoles or playback systems for special applications.

The design of their typical loudspeakers, such as the E-coustic Systems 440 model, is dictated not by sound reinforcement power levels, but mainly by its high-fidelity, critical to the success or failure of the installation. The simple, front-ported design delivers wide-band power uniformity and excellent low-frequency response and withstands installation in different venues with consistent results.

The goal of these systems consists of ensuring the perfect acoustics for the space, according to the application. Coverage requirements are achieved in this case purely with the number of loudspeakers used. Achieving the ideal conditions, conversations can occur inside the space with maximum comfort—as we discovered—recordings are improved, and rehearsing and performing is a pleasure. If sound reinforcement of any of the sources is required, this can be easily achieved using independent



The E-Coustic Systems control panel placed on the doorway to Cary Hall allows full control by users of the active acoustics, according to the requirements and different usages of the room.



The complete E-Coustic Systems for the room active acoustics and all the auxiliary signal management is placed in these two half racks. The left side contains the E-Coustic Systems MP-8 Microphone Preamplifier, power conditioner, network router, Crestron control system, and the two main E-Coustic Mainframe III (that run all the advanced acoustics algorithms), and 512-channel Matrix Processor units. The right side is where the AO-32 Analog Output Interfaces provide the signal conversion from the digital processors back to analog balanced circuits, followed by the eight-channel power amplifiers that feed all the 56 speaker channels in the room.

speakers placed in the room—but they don’t need that many or for them to be very powerful since the acoustic conditions are already perfect.

Inside Cary Hall

Using the opportunity of the Audio Engineering Society (AES) convention being at the nearby Jacob Javits Center, I requested a visit to the emblematic Cary Hall in the DiMenna Center, where I was privileged to meet Steve Barbar and learn about the room, the installation, and E-Coustic Systems. This is also a space that is conveniently used by the company for demonstrations, being such a versatile and effectively convincing example of what active acoustics can do.

Entering the impressive room, Barbar quickly pointed to the familiar Crestron touchscreen control panel placed in one of the doorways. A panel that can easily be understood by users of the room, appreciated by maestros, musicians, and sound engineers alike. Level control, Mute System, and reverberation time presets: 1.8 to 2.8 seconds were visible, and more could be selected.

Given that no musicians were using the space at the time, we used a portable speaker playing different solo instruments and tracks. Perfect propagation and definition of the source in the



Steve Barbar showing us the positions of the very few microphones suspended from the ceiling. “This is one of those things that you have to know what to do with it. How and why I put my speakers and microphones this way. Nothing here interferes with the perception of clarity.”

wide space. While we moved freely around the room and checked the systems in place, we kept talking comfortably no matter the distance. Very impressive.

“This room is pretty quiet. It’s a good quiet room,” Barbar says looking at our pleased faces. “This being an orchestral rehearsal space, I see that people value the acoustic quality and feel comfortable here without even caring about how it works. They are doing recordings here, and not just any recordings... They are doing real stuff here.”

Of course, with more than 30 years of systems in the market in many different types of venues, I was curious to learn how frequently E-coustic Systems updates those installations. Apparently not as frequently as common-sense dictates, also because normally they work so well that patrons tend to forget they exist. Which causes panic calls when they are not working as intended.

As Barbar highlights, electronics evolve much faster and these systems in the field could benefit from regular maintenance and upgrades. “There is a system here in a church in New York that is 30 years old. It’s been running for 30 years, they know they have to do something, and every year I get panic calls when the system is not working. That’s one of the original LARES systems.”

In comparison, the Cary Hall system is a relatively modern installation and is the result of a cooperation between E-coustic Systems and The DiMenna Center. “The hardware here is now five years old—it’s our system, we have to keep it up. I just replaced the Crestron control, we placed a network management system in, which allows me to log into the server and see if this system is up and running. I can push code to the Crestron, I can push code to the processors. The new processors have motherboards with a dedicated management port. Over a KVM, we can actually watch everything boot up and change any of the bios settings.”

During our conversation I learned that although E-coustic Systems keeps testing the latest generation Intel platforms with more cores, the bottleneck for this type of installation is now less in the processing power and more in the channel count. “AMD and NVIDIA are really fast chips, but most things top up at 128 channels, for audio—we don’t. The small matrix processor here is 512 channels point-to-point-to-point. When we are trying to get 8000 channels, you’re going to generate some heat. It’s our decision that we are going to use server class engines,” explains Barbar.

And channels in these processing systems is separate from the number of speakers. From typical systems of 156 speakers in performance spaces, to single large rooms like Cary Hall, where there are 56 speakers—two layers of lateral upper and lower on each wall. In this case, each speaker is spaced at least 6’ apart. For some installations, where more energy is required in a similar area, E-coustic Systems can double the number of speakers, but they use a different model. Always targeting power uniformity.

“Any time you put up a number of loudspeakers along a line, you create an array whether you like it or not. If you don’t have power uniformity, you begin to hear these beams as sources, particularly when they are lateral. And any time you use a directional speaker, any kind of waveguide, you are going to get some sort of beamforming.”

And the speaker models specified must remain relatively stable over the years, because the company needs to keep those installations working consistently for 20 years or more.

"I've been using SEAS drivers for years. It's great stuff because it's repeatable. That's the important part. We don't see many differences between the drivers. The important thing is that I can get the same driver for this loudspeaker after 28 years. Exactly the same driver."

And that choice of drivers is like every other single component in any of their systems, where reliability and consistency, with the highest level of performance is key. Those are requirements that do not necessarily fit in any typical manufacturer's or application market standards.

"You cannot just use commercial stuff. Every little piece matters here. You end up chasing studio-grade equipment, and that cost is a big part of the overall cost of this system.

"What people don't really understand well enough is that everything matters. In a room this quiet, you have to have low noise at the microphone capsules. You have to have a low noise preamp. You have to have low noise in all your components in between. You do need power uniformity. But the biggest piece of it is that this algorithm has clarity. We're at 3 seconds of reverb here and we can sit here and talk. It sounds clear!"

And another thing that these systems compensate for is the background noise itself, such as when the HVAC is turned on or off. With the support of four subwoofers in the room, the system handles the full frequency spectrum of frequencies.

Not being either the largest or the most representative of an E-coustic Systems installation, Cary Hall is clearly a great space for demonstrations and to create a positive impression about the company's technology. And of course, this is also a place with a very high profile of users.

"Getting into this venue is tough. Finding an hour here, since they are always busy with all the guest recordings they do here, it's crazy. But this is also a place where it is easy to get a duet, trio, or quartet, to play because the orchestra's here full time. And it's quiet."

As we confessed to Barbar, our curiosity about the room increased when we learned that companies such as Disney feeling comfortable to record there. "Disney loves this place! These cast recordings they do here are a big thing. They take weeks with it," he explained. "We had an open house here four or five years ago, and when they heard the room, they have been here ever since. The meeting room was turned into a recording control room. We provided all the interconnectivity for the recordings behind these panels. And the system is always on. If they want to adjust it, they know how to use it."

Beyond Active Acoustics

During our conversation with Barbar, we realized that we were just touching the surface of the things that E-coustic Systems does. After all, it's clear that very few people know so much about what can be done with active acoustics, and how not to do it. This team pioneered the best digital reverberations for studio production in the industry at Lexicon. But they quickly evolved



Steve Barbar, co-inventor of the Lexicon Acoustic Reinforcement Enhancement System (LARES) with David Griesinger at Lexicon, Inc., is currently the president of E-Coustic Systems and the main force behind the company's designs and activities in the field.

from that experience into live performance systems, where things are exponentially more complex. And gradually E-coustic Systems evolved into the most sophisticated processing algorithms for effective coding of aural cues. A work with substantial implications for the acoustic design of spaces built for music and speech that granted David Griesinger the Wallace Clement Sabine Award from the Acoustical Society of America (ASA).

"We started in '88 with this, and we went through a bunch of trial and errors. We were down the rabbit hole a good shot," Steve Barbar confesses.

"One factor here is that early energy matters. The way this system is laid out is that early goes everywhere. It's timed everywhere. Late reverberation goes there. Late lateral goes here. Front and back goes the other way. We have independent control of all of that. But it's all timed and adjusted by each speaker to each location. So that when you walk the room, the room is working with the system, not the system working against the room. That's a big part of what creates clarity.

"Part of any system that we tune—the biggest part of it—is the design. You need it in that place for a reason. I put my microphones where they are needed in the room for a reason. And I put my loudspeakers that are required for a reason. And if I've done my work right, inherently knowing my loudspeakers, inherently knowing the feedback loop gain of the system, I can then determine whether I can meet the objectives or not. Again, it's what you are doing and why that counts."

www.ecousticsystems.com

References

- [1] ANSI/ASA S12.2-2019, "Criteria For Evaluating Room Noise," American National Standards Institute, <https://webstore.ansi.org/standards/asa/ansiasas122019>
- [2] "About Orchestra of St. Luke's," YouTube, <https://youtu.be/VIMmD-FAnKM>

Meyer Sound

Constellation

It's hard to say which of the active acoustics solutions have more installations in use every day, but Meyer Sound's Constellation can easily claim to be the most recognized system.

This active acoustics system was introduced in 2006 and has been discreetly and gradually leading in public recognition for variable acoustics and the design of performance spaces. Using an extensive array of ambient sensing microphones, sophisticated digital signal processing (DSP), and loudspeakers, Constellation modifies the reverberant characteristics of a venue and redistributes sound throughout the space, ensuring an enhanced experience in every seat. Each loudspeaker and microphone is connected to the Meyer Sound D-Mitri digital audio processing system, which includes microphone inputs, loudspeaker outputs, and the complex signal processing necessary to implement active acoustics management.

It wasn't the first approach to active (or virtual) acoustics, but it was certainly the most integrated, full system's approach to the concept—normally tied to the sound reinforcement system available to the space itself. John Meyer's personal dedication and passion for the approach led Meyer Sound Laboratories to continuously improve and refine the Constellation Acoustic System, expanding its application to venues and installations globally.

As the processing technology evolved, Meyer Sound's Constellation has earned acceptance across a broadening range of acoustic system applications in classrooms, boardrooms, houses of worship, and restaurants alongside the technology's initial focus on performing arts centers and music venues.

Constellation provides the flexibility of variable acoustics across a wide spectrum of hall sizes and applications, from the 2,500 capacity Jakarta International Expo (JIExpo) Theater to the Seattle Symphony's intimate, low-ceiling Octave 9: Raisbeck Music Center, now transformed into an ideal space for a variety of chamber and experimental music concerts. High-profile



The San Francisco Symphony's SoundBox performance space. Typical Meyer Sound Constellation systems combine active acoustics with sound reinforcement requirements, and immersive sound installations, with all components and services supplied by Meyer Sound, including a dedicated acoustics team leading the design and commissioning.



Meyer Sound Constellation has transformed hundreds of venues worldwide through temporary installs and touring systems.

installations include the San Francisco Symphony's innovative SoundBox, the New World Symphony's outdoor SoundScape Park WALLCAST, and the Nine Trees Shanghai Future Art Centre—the first Constellation system in China. Constellation also expands dimensionality for immersive spectacles, including regular resident Las Vegas, NV, productions.

The acoustical flexibility afforded by Constellation has provided solutions in applications outside musical performance. Houses of worship are employing the power and flexibility of Constellation to improve congregational involvement and to host both contemporary and traditional worship in the same space. In education, leading institutions like University of Copenhagen's Maersk Tower research building and Sweden's Karolinska Institute medical school are using Constellation to allow more direct and intimate communication between students and professors in larger classrooms and lecture halls. Constellation is also employed to create multiple acoustical environments for research in human hearing at McMaster University in Hamilton, Ontario, Canada, and Northwestern University in Illinois.

And Constellation has been adopted to improve corporate communications in auditoriums (Telstra in Australia, for example) and boardrooms (Jyske Bank in Denmark), while also offering push-button options to convert the room into a performance venue for company or community music ensembles. Even restaurants are now discovering the unique ability of Constellation



A Meyer Sound Constellation is used in a bank's boardroom, with the multiple microphones and compact speakers visible in the ceiling.

to create a lively acoustical ambience while still allowing quiet, intimate conversations among patrons.

A Complete Systems Approach

Constellation’s capability to transform any room’s acoustics allows architects to conceive spaces differently from the outset, according to John Pellowe, Project Director—Constellation Systems at Meyer Sound. “The room acoustics are no longer dictated by the physical structure alone,” he observes. “Architects now can design performance spaces with a lower room volume without compromising the acoustics. This not only reduces construction costs but also long-term energy use. More and more, we are finding new performance spaces designed from the outset for Constellation, as was the case with new halls in Norway, Estonia, and Indonesia.”

Constellation is always supplied as a turnkey solution, with all hardware and software components supplied through Meyer Sound—the microphones conforming to Meyer Sound specifications are made by DPA—and with Meyer Sound’s Constellation team responsible for design, supervision of installation, and final commissioning and tuning. At the heart of Constellation is the patented Variable Room Acoustic System (VRAS) created by New Zealand acoustician Dr. Mark Poletti. Since its introduction in 2006, Meyer Sound Constellation acoustic systems have been completed or are currently underway in 27 countries on six continents.

www.meyersound.com



A recent example of a Meyer Sound Constellation system installed at the Cornish College of the Arts’ Raisbeck Auditorium, in downtown Seattle, WA. The installation comprises a total of 110 Meyer Sound loudspeakers, deployed in four sub-systems. One of those sub-systems is a Constellation and Atmos-capable cinema surround system that uses 70 small full-range loudspeakers and 16 subwoofers. Sixty-four miniature condenser microphones are arrayed around the space for ambient acoustical sensing.

Amadeus Acoustics

From Active Acoustics to Immersive Performance

Austrian company Amadeus Acoustics is a more recent player in the active acoustics space, founded in 2018, and inheriting all the accumulated experience from the founder’s team, led by Thorsten Rohde and Fabio Kaiser.

They differentiate their approach based on a full system design that expands from Acoustic Enhancement to Immersive Audio installations based on “variable and optimal acoustics with absolute predictability.” Also, the company’s system approach is manufacturer-independent for loudspeakers, amplifiers, microphones, and network components, other than using its own Core processor and proprietary software.

From small theaters to big concert halls, Amadeus Acoustics basically offers systems that are independent of the architecture or location and respond to the needs of multipurpose rooms and live performance events—including outdoors and orchestra shells.

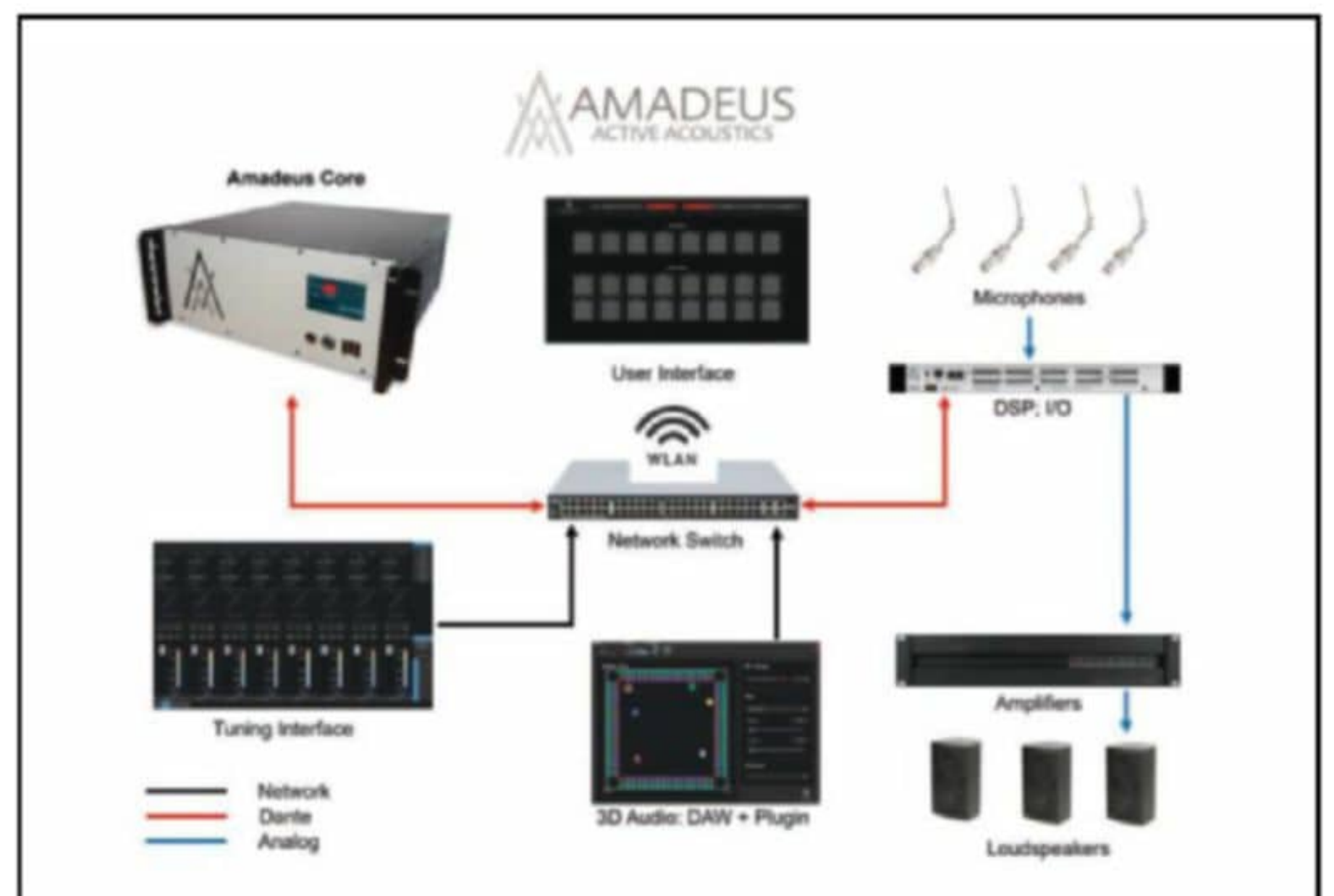
Active Acoustics as a Service

Learning from the experience of previous pioneers in this space, Amadeus Acoustics focused on building a flexible acoustic enhancement solution that can be offered both for permanent installations but also for temporary events and as a service.

The Amadeus Core processor contains algorithms to calculate early reflections based on a 3D model of the room and the directional creation of source positions. This creates an inherently natural sound, as the active acoustics are seamlessly interleaved with the prevalent acoustics of the room.

The processing itself is time-invariant without coloration or any noticeable latency. A 128-channel reverb generator is included, which takes early reflections patterns into account and expands the capabilities of the reverb enhancement.

The Core processor can run active room acoustics and immersive/3D audio, feeding separate devices. As the company’s 3D Audio module runs in parallel to the acoustics module, Amadeus



A system diagram of a typical Amadeus Active Acoustics installation where the only dedicated technology is the Amadeus Core digital processor and software.

sees this integration as an added benefit for its approach—and one that is increasingly being adopted by other manufacturers (e.g., Meyer Sound, d&b audiotechnik, and L-Acoustics), given the system requirements for immersive audio installations.

“For the first time, our system combines the two different approaches to sound design in rooms. The Amadeus Core is basically a bullet-proof computer running the Amadeus Core software. This approach reduces the development and manufacturing costs compared to special DSP hardware platforms, which require very complex programming. Also costs for repairing or updating the system are therefore much lower,” states the company.

The company’s system approach also benefits from its use of Dante network integration to lower the costs, using two base systems with 128 channels and 32 channels.

In terms of the installation, the Amadeus Active Acoustics approach is somewhat similar to the Constellation system, using related loudspeakers and microphones to completely cover the room, on walls, ceiling, under balconies, and on stage. In terms of processing, Amadeus is able to offer a hybrid system, with a



A simplified illustration of an Amadeus Active Acoustics room is shown with microphones and speakers.

regenerative approach for natural reverberation enhancement, or an in-line approach for increased loudness indoors or generation of the acoustics outdoors. The signal processing also uses a double Matrix Approach and additional decorrelated algorithmic reverb based on neutral Feedback Delay Networks, independent for each channel.

The software allows independent control of sound energy distribution within the room, with time, sound, density, and reverb shaping, all with a dedicated graphic user interface. The 3D sound or immersive audio module, expands this approach to the sound reinforcement of the room, using an independent speaker system that supports virtual surround sound to simulate 7.1, 9.1.4, or even Dolby Atmos loudspeaker layouts.

The Amadeus Core system uses a web-based control interface, allowing the company to offer long-term service agreements.

www.amadeus-acoustics.com



Fehlmann Architects and Kahle Acoustics worked together to reshape the Théâtre de Beaulieu in Lausanne, Switzerland. Amadeus Acoustics worked closely with Kahle Acoustics to integrate optimized acoustics in all parameters, amplify the room response, and provide even distribution throughout the auditorium. Custom presets were created to accommodate two orchestras that regularly perform there.

L-Acoustics

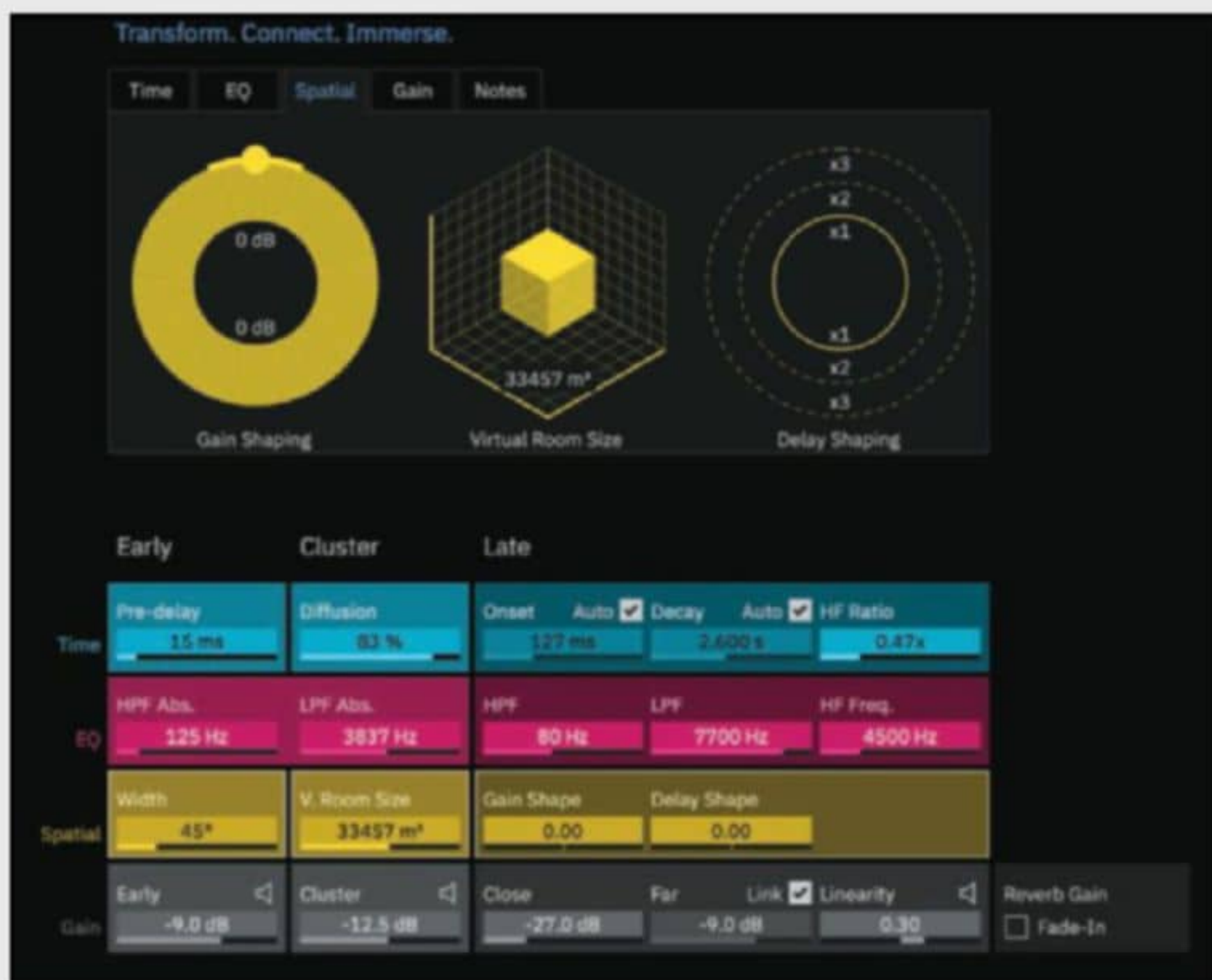
Ambiance Virtual Acoustics System, Powered by L-ISA

In 2022, L-Acoustics promoted the new L-ISA Ambiance virtual acoustics enhancement system as a natural expansion for its sound reinforcement systems and notably, its more recently immersive audio efforts. For the French company—a global leader in live performance and touring systems—virtual acoustics or active acoustics is the logical next step in the evolution from immersive audio.

Some manufacturers that never conceived immersive audio as such have long been researching how to emulate different room acoustics in the same space, and indirectly found that by

distributing speakers across the room, sound reinforcement also becomes exponentially more effective. More recently, other loudspeaker manufacturers that have long been proposing separate active acoustics systems—such as Meyer Sound—are also merging that concept with new immersive sound solutions, while a pioneer in immersive, such as d&b audiotechnik with its Soundscape solution, has always incorporated an acoustic enhancement component.

L-Acoustics now jumps to the forefront of active acoustics, since it is currently the market leader in large immersive audio

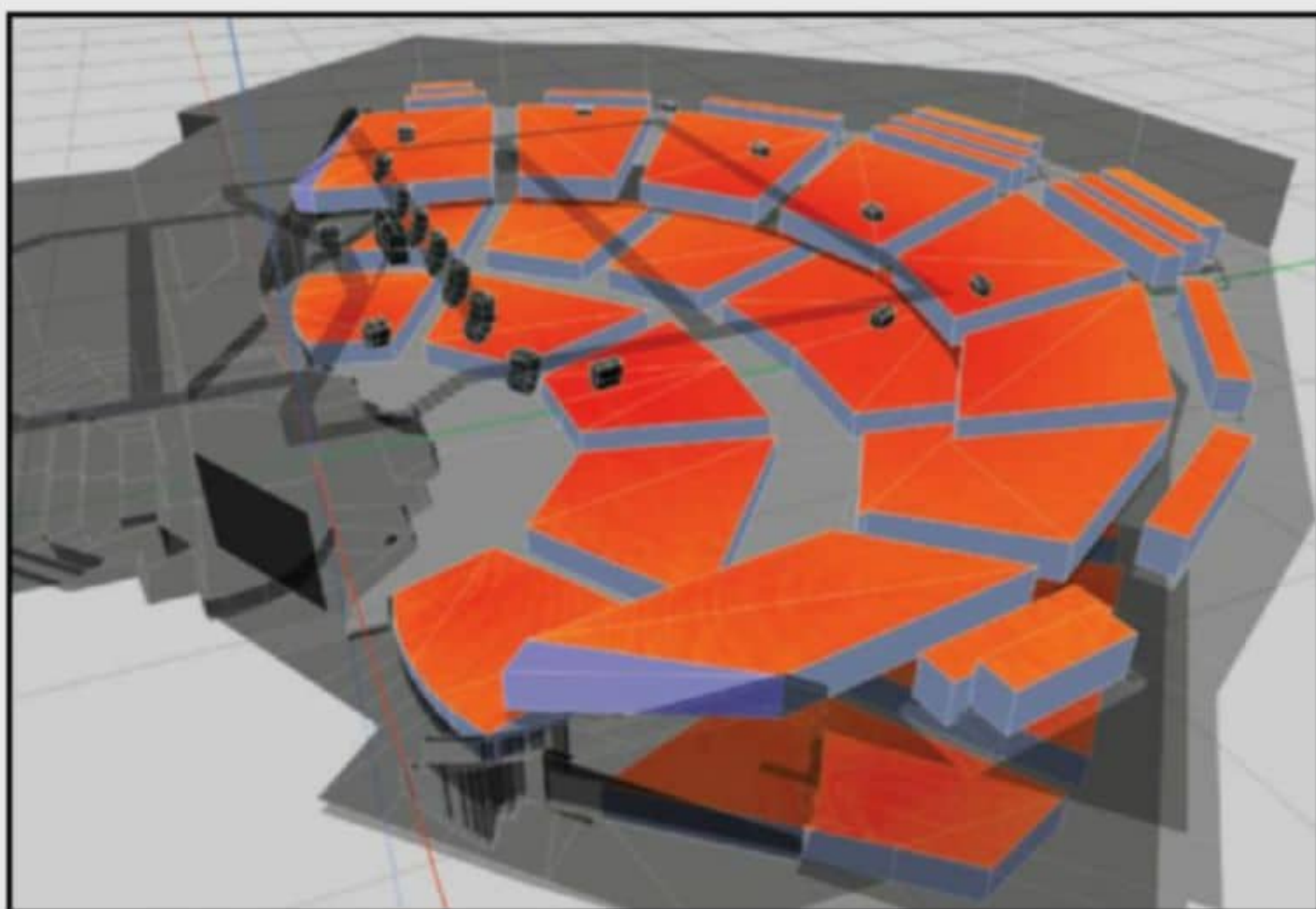


Advanced 3D Room Engine. L-Acoustics Ambiance uses an array of distributed microphones to capture the existing acoustic energy of the space and processes these signals through the spatially aware L-ISA Room Engine.

installations, and it has its work half cut to transform those spaces. For some time already, L-Acoustics has been designing a new generation of electroacoustic systems that, while intended for immersive applications, are perfect for active acoustics. And the powerful L-ISA processors are also in place and ready. The L-ISA Ambiance virtual acoustics enhancement system uses L-Acoustics loudspeaker systems, microphones, and an advanced 3D Room Engine powered by the L-ISA Processor II.

Acoustics Enhancement System

The idea for Ambiance is to empower acousticians, theater and audio-visual consultants, composers, and sound designers to transform the acoustic properties and qualities of a venue space to match the desired atmosphere. Using microphones with L-Acoustics hardware and software, sound designers can use L-ISA Ambiance to “reconnect space and sound to immerse audiences in new acoustic worlds,” as L-Acoustics describes.



L-Acoustics Ambiance active acoustics uses the same L-ISA immersive loudspeaker and electronic systems to create immersive experiences, and adding an acoustic layer to individually adjust early, cluster, and late reflections.

The architecture and acoustics of a venue play an integral part in the audience and performer experience. From concert halls with long reverberation times that impart a lush feel for orchestras, to cinemas and halls with minimal reverb that allows for clear speech intelligibility, the experience of a space is as much due to the feel of its sound as to its decor. Where once venues were designed for single-use, today’s spaces are multi-purpose, changing with the calendar to better serve wider audiences.

Adapting acoustics through architecture is time-consuming, budget-hungry, and does not guarantee flexibility. The L-ISA Ambiance virtual acoustic enhancement system uses hardware—speakers, amplifiers, and processors—that are already a necessary part of modern venues. Adding strategically placed microphones to capture the existing acoustical energy of a space, Ambiance processes signals from the mics through the L-ISA Room Engine that’s built into the L-ISA Processor II, where reflections can then be individually managed with separate EQ, time, and space parameters, empowering a sound designer to create a desired sound atmosphere for the space. L-ISA Ambiance can be controlled via the L-ISA Controller on both Mac and Windows platforms, through the built-in configurations on the Processor, or with Q-Sys and Crestron third-party control systems.

“An L-ISA Ambiance system provides flexibility to a venue, opening up acoustic enhancement to a wider market, without the need to alter the look of the space,” explains Jordan Tani, Product and Technology Marketing Engineer at L-Acoustics. “Installing Ambiance not only gives a space future-proof flexibility, but it also opens up creative possibilities for designers to construct—and reconstruct—the aural feel of the space to their desire.”

The L-ISA Room Engine allows sound designers to spatially distribute early, late, and cluster reflections to the loudspeaker system to provide a true multi-channel room engine that considers the geometry and volume of the space itself. The L-ISA Room Engine gives users the power to individually customize gain, EQ, time, and space parameters to craft fine-tuned acoustic profiles. By managing these parameters, sound designers can tailor the feel of a space to create the desired atmosphere for the audience, the performers, or both.

L-ISA Ambiance can be used for acoustic enhancement applications or combined with L-ISA Immersive Hyperreal systems for object-based mixing in live performances or experiential playback in theme parks, immersive art installations or other experiential events. L-ISA Ambiance is already shaping sound spaces at numerous sound experiences and installations globally.

www.l-acoustics.com



The first permanent L-Acoustics Ambiance Acoustic System was installed at The New Theatre, in Pulse, Czech Republic’s fourth largest city. With Ambiance, the room is now pre-tuned for orchestral music, operas, or theater productions, with a reverberation time from 0.95s up to 2.0s available with the simple press of a button on the L-ISA Controller.

SonicEdge Ultrasound Speaker

Modeling and Application

Acculution has partnered with SonicEdge to design audio devices using SonicEdge's disruptive microspeakers for acoustic design. These speakers represent a paradigm shift in their operation and acoustic integration, and this article showcases some of the audio design work.

By
René Christensen

SonicEdge developed a unique speaker based on a MEMS technology that generates sound in a new way. The company was founded in Israel in 2019 and has expanded rapidly worldwide. SonicEdge's office in Denmark has been part of the Sound Hub Denmark and Danish Sound Cluster since 2021. The SonicEdge speakers utilize active ultrasound modulation [1], providing enhanced audio at low frequencies and extended operation up to 20kHz and beyond. These speakers are smaller in size, produce no vibrations, have a reduced back cavity, are easy to use and integrate, and can be manufactured with high uniformity, similar to semiconductors. SonicEdge's initial focus is on the \$80 billion USD true wireless stereo (TWS) market, where their speakers enable smaller, more comfortable, and better sounding earphones.



Photo 1: The SonicEdge MEMS chip

Transducer Principle

The SonicEdge transducer is based on actively modulated ultrasound to produce sound. Several key points set this transducer apart from other types, such as electrodynamic transducers or balanced armature receivers [2]. SonicEdge's first product is the SE1000. It has 200 modulated ultrasound transducers, each composed of an ultrasound speaker and an acoustic modulator. The total driver active area is 10mm². **Photo 1** shows the MEMS chip and the packaged device. The individual transducer cross-section is shown in **Photo 2**.

The structure is composed of three membranes. The bottom membrane is called an ultrasound speaker and the top membrane is called a demodulator. Both membranes are driven electrostatically, with the middle membrane functioning as a ground. In **Figure 1**, f_1 and f_2 represent the vibration frequencies. The bottom membrane is excited via a signal, which is the multiplication of an acoustic signal $a(t)$ and a carrier signal $\sin(2\pi f_c t)$:

$$x_1(t) = a(t) \sin(2\pi f_c t)$$

This results in a so-called Dual Band Modulation, resulting in two sidebands around the carrier frequency. If we illustrate this with a simple sinusoidal acoustic signal, the result is the sum of two signals—one having frequency content at the sum of the carrier frequency and the acoustical frequency, and the other having the difference frequency:

$$x_1(t) = \sin(2\pi(f_a + f_c)t) + \sin(2\pi(f_a - f_c)t)$$

Note that both components are at ultrasound frequencies, which generally holds for any relevant acoustical signal. The top membrane is actuated to vibrate, and its vibration defines the channel cross-section between the top and middle membrane. As a result, the movement of the top membrane changes the acoustic impedance of the ultrasound signal, so the resulting signal is amplitude modulation (AM) of the ultrasound signal. The modulator acoustic action is given by:

$$Mod = L \left(\frac{1}{2} + mB \sin(2\pi f_c t) \right)$$

L and m can be found from the acoustic loss imposed by the modulator on the acoustic beam in the open and closed position of the modulator.

The final output from the single cell is proportional to the sum of the carrier signal and the modulated air channel as in:

$$x_{out}(t) = \frac{1}{2} kLa(t) \sin(2\pi f_c t) + \frac{1}{2} kLmBa(t) (1 + \sin(2\pi 2f_c t))$$

Here, the audio output can be recognized in the middle part of the equation. Whereas ultrasonic output is present in addition to the audio output, the carrier frequency is at hundreds of kilohertz, and the presence of ultrasonic energy does not cause any health issues to humans, even with long-term exposure [3].

It may seem like a roundabout way to generate the acoustic output compared to simply applying the audio signal to a single membrane. But the unique setup with modulated ultrasound comes with several advantages:

- The only “part” moving at acoustic frequencies is the air, hence no mechanical vibrations, and there is no need to address mechanical vibration considerations in earphone or hearing aid design.
- The transducer is essentially a constant flow generator and will be unaffected by external acoustic loads.
- The transducer is modeled as a constant “current” source or constant volume velocity in the acoustic domain when using lumped or finite element modeling.

- Size reduction results from the understanding that SonicEdge’s speaker replaces the membrane in standard speakers with a pump operating at hundreds of kilohertz, pushing air more effectively and requiring 10 times less area and 10 times less displacement.

Modeling of SonicEdge Speaker

A standard electrodynamic speaker can be described by a two-way Lorentz force coupling between the voice coil (electromagnetic)

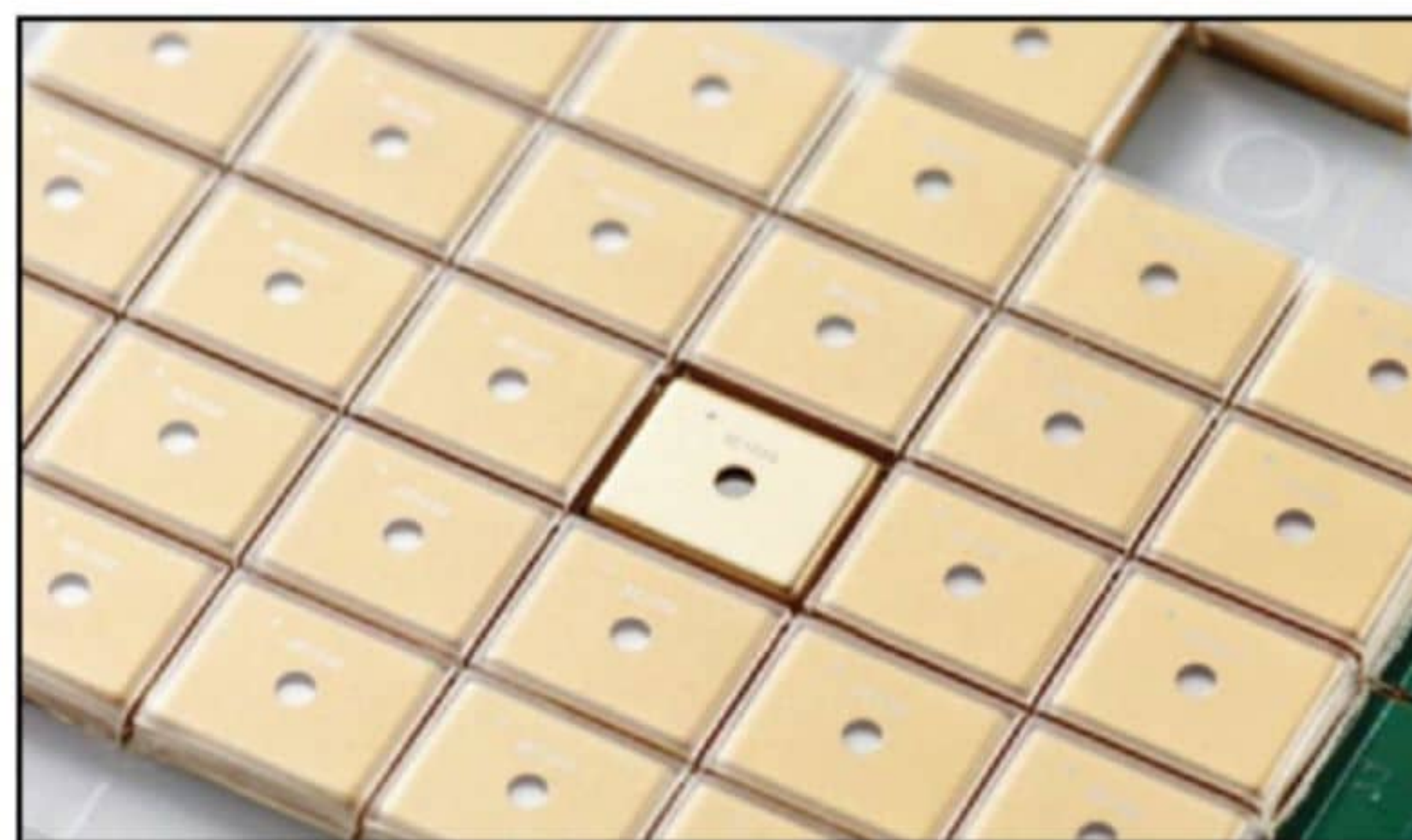


Photo 2: The MEMS chip is packaged using MEMS microphone design rules.

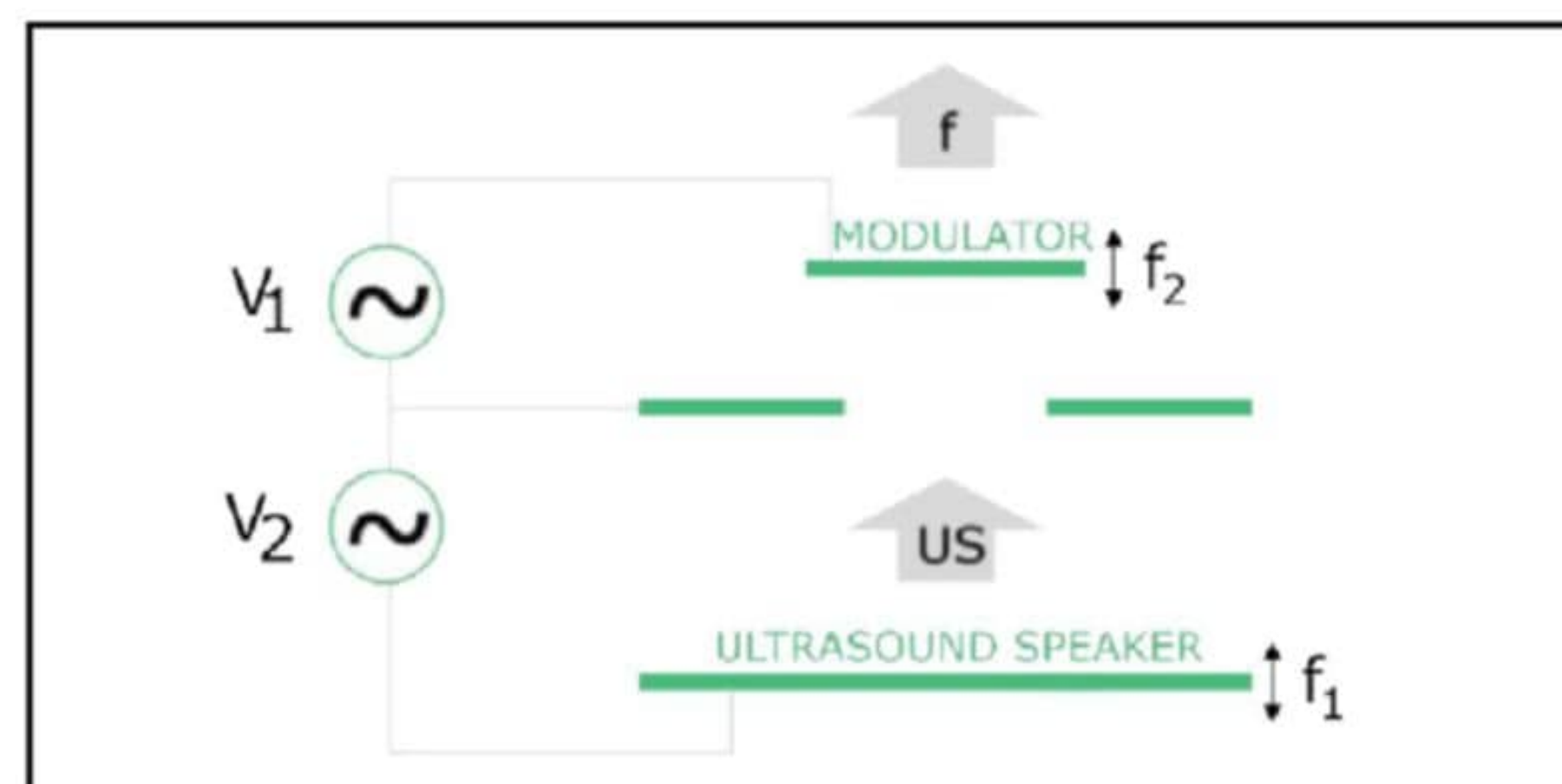


Figure 1: Schematic view of the cross-section of the transducer

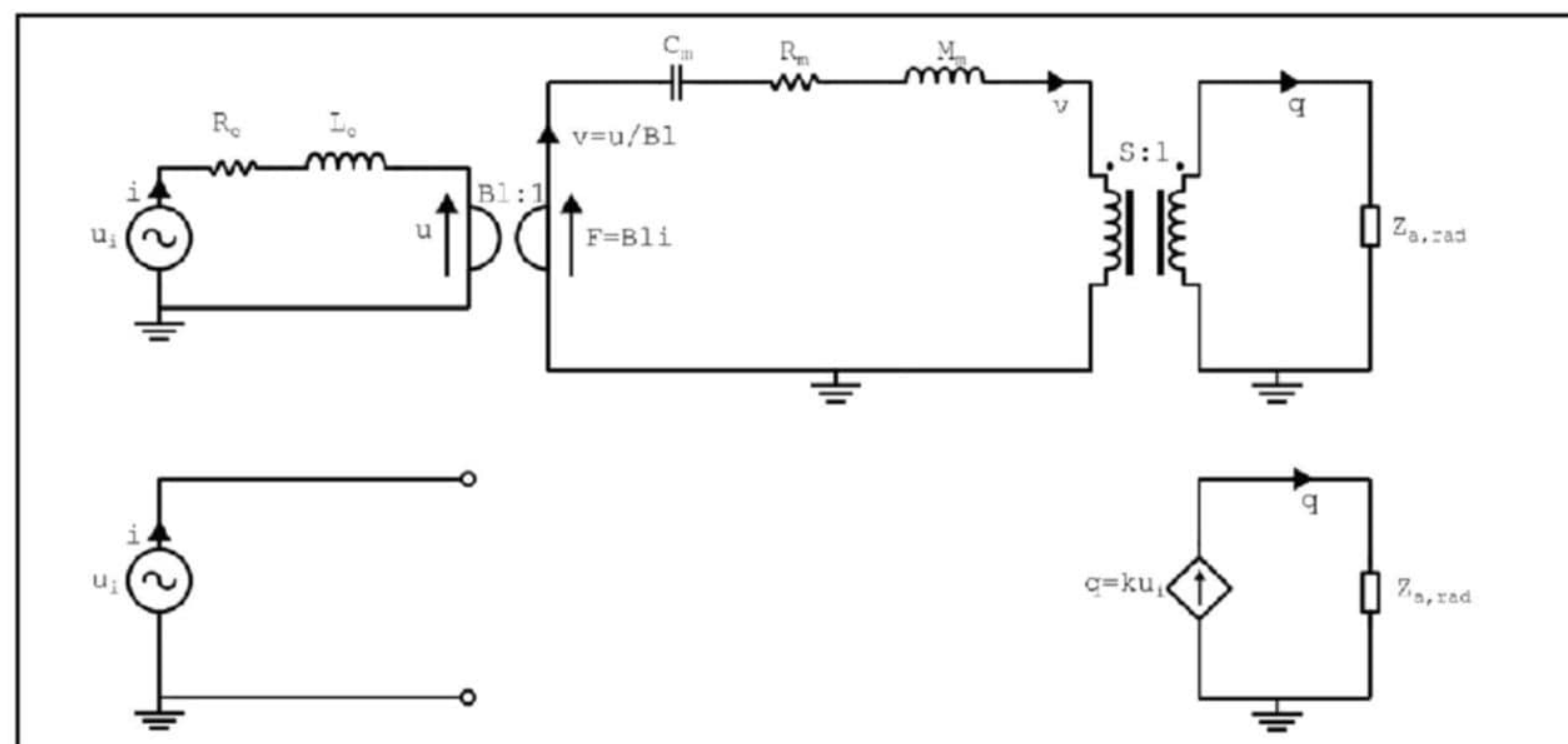


Figure 2: Lumped model circuits for a standard electrodynamic driver (top) and the SonicEdge driver (bottom)

and the moving parts (structural mechanics). The mechanical setup is that of a spring-mass system with an associated fundamental frequency. Above this fundamental frequency, the behavior is mass-controlled with a resulting flat acceleration response, whereas below the fundamental frequency, the behavior is stiffness

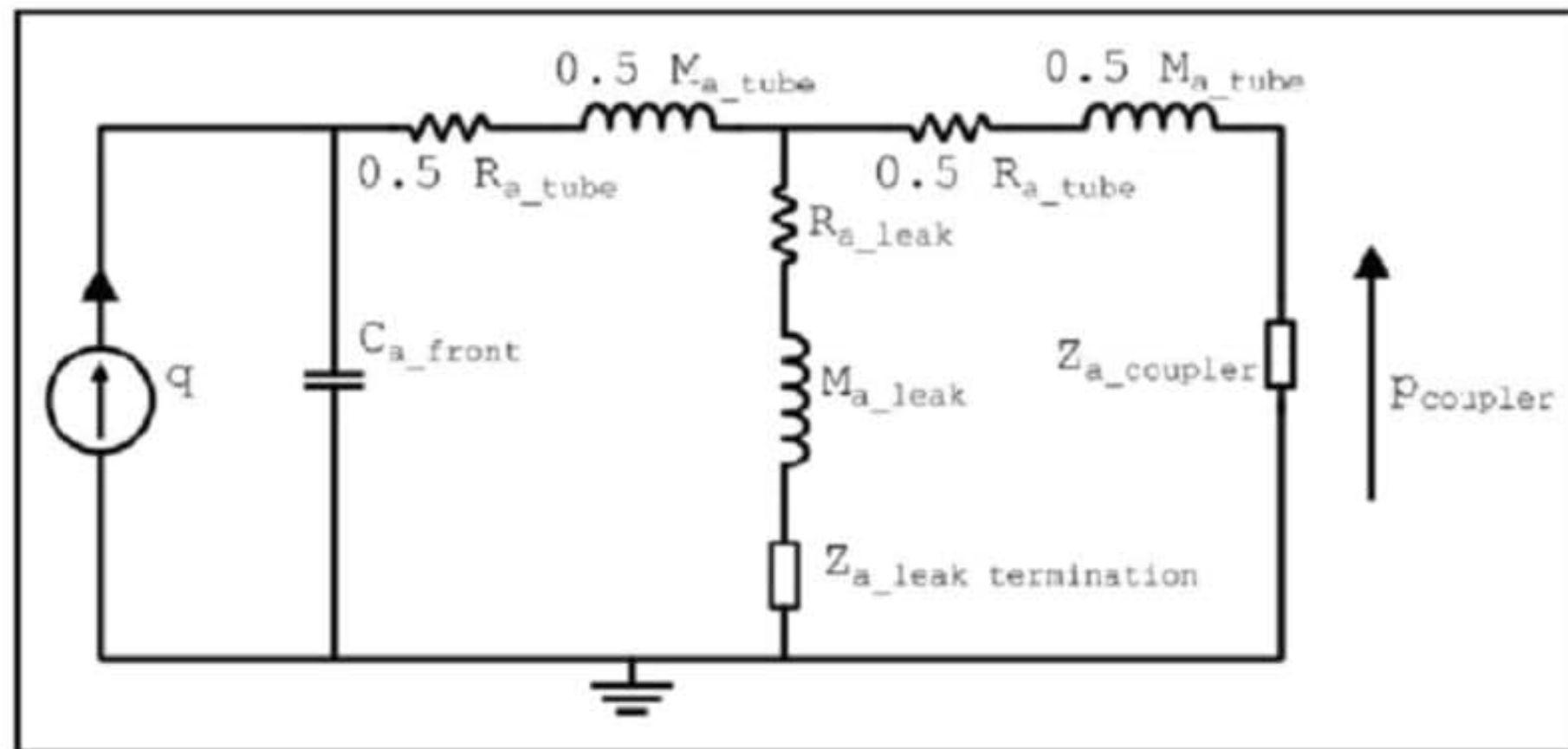


Figure 3: A lumped model for the transducer placed in front volume with a leaky tube connection to a coupler

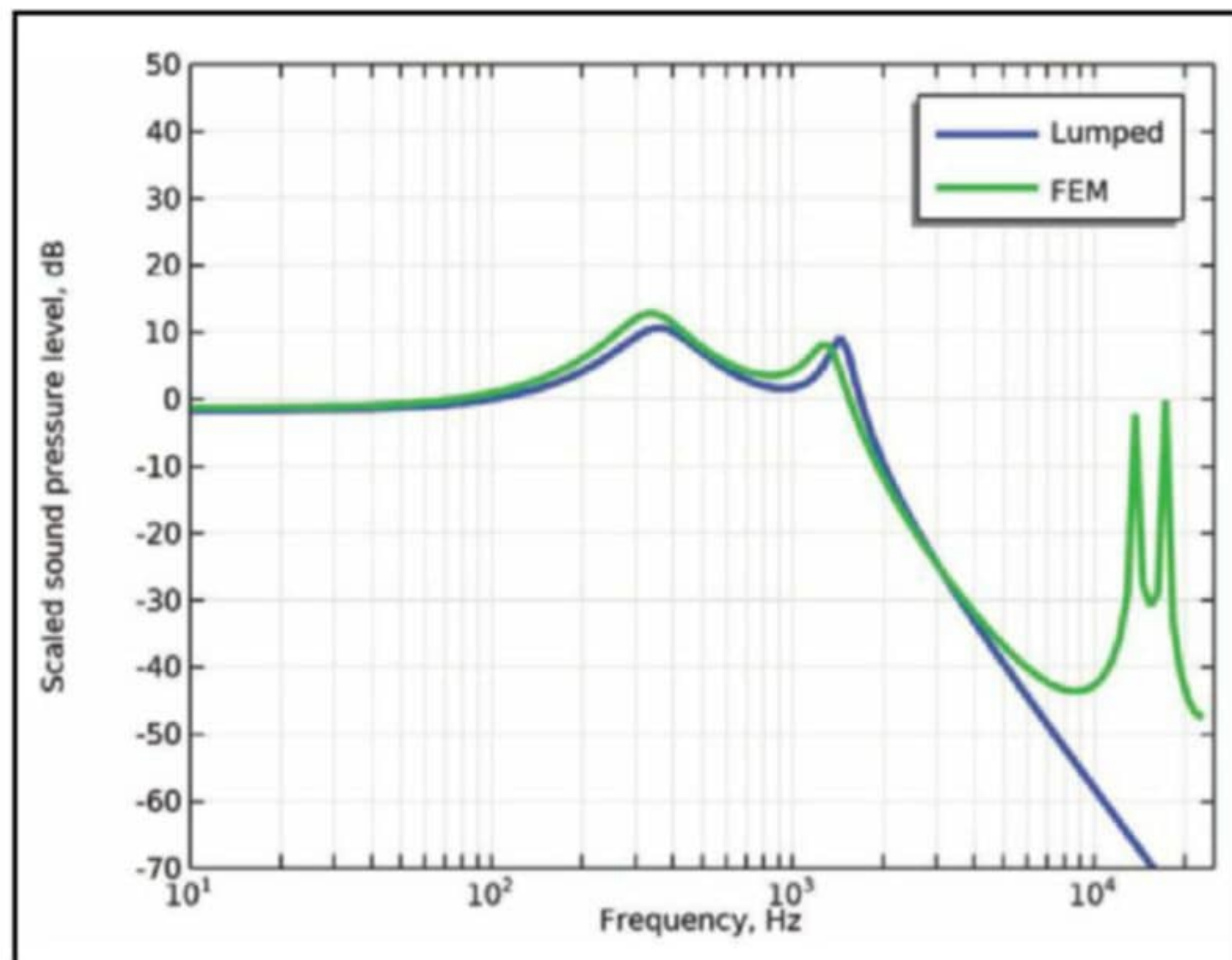


Figure 4: Comparison of sound pressure levels for the lumped model and the finite element model

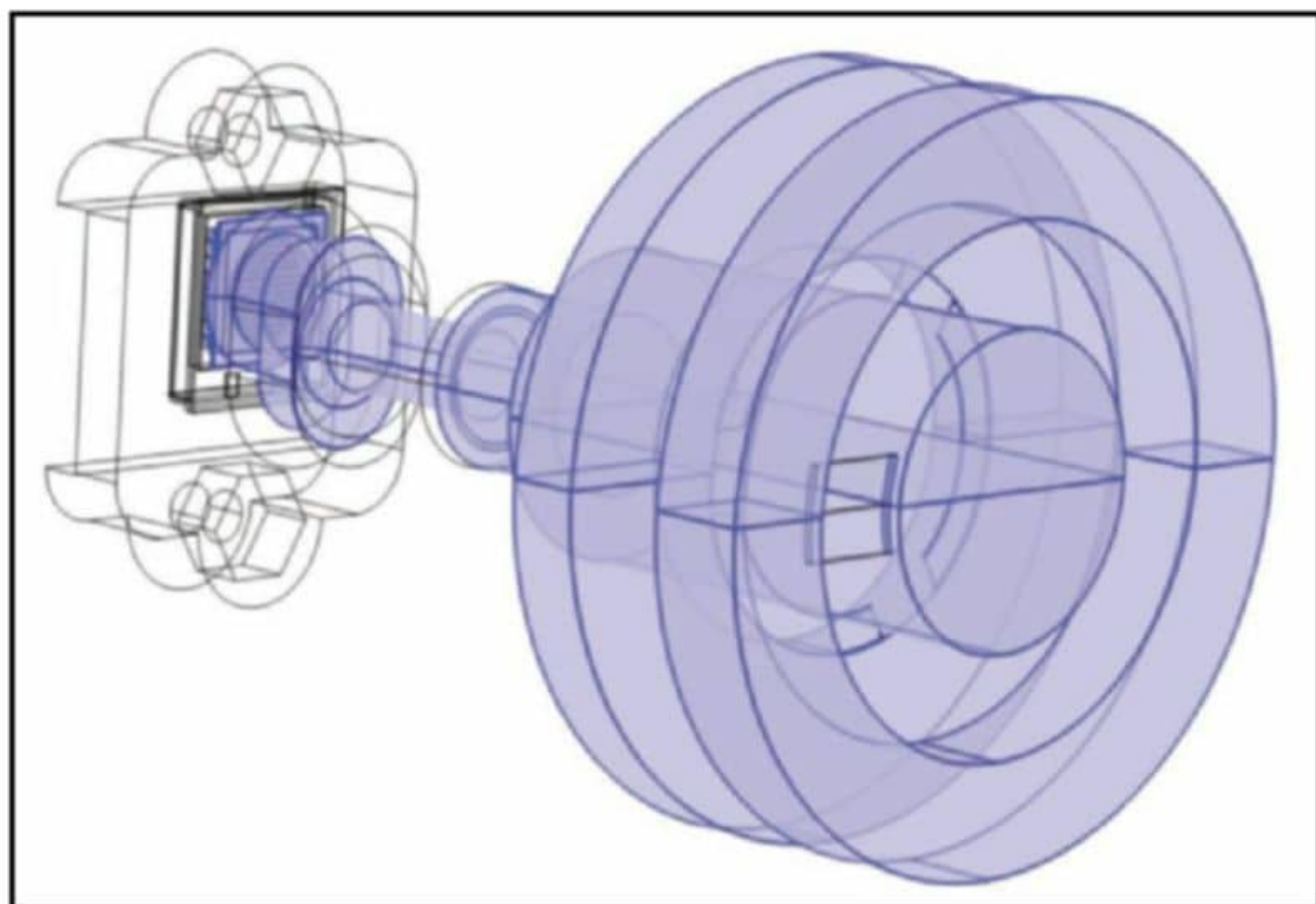


Figure 5: The SE1000 with housing is placed in a 711 coupler.

controlled with a flat displacement response. For applications where the loudspeaker is playing in a “free field,” the resulting sound pressure level will be flat above the fundamental frequency but will not produce much sound below it. Also, the transducer is affected by the said acoustic environment. For example, a rear volume behind it will increase the fundamental frequency. To model a resulting sound pressure, one must account for the two-way coupling between structural mechanics and acoustics.

The SonicEdge transducer is fundamentally different and can be modeled more simply with a current source. Since there is no internal impedance in this model, the acoustic impedance will not affect the volume velocity, which has been validated by measurements. The volume velocity, as well as the acoustic pressure downstream, will depend on the acoustic environment, but the audio design work will be quite different with this transducer, as it has no distinct fundamental frequency as opposed to other transducers that have a spring-mass type mechanical setup internally.

In a previous article [2], we looked at the modeling of an electrodynamic speaker driver and a Balanced Armature Receiver.

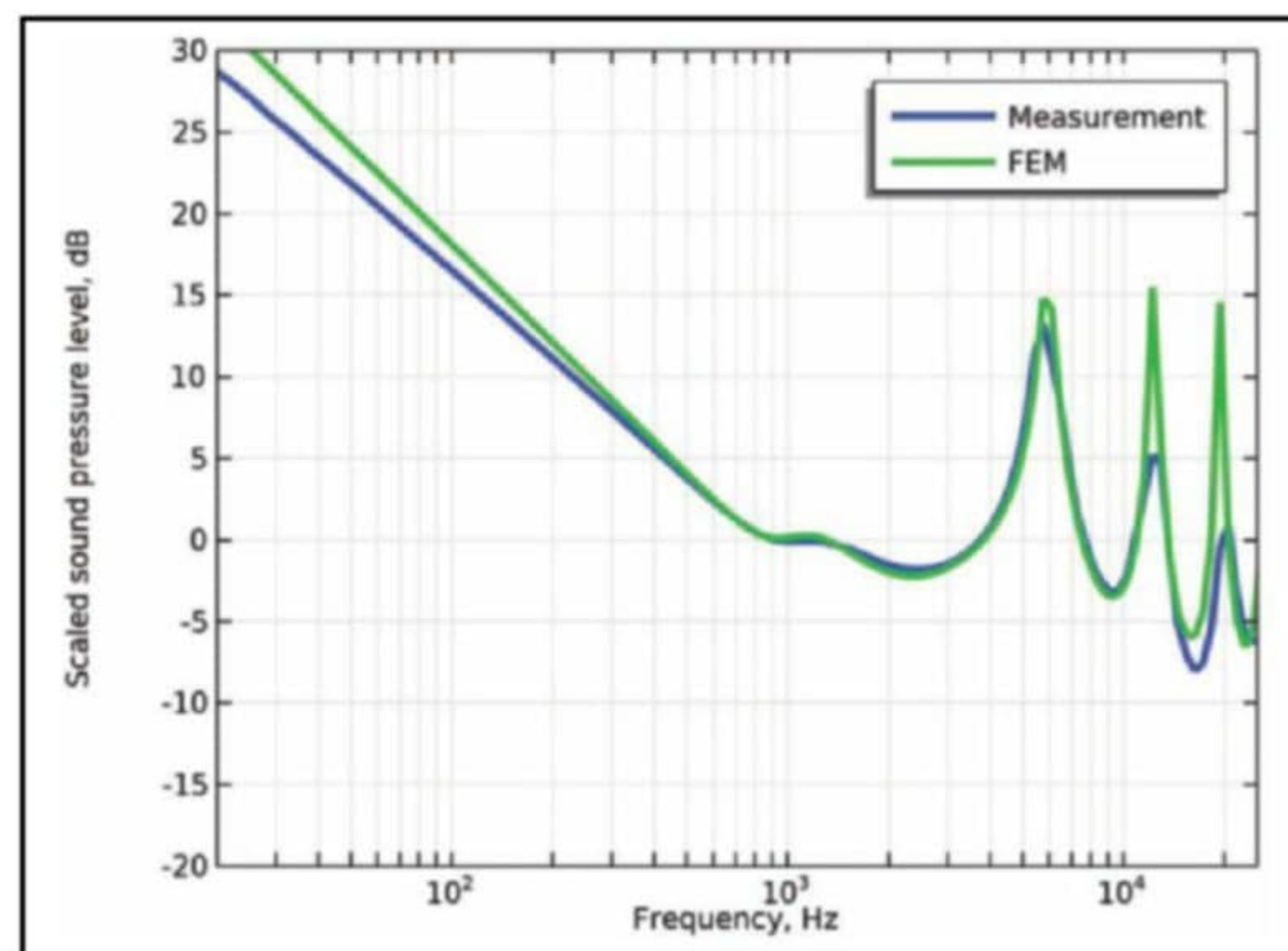


Figure 6: Comparison of measurement and simulated sound pressure levels in the 711 coupler

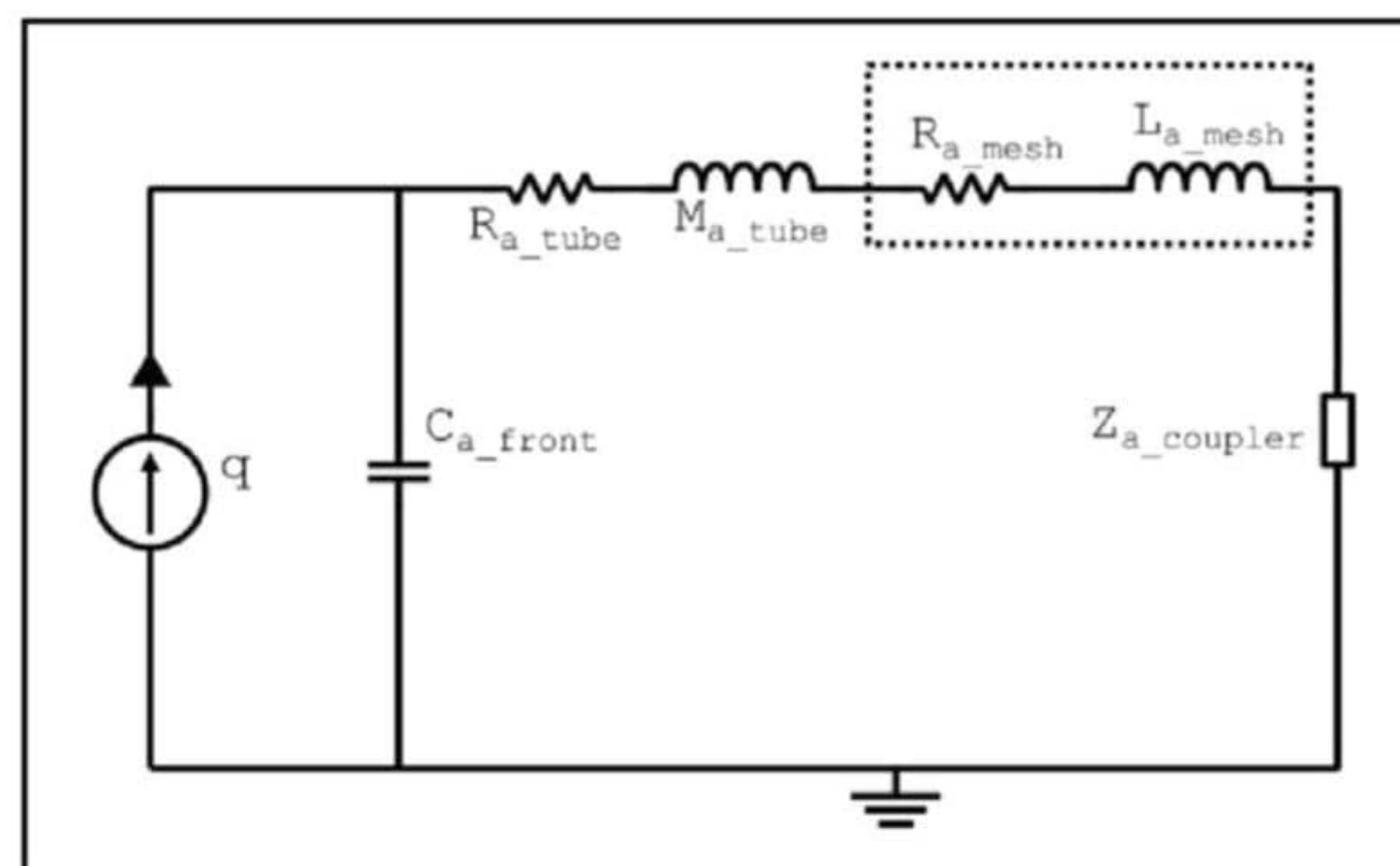


Figure 7: Lumped circuits of the SonicEdge speaker in an earphone with a resistive mesh

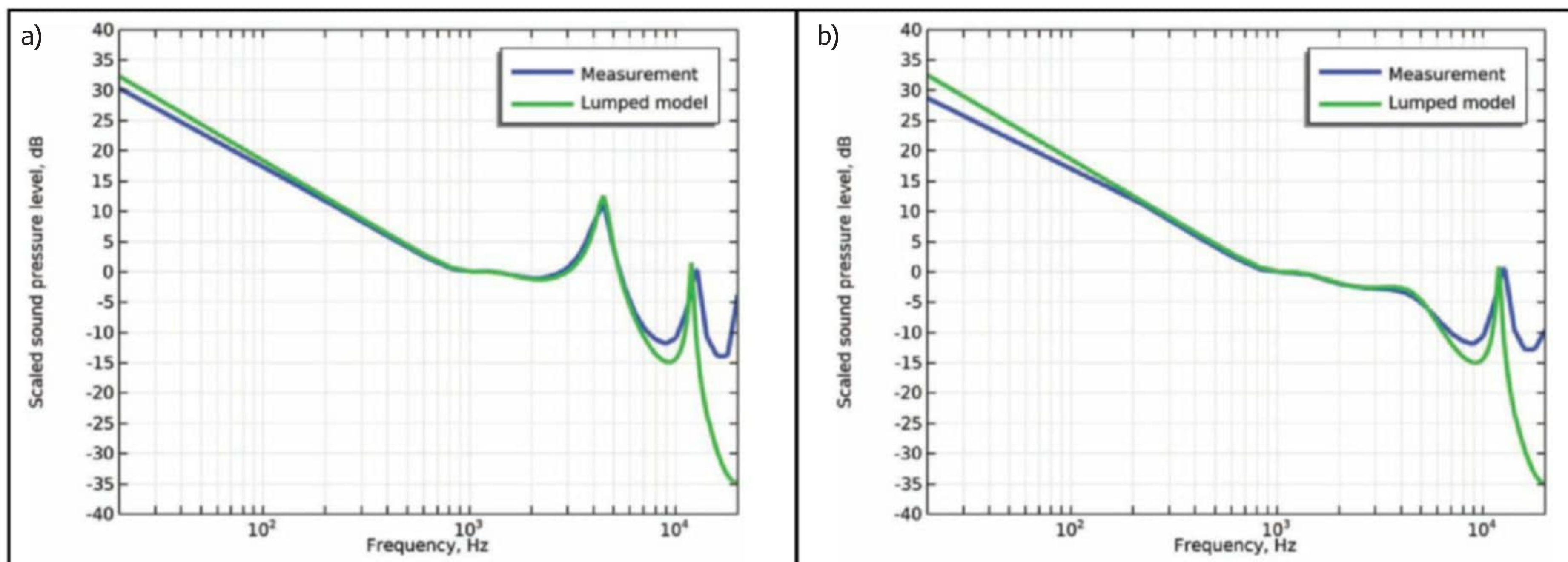


Figure 8: Comparison between measurement and lumped model of a SonicEdge speaker in an earphone without a resistive mesh (a), and with a resistive mesh (b).

In **Figure 2**, we compare the lumped element description of a SonicEdge transducer to a standard electrodynamic driver.

Examples

Two modeling approaches are demonstrated and compared for the SonicEdge transducer using the lumped and the finite modeling, respectively. Both models include the transducer as a volume velocity (or current) source. Features such as vent and leakage ports are introduced in the acoustic path to shape the sound pressure level frequency response.

Specifically, the transducer plays into a front volume and connects to a larger coupler volume via a tube with a leak whose radius and length can be varied to tune the acoustic pressure in the coupler. The lumped element model for this setup is shown in **Figure 3**, and the responses from the two modeling approaches are compared in **Figure 4**. Good agreement can be seen up to 10kHz, where the lumped element assumption is challenged.

Going toward a more realistic geometry, the SE1000 was placed in a housing and attached to a 711 Coupler, described in the international standard IEC 60318-4, as shown in **Figure 5**, where the SE1000 geometry is explicitly included, and each single cell has an associated velocity boundary condition. For these more complex geometries, it can be challenging to set up accurate lumped models, and so by using simulation software such as COMSOL Multiphysics, different housing configurations can be tried out quickly and efficiently. With the focus here being on the placement of the first resonance, the finite element can take into account acoustic losses in a simplified manner as the peak height is of less interest in the initial investigations, with the measured and simulated sound pressure levels shown in **Figure 6**.

Resistive meshes provided by SAATI have also been included in the acoustic design. The motivation of such a mesh is to dampen the resonance introduced by the housing and shape the spectrum response as desired.

In the following example, a mesh from SAATI's Acoustex product range with a specific airflow resistance of 160 MKS rayls

is mounted at the end of the housing tube. The lumped circuit in **Figure 7** illustrates this situation and compared to measurements in **Figure 8**, with and without the mesh included.

Conclusion

The novel SonicEdge transducer can deliver constant volume velocity across audio frequencies and beyond. This transducer differs from other available transducers in that it can be modeled simply as a constant volume velocity source, an "audio-pump" with negligible vibrations, alleviating certain noise-cancellation and feedback application challenges. Together with Acculution, SonicEdge demonstrated how to model the transducer in lumped-element and finite-element acoustic simulations and how to tailor a desired SPL response using acoustic features. 

About the Author

René Christensen (BSEE, MSc, PhD) has been working with simulations in the loudspeaker and hearing aid industry for several years, and in 2021 he started his own company, Acculution ApS, with a focus on mathematical modelling of many different products, such as transducers. The overall aim is to use academic research results for engineering better solutions faster, training engineers in this type of virtual prototyping, and working with students across the world to progress the field.



References

- [1] M. Margalit, "Techniques for Generating Audio Signals, Patent US8861752, October 14, 2014.
- [2] R. Christensen, "Lumped Element Modeling of Transducers," *audioXpress*, March 2023.
- [3] Health Protection Agency, "Health Effects of Exposure to Ultrasound and Infrasound," 2010.



Noise Cancellation in Automotive Audio

What's the Destination?

By
Philipp Paul Klose

This article investigates how active noise cancellation (ANC) can be beneficial in an automotive application and looks at the challenges and technical implications in the development phase. It also provides an outlook for applications, products, and industry adoption.

Acoustic noise cancellation became an ubiquitous commodity in the last few years. The advent of Bluetooth headsets and true wireless stereo (TWS) earbuds at all price points allowed a broad range of consumers to become familiar with the benefits and the terminology associated with noise cancelling. By the time I was researching for this article, headphones with ANC priced as low as \$35 were available to buy online. Is it time to implement advanced ANC into the audio systems of cars? And why is this technology not already in every car?

Technology Is the Driver

Within the automotive industry there are plenty of proverbs to describe complex relations and business decisions in an easy to grasp manner. One of them is:

"Cars are the perfect test platform for all technologies developed by humankind." (Christian von Koenigsegg)

At one point or another, all technologies were tried or have been used in a car: From fuses, synthetic rubber, MOSFETS, combustion engines, electric engines, nuclear propulsion, turbines, GPS, MEMS sensors, material compounds ... The list is literally endless! The reason for this much conducted

research and development lies in the highly innovative nature of car companies due to much competition in the markets. The attractive turnover rates and margins in the auto business plus the significance of automobiles for the macro-economic development of a country is a big motivator for constant change and experimentation. In order to make use of a technology in automotive, it must be a finished technology that fulfills hard automotive business requirements.

The idea to use anti-phased artificial electro-acoustic noise to cancel unwanted noises goes back to John William Strutt, who adopted the theorem of interference and cancellation of optic waves onto the acoustic domain in 1877. Noise cancellation implementations have since been tried on and off in different forms and functions in cars. Now, with the software-defined car coming and autonomous driving looming, the automotive audio industry is currently a very exciting business of which to be part.

Within the last few years, many enabler technologies needed for ANC integration have come to market. A2B from Analog Devices [1] allows signals to be sent cost-effectively through the cabin in a stripped-down Ethernet protocol over unshielded twisted-pair cable. Development platforms such as DSP Concepts' AudioWeaver [2] and Harman AudioworX [3] gave the power of

simplified development coupled with immediate exploration into the hands of many. Also, multi-stage-voltage amplification ICs such as the Infineon MERUS [4] technology with super-high efficiency are making jumps from year to year. All this makes the integration efforts for OEMs much easier. In the field of research, the FxLMS (filtered-x-least mean square) algorithm is currently identified as state of the art. The needed DSPs for ANC operations have improved a lot recently in available MIPS and multi-core types are now available at a reasonable cost. The integration of DSP into other SoCs as co-processors made the DSP applications in products more common.

What Does the Customer Expect and Where is the Noise Coming From?

It is fair from a customer’s point of view to expect to at least match the performance from a \$300 USD headphone to a \$100,000 USD car. ANC monsters such as the Sony WH-1000XM4 [5] deliver more than 30dB of dampening when worn. But with current technology this is very hard to achieve in the car.

While ANC was a hype topic in automotive about 10 years ago, the hype has now settled to a more down-to-earth view. Many implementations were thrown into the market back then but were often not understood or not up to the expectation of customers. The often-implemented engine harmonic cancellation was perceived not as spectacular as broadband ANC known from early headphones. Higher emissions regulations taking effect in Europe, which lead in return to quieter exhaust systems, made the EHC integrations for most cars obsolete. Discontent customers, low margins, power-hungry amplifiers, and high costs in service for OEMs lead to a quiet dissipation of ANC in cars.

“Automotive—Where weight is king and price is queen.” (unknown)

The first electric cars appeared on the horizon shortly afterward and made, with the absence of combustion engines and exhaust systems, the integration of ANC seemingly more useless. But the denial of customers for pure electric tires (high and narrow) for more conventional tires (lower and fatter) played chances back to ANC. Conventional tires get squashed more under the high mass load of EVs, creating way more tire noise than conventional cars. Other sound sources in the car stayed mostly unchanged. The overall SPL has not changed, just the perceived loudness and composition of the noise in and outside of the car. During the trial

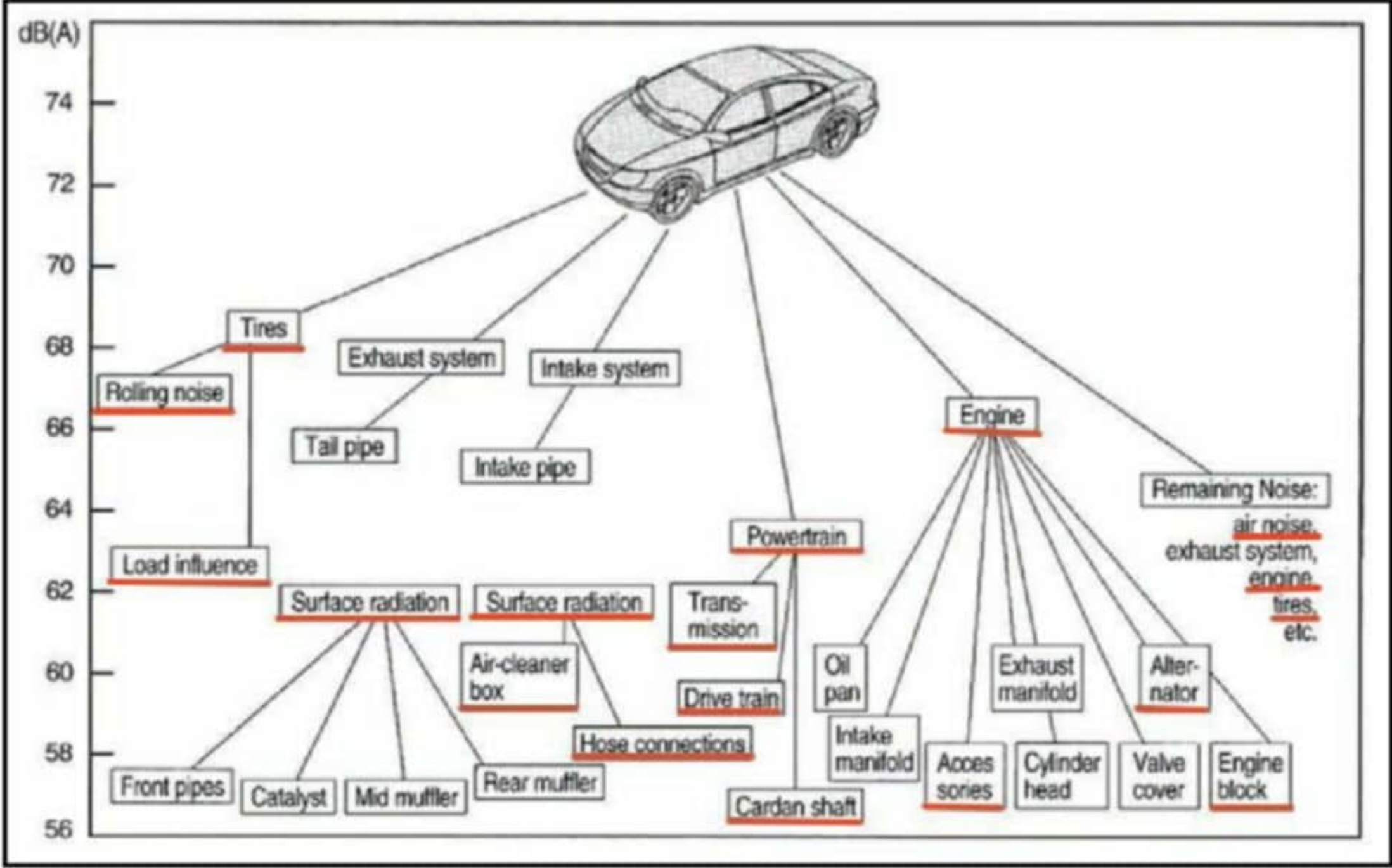


Figure 1: Noise sources and their levels in Internal Combustion Engine (ICE) cars. The red underlined sources are relevant for EVs too (Image source: Robert Bosch GmbH, Automotive Handbook. 6th ed., SAE, 2004).

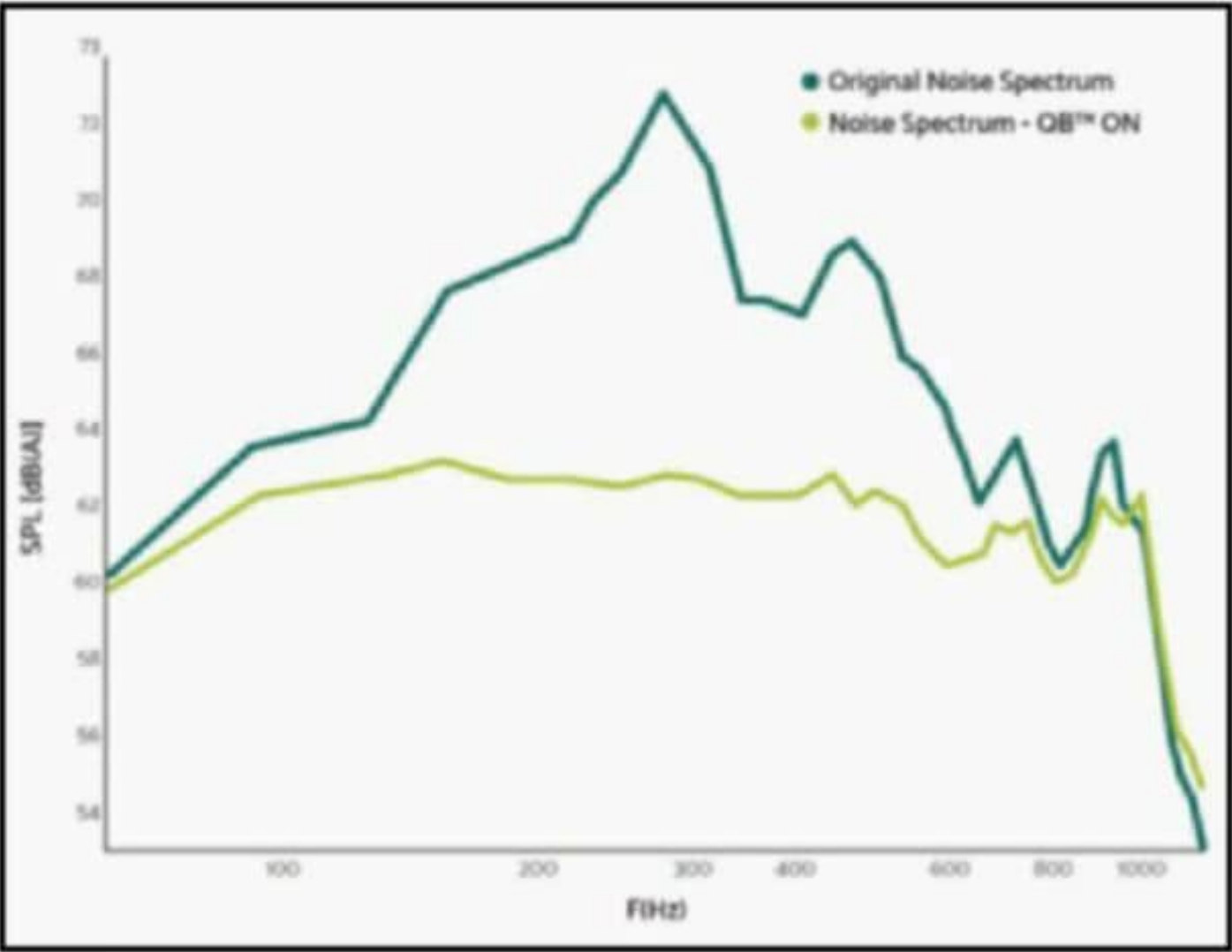


Figure 2: Potential noise dampening performance of a Silentium ANC system. Note that the frequency range goes to 1kHz. (Image source: www.silentium.com/automotive-case-study-2).

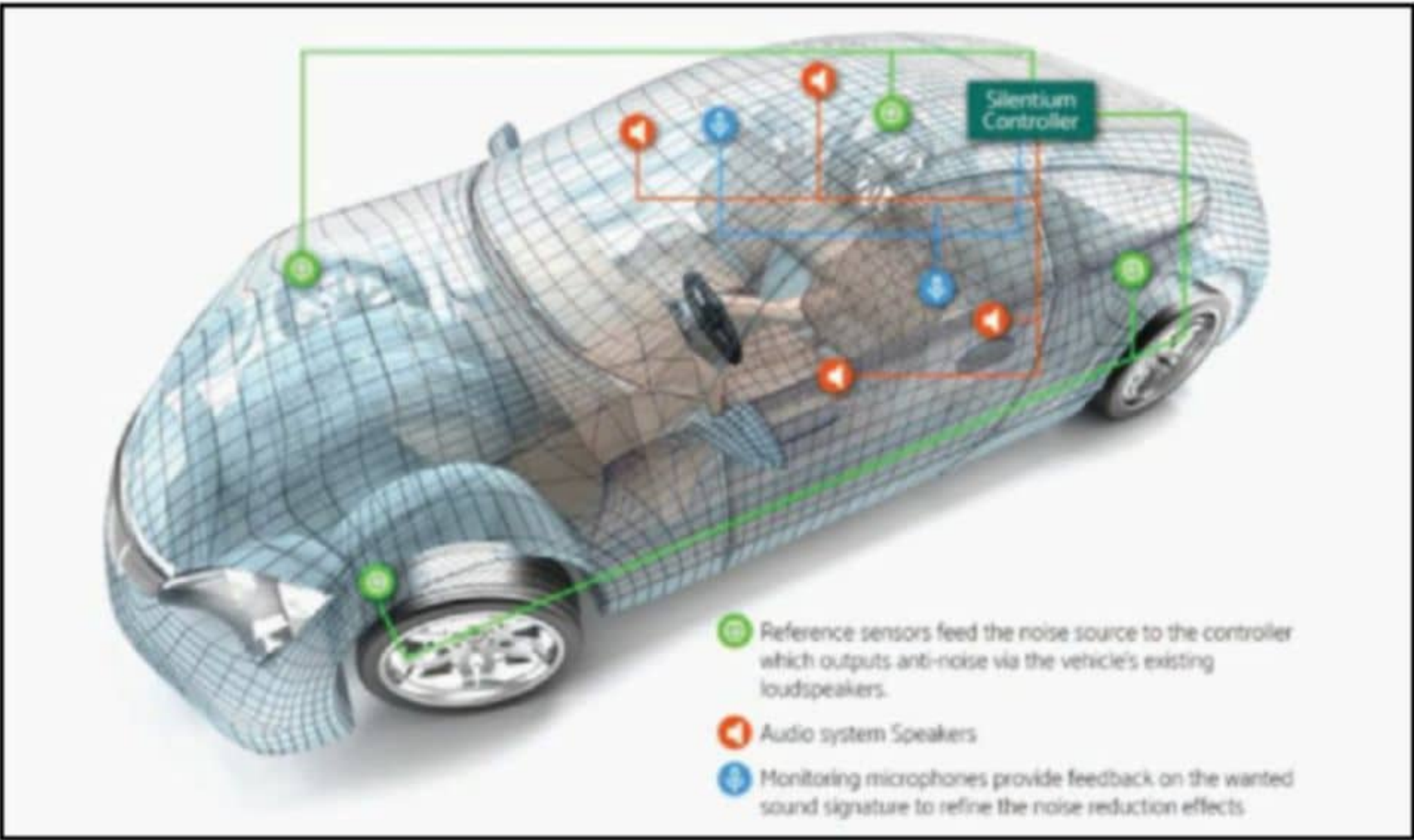


Figure 3: System design for an ANC system from Silentium. The sensors on the wheels are accelerometers. Mixed with the error signal from the microphones in the interior, the ANC signal is generated for the four speakers in the door. (Image source: www.silentium.com/automotive-case-study-2).

about the death date of Internal Combustion Engine (ICE) cars, many new players emerged, and older players embraced the concept of the software-defined car. Bose [6], Silentium [7], Arkamys [8], QNX [9] and others introduced their own approaches and solutions for ANC, which often are not systems anymore (includes everything needed), but rather parts of a system like software or sensor processing technology, therefore catering to the modular platform development needs of OEMs better.

Solution A: Usage of In-Car-Speakers

The straightforward approach to implementing a wide-band ANC system into a car is to use the already built-in audio system: The speakers are there, we have DSP, we know where the driver and passengers are located, and we have amplification. Let's just add some sensors, take some signals from the car's CAN buses, mix the different signals right before the amplification stage and we are good to go. The added cost can be countered by removing sound-deadening materials that we no longer need. What sounds like a perfect plan for a test car is unfortunately not so easy in the world of series automotive production. Several challenges appear very soon and make a good result almost impossible to reach. So, what is the problem?

First, we must cater for different seating positions. Although most cars are driven by only one person (exact numbers vary around the world), ANC is an important feature for the comfort of

passengers, since noise is perceived differently if you are in an active or passive role in a machine interaction. While drivers process the provided sound feedback impression for their driving decisions, passengers do not "use" the noise and find it therefore disturbing. All seating positions must have good performance. But the overall sound performance through speakers is different on every seat and the reproduction of a coherent sound field requires plenty of calibration time.

Compared to ANC headphones, in the car cabin the distance between the transducers and ears is longer. That makes the controlled injection of mid and high frequency anti-phased sound difficult due to the short wavelengths that are needed here. UWB sensors, radar for presence detection and infrared head position sensors can help here with high-frequency fine adjustments for ANC but are far from being standard soon and very likely only deployed in the front row. Headrest speakers influence the sound field closer to the ear of the customer, but their deployment is mostly limited to higher end cars and is only used in front rows on captain's chairs. Bench seats with headrest speakers are not available yet.

When the customer is moving, strange phasing patterns can appear. But they have a profound base for covering a wide frequency band. The needed hardware integration of error microphones, accelerometers at the axles and wheel struts seem like a small job, but this is cost intensive. Since many car platforms and kits are not prepared for these kinds of sensors in the first place (as of now), the sourcing, implementation and design changes come with costs that are hard to justify in the product development. To give an impression about the cost of such an effort: The integration of a simple part number into a production system of an auto maker costs between \$20,000 and \$60,000 USD! The development of a new motor for an electric window takes seven years due to all the testing necessary. Since standard loudspeakers from car audio systems do not play low enough or do not accept continuous high-power demands well, larger premium speakers and subwoofers must be used to get a good basis for acoustic performance.

Another argument for the integration of ANC is that sound-proofing materials can be saved. This is in a lot of cases not feasible, since the sound proofing materials (e.g., basotect, foams, carpets, mats...) also have other integrated functions. They provide spacing for parts, carry wires, or keep parts together or work as mass-spring system. The expected savings are minuscule and come with high efforts in engineering, since a lot of decisions must be made, and alternatives must be found. A

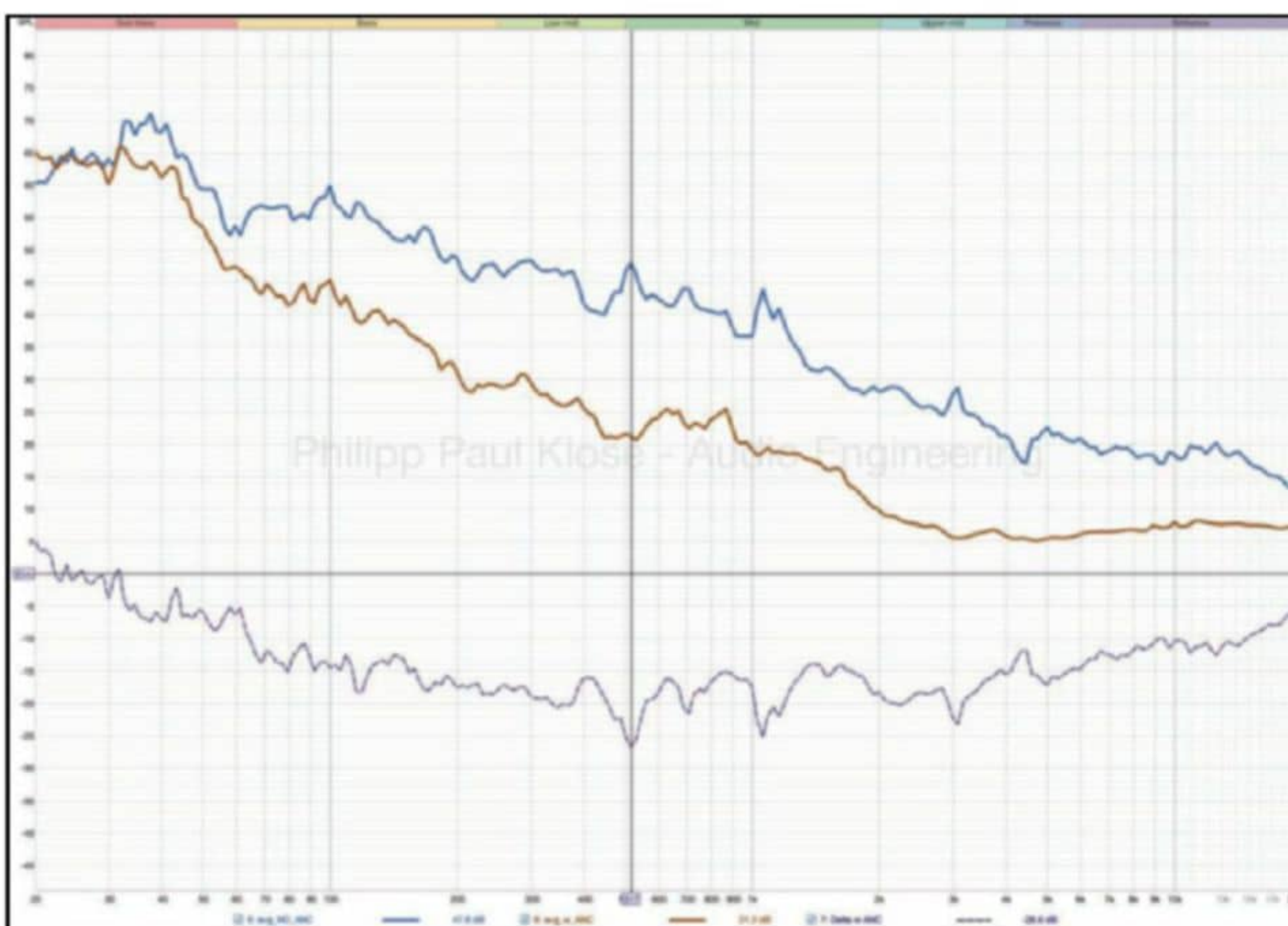


Figure 4: Noise analysis of an Audi e-tron 2022 accelerating from 50kph to 100kph in blue. Brown is the same noise recorded through a BOSE QC35 II over-ear headphone. The dampening factor (dashed blue) is in the negative and reaches up to 26dB at 520Hz! The ANC algorithm works up to ~1.9kHz, the dampening above that is realized through the acoustic dampening design of the headphone itself and not software.

potential failure mode in the ANC system (e.g., due to a missing signal or faulty sensor) could then lead to an unhappy customer and a premature visit to the workshop. Within the software domain the work continues.

Many cars are using DSP nowadays to manage signals and correct for speaker deficiencies. Although the MIPS needed for an ANC algorithm are relatively low, it will be hard to find a DSP in the field that is overpowered and has overhead for additional tasks. Due to the mix stage necessary in signal flow, a massive SW change is to be implemented here. An additional DSP requires re-tooling and re-certification, the prices of which is in the millions. In the HMI a good set of features and descriptors must be created for the user. Due to the several hundred features already available in infotainment systems today, it is wise to make a good standard selection of options and hide away as many options as possible to the user. The function and layout of options must be easy to grasp.

From the business side the question arises, on what kind of business model ANC is integrated into the car. Should it be standard? Then the car companies cannot charge extra for it and might



Photo 1: Simple, easy, and relatively cheap: The internationally valid sign of “please do not disturb” is the use of headphones. The dampening factor is comparable to speaker-based ANC systems. The usability can be easily understood by every passenger in the car and the development of the headphones can be de-coupled from automotive platform development cycles. The high dampening factor in the complete audio spectrum will appeal to OEMs and customers alike.

EXPERIENCE THE SILENCE

Elevate Your Soundproofing

Acoustic Geometry's world-class

SOUNDPROOF WINDOWS

Tailor-engineered for utmost performance and style, our windows blend elegance with affordability, setting new standards in noise control solutions.

Why Choose Acoustic Geometry?

- **Affordability Meets Excellence:** We've performance-tested our windows to ensure top-quality noise control.
- **STC Ratings You Can Trust:** from 45 to 59, we offer windows tailored to your specific needs.
- **Customization at Its Best:** custom-sized windows with a choice of anodized aluminum frames, available in standard and specialty finishes.

- **Quick Delivery and Easy Installation:** Fast delivery and provide easy-to-follow installation instructions.
- **Varied Glass Options:** Choose from safety glass, mirror pane, tinted glass, and more to customize your window's look and feel.
- **Safety Ensured:** We ship our glass separately to prevent any breakage, ensuring your product reaches you in pristine condition.

ACOUSTIC GEOMETRY
BY ACOUSTICAL SURFACES

www.acousticgeometry.com
123 Columbia Ct N, Chaska, MN 55318
Email: sales@acousticgeometry.com
tel. 1.888.227.6645

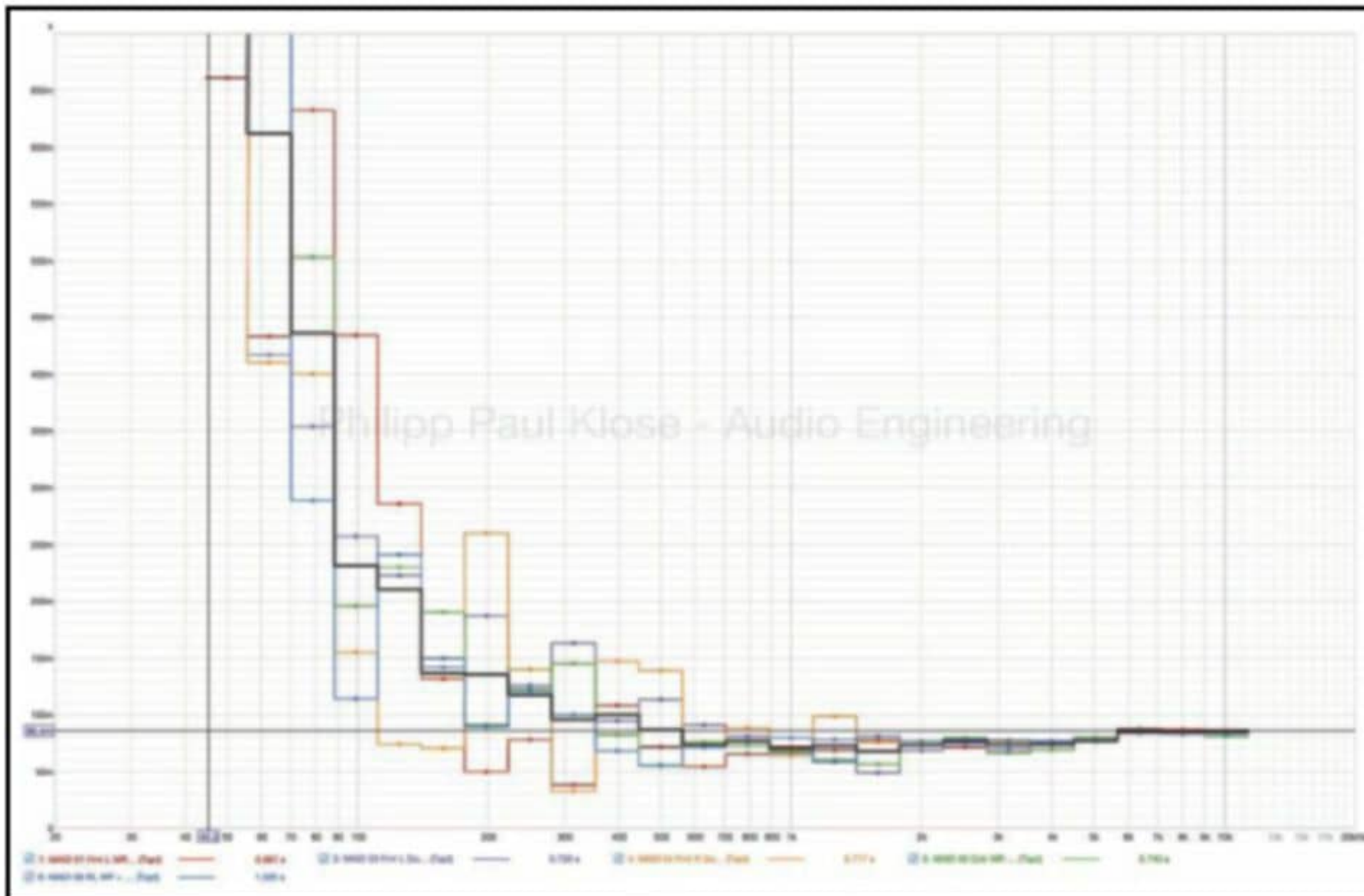


Figure 5: RT60 reverberation times measurement in a car cabin: The overall decay is super short with an average of around 90ms.

References

- [1] "A2B Audio Bus: An Easier, Simpler Solution for Audio Designs," Analog Devices, www.analog.com/en/applications/technology/a2b-audio-bus.html
- [2] DSP Concepts, <https://w.dspconcepts.com>
- [3] AudioworX, Harman Automotive <https://car.harman.com/solutions/car-audio/audioworx>
- [4] "Class D Audio Amplifier IC," Infineon www.infineon.com/cms/en/product/power/class-d-audio-amplifier-ic
- [5] "Sound," RTINGS.com www.rtings.com/headphones/reviews/sony/wh-1000xm4-wireless#test_321
- [6] "The Power of Sound," Bose Automotive, <https://automotive.bose.com>
- [7] Silentium, www.silentium.com
- [8] ARKAMYS, www.arkamys.com
- [9] "QNX Acoustics for Active Noise Control," QNX www.qnx.com/products/anc/index.html
- [10] "Survey: Consumers Hit the Brakes on Subscription-Based Features," Autolist, February 27, 2023 www.autolist.com/news-and-analysis/2023-survey-subscription-features
- [11] J. Vincent, "BMW starts selling heated seat subscriptions for \$18 a month," The Verge, July 12, 2022, www.theverge.com/2022/7/12/23204950/bmw-subscriptions-microtransactions-heated-seats-feature
- [12] Auracast, www.bluetooth.com/auracast
- [13] E-Cooustic Systems, <http://ecoousticsystems.com>
- [14] U. Muesch and F. Kettler, "Instrumental Testing of In-Car Communication Systems," HEAD acoustics https://pub.dega-akustik.de/DAGA_2015/data/articles/000270.pdf
- [15] Asahi Kasei, <https://automotive.asahi-kasei.eu/de/start>
- [16] Noveto, www.noveto.com
- [17] "Jazz Club Sound Inside a Volvo with Bowers & Wilkins Speakers and Dirac Processing," *audioXpress*, December 29, 2020 <https://audioxpress.com/news/jazz-club-sound-inside-a-volvo-with-bowers-wilkins-speakers-and-dirac-processing>

have to compensate system cost through a higher base MSRP. Options that are activated through OTA have a hard standing [10] and currently see continuous backlash [11]. Be aware that buyers and leasers of cars are older than the average person, even more so with luxury ANC cars. An optional extra on the other hand could not be interesting enough for many users, if they do not understand the feature in the shopping process or the salesperson does not see the point (often a fate of cheaper premium sound systems in cars).

If ANC is an optional extra, the potential savings through the omission of acoustic dampening material savings are completely out of the door. There are some tough decisions to make here! Although this is a technical magazine and I am an audio engineer, the usability and social factors of products is an important aspect for me when I am involved in a project. What is the use of technology when it is not helping people? The concern I have here is, that cars, although they are often used by only one person, are still social places and will become even more social places when autonomous shuttle services are getting more popular.

I currently see a potential for ANC in the car hindering people communicating with each other and building up technical barriers in usage and communication through important blocked sounds, missed signals and therefore creating frustration. With voice recognition technology in the car, this barrier can already be experienced. Although voice recognition is a brilliant and easy way to interact with a car in traffic, when I have passengers in the car, they always find it disturbing when I am using the voice recognition. The feedback is, that it is hard to discern if I am talking with the car or them (due to context unawareness and driving noise) and the pausing of media playback from the infotainment in the "listening and thinking phase" of the voice recognition system is disturbing. Implemented in such an unpleasant way, the technology is not helping with my real needs. I only use voice recognition when I am alone in the car.

It is obvious that the speaker-based approach is viable, but it comes with high development costs and wide-reaching implications. Time is needed to iron out trade-offs with several neighboring function owners and colliding parts. Since the forecast of the real benefit of the integrated ANC system is hard to predict, car companies are certainly eyeing it, but they are shy of emerging new profound solutions. My guess is that we will see this kind of approach in the higher end of the market where the additional parts and engineering costs can be more easily transferred to the customer, like executive sedans,

SUVs and high-end trucks. I do not expect that this technology will trickle down over a foreseeable time into lower segments.

Solution B: Headphones

While the speaker-based ANC approach is hard, an easier approach is the application of headphones in the car. While it seems almost absurdly simple, the pros for this kind of implementation are manifold. First, headphones are accepted nowadays by many customers. Their look and feeling is familiar. Why not use that? Their fast development cycle can be decoupled from the car’s development cycle. But the real potential lies within connectivity.

Within the headphones and their rising processing power, many business cases are concealed. OTC hearing aids are a well-known and important use case gaining traction, but in the car, we could enable broadband ANC and powerful in-car-communication, video-conferencing services, personal assistants and have a direct interface for health applications. Auracast [12] could make the perfect sound-zone available for the passenger and send car-relevant auditory display information like GPS prompts or speed callouts (regulatory from 2025). Besides the factor of headphones being easy-to-use and understand, the social factor of headphones is better than with audio system-based ANC. If you want to talk to each other, just take off your headphones. It is not necessary to switch off your sound bubble through a menu first. If someone wears headphones, it means “do not disturb.” Easy. The potential for user frustration is way lower than with current speaker-based ANC approaches.

Gear Up: Virtual Acoustics

In close relation to cleaning the acoustic space through ANC, is also the enhancement of the acoustics with active acoustics. As it is often not possible to eliminate noises, auto NVH sound designers spend a lot of time shaping and influencing occurring sound events to make them more pleasurable. Think here of measures comparable to installing an electric water fountain in the garden to mask noise from cars on the street. In concert halls the approach of enhancing reverberated and amplifying direct sound has been applied successfully and very discreetly for a long time now, for example by the company E-Coustic [13]. Maybe it is time to have an interdisciplinary exchange here?

People mover vehicles with five or more seats have gained a lot of traction in Asian and North American markets recently, which makes the application of In-Car-Communications (ICC) systems a worthwhile investment. The current

(+) is a high or a lot, (-) is low or less	In-Car Speaker	Headphones
Integration Effort	+++	--
Development Cost	+++	--
Software Changes	+++	+
Platform/Kit Changes	++	-
End User Price	+	--

Table 1: It seems that headphones could become the most popular option by sales and total numbers for noise cancellation in the car.

implementations of these systems improve intelligibility but are still lacking performance at Autobahn speeds [14]. A combination of ICC with ANC system is an interesting coalition, improving the overall perception of noise through the creation of a cozier acoustic environment. Within the car cabin the reverberation times are very low, leading to irritating locating effects known from over ear headphones. The low reverberation times in cabins is being tackled by Asahi Kasei [15] through the application of reverb algorithms and Noveto [16] through head tracking and an algorithmic enhancement approach. Since head movement tracking systems for drivers are a must for SAE Level2+ ADAS, it is a logical idea to re-use the sensor signal, if the platform/kit of car permits to do so. On the same note, Volvo is acting here. The current infotainment system by Harman has implemented the impulse response of the Gothenburg concert hall [17], bringing the listener into a completely different environment.

Conclusion

While the hype around ANC in automotive has settled a bit, I see high potential in the future to make customers happy with quiet cars and enable new relaxing means of transportation. Many important enabling technologies are paving the way for meaningful products.

In my opinion, headphones will become overall the most popular option by sales and total numbers for noise cancellation in the car. The headphones solution will be the only option for cheaper cars and a mix of headphones and speaker-based solutions will be found in the middle and higher ends of the market. But I am happy to be surprised here! The software component and business cases associated with hearables will be available through unified software platforms in all price classes. The mix of both headphones and speakers for ANC will dampen the noise perception by an amount that can be perceived by every user and will make much more journeys enjoyable. 🎧

About the Author

Philipp Paul Klose is a senior engineer for automotive audio concepts and functions at CARIAD SE and previously served as an audio system development engineer and a project lead and infotainment test engineer. He holds a Bachelor of Arts (Hons.) degree in Audio Production. Within his 10 years of automotive experience, he has worked for customers like Audi, Volkswagen, Porsche, Lamborghini, and Ferrari.



Designing a True Midrange Driver

By
Jørn Rune Kviserud
(Tonalab of Norway)



The mid frequencies in a multi-way speaker are often handled by a woofer. This is the case in 2-way, 2.5-way, and even some 3-way speakers. The sound quality will immediately improve once the low end is removed from the midrange driver. However, there are also many things that can be done to the driver to improve its properties as a midrange driver.

There are numerous ways to create a good sounding driver. Many choices come with a set of advantages and disadvantages. One widely embraced method is where a driver is technically almost identical to another driver from the same manufacturer but is sold for a completely different application. This is often the case for midrange drivers. Not many midrange drivers on the market are a result of a development project targeted at making the best possible midrange driver. Almost all midrange drivers are lightly modified woofers. Often just a shorter coil and a surround with slightly higher loss separates them from the woofer version. In some cases, their motor is identical to the woofer motor. On paper, only considering the frequency response, this may seem like a good idea, but there are some drawbacks that are generally not considered during development.

The idea of re-using components to make variants of the same driver is appealing from a logistics point of view. But sometimes there are other factors that make the development head in this particular direction. It can be as simple as “this woofer is already popular as a midrange,” or “this is what our competitors do.” But it could be that some engineers are simply not aware of the factors that affect midrange performance, or even where to look. When told that two drivers do in fact perform very differently, some may argue that it is not backed up by facts or datasheets.

Here, we look at how one can trade the properties that make a woofer usable as a woofer and instead end up with a more purpose-built midrange driver.

The 10-Point Focus List

I would like to introduce a 10-point list for development of midrange drivers. Many products on the market have met several of these points already, but it is important to note that many of them only work well when combined with other points on the list. You can easily achieve an exceptionally smooth response while at the same time having very high losses. You can also achieve very low inductance at 1kHz, while at the same time having far too much inductive coupling between the coil and the motor at lower frequencies. And you can use a cone that weighs close to nothing, only to find that it is not capable of handling the mass of the coil and the suspension, let alone keep resonances in its working range under control.

For the sake of improving midrange sound quality, this list of crucial parameters requires attention. As mentioned, the individual points might be misleading or not useful on their own, it is the combination that makes it work.

1: The driver should not have any non-subtle characteristics within its working range.

This is probably the most intuitive point on the list, but there are a handful of things to consider. First, we should look at the raw unsmoothed frequency response. It should generally be smooth without sudden shifts. If we investigate the impedance response, that should also be smooth. Very often, the impedance curve contains clues to small resonances that may be quite audible, and it

is often overlooked. Also, investigate the driver in the time domain, and look at the distortion to see if anything stands out. The goal is to make sure the driver does not apply a modal signature to the sound that could highlight some frequencies.

Siegfried Linkwitz wrote a brilliant chapter on energy storage in cones many years ago. His website and the article are still online [1].

Please note in **Figure 1** and **Figure 2** how a seemingly “safe” range around 2kHz is extremely modal when this metal cone is viewed in the time domain. It is also interesting to see how heavily distorted the waveform is when the signal matches this mode.

2: The coil should be seriously light.

The diaphragm is a part that gets lots of attention, and with good reason. It must move very fast and transfer significant forces without losing its shape. When testing diaphragms, stiffness, mass, and shape are obvious factors that affect its modal behavior. But as soon as we attach a surround, a spider, and a coil, everything changes. The cone must carry the moving mass of both the coil and the suspension. The masses work against the compliance of the cone.

While the suspension is not acted upon by any other forces, the coil is also acted upon by the motor force. This means that the effect of the coil mass depends both on the phase relationship between the cone and the signal, and the coupling to the amplifier. The phase is mainly distorted by modes in the cone or in the suspension, leading to an exaggeration of already existing modes. The coupling to the amplifier, for a midrange driver, typically has at least one significant resistor to match its sensitivity to the woofer. This means it can be hard to determine how well the driver in question behaves just based on the frequency response measured with the driver directly connected to an amplifier.

We can see in **Figure 3** the exaggerated modes on drivers that are made in almost identical two- and four-layer versions, as with the SEAS Extreme woofers. The four-layer version (shown in **Figure 4**) has significantly more mass than the two-layer version.

Using 10” high excursion subwoofers may come across as a bad example for midrange drivers. However, it is the same applied physics as with a midrange driver, only far easier to view directly on the frequency response. When working with midrange drivers it can be difficult to spot something that is clearly audible.

The goal of keeping the coil mass low is to keep the cone mass and stiffness as dominant

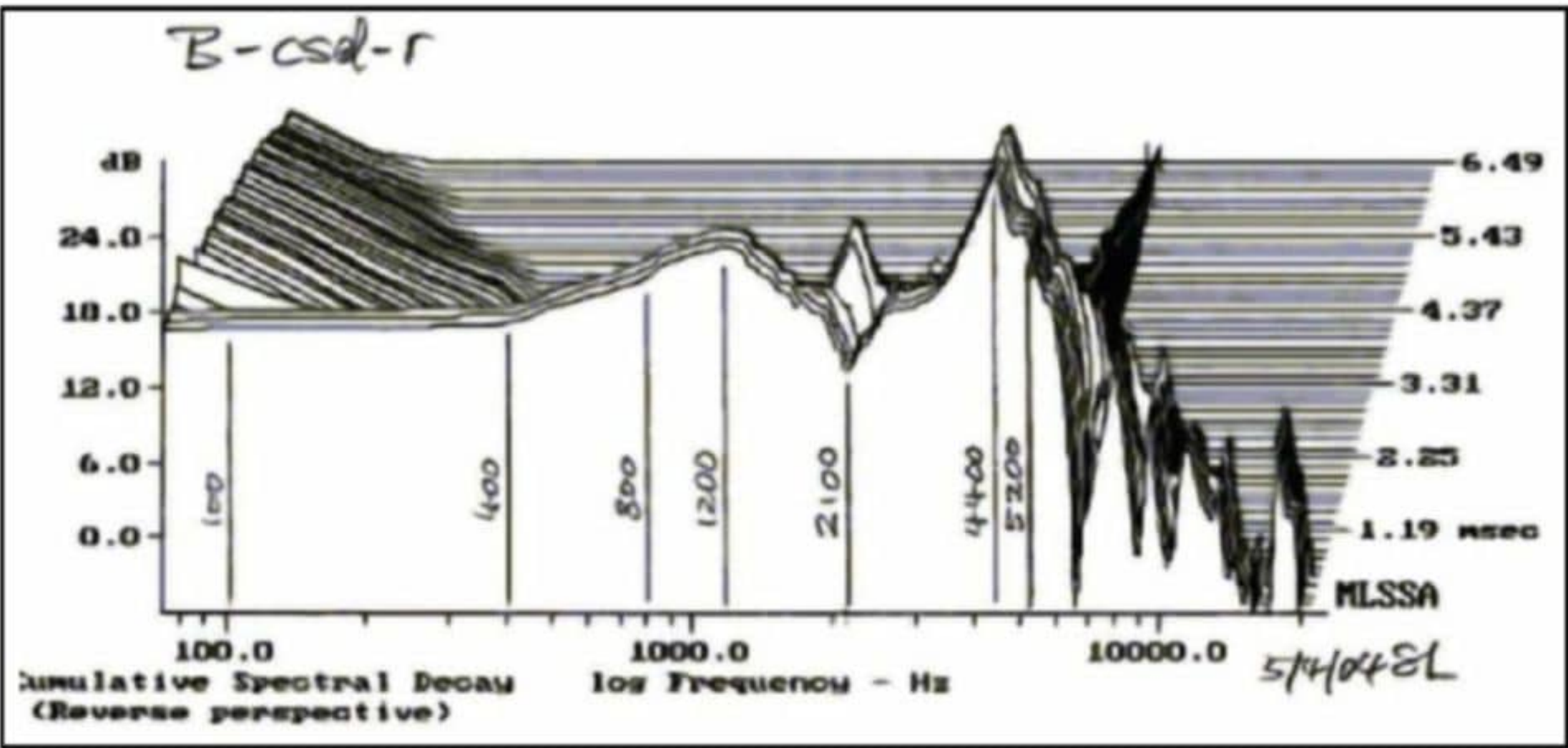


Figure 1: Backward cumulative spectral decay (CSD) for an 8” aluminum midwoofer (Image source: The Linkwitz Lab website, used with permission)

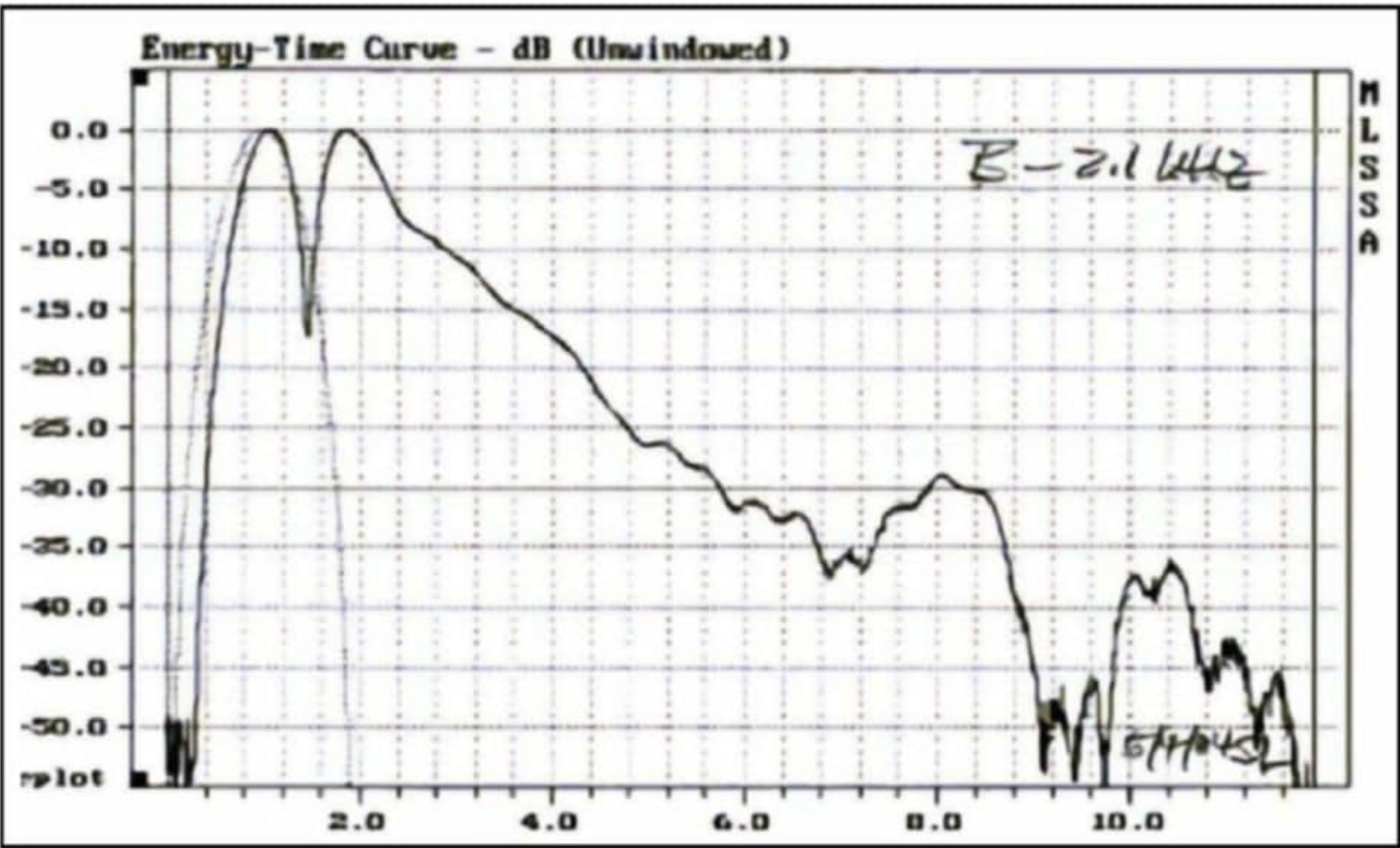


Figure 2: Energy Time Curve (ETC) for an 8” aluminum midwoofer (Image source: The Linkwitz Lab website, used with permission)

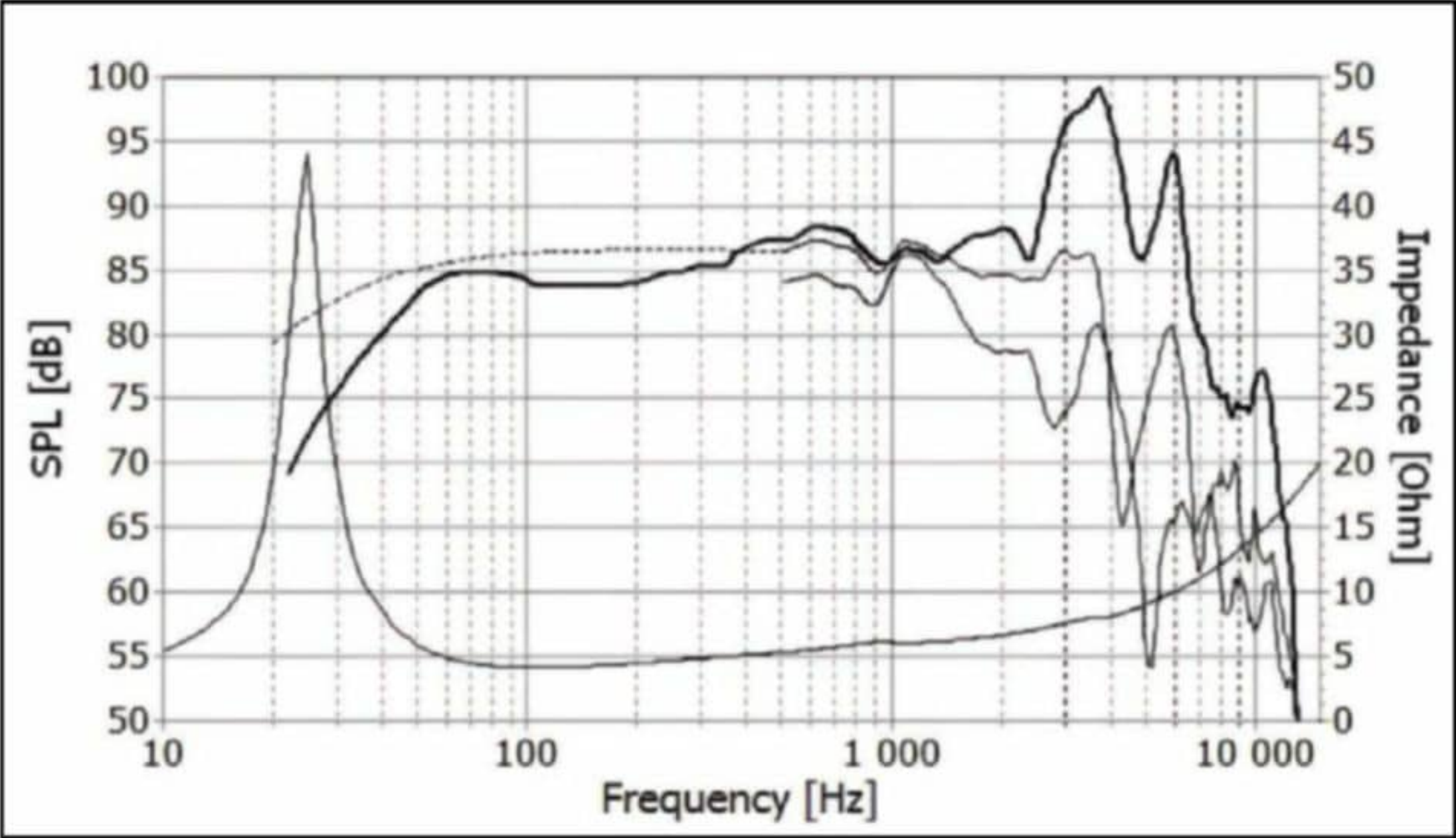


Figure 3: The Seas L26R0Y with a two-layer voice coil (Image source: The SEAS website, used with permission)

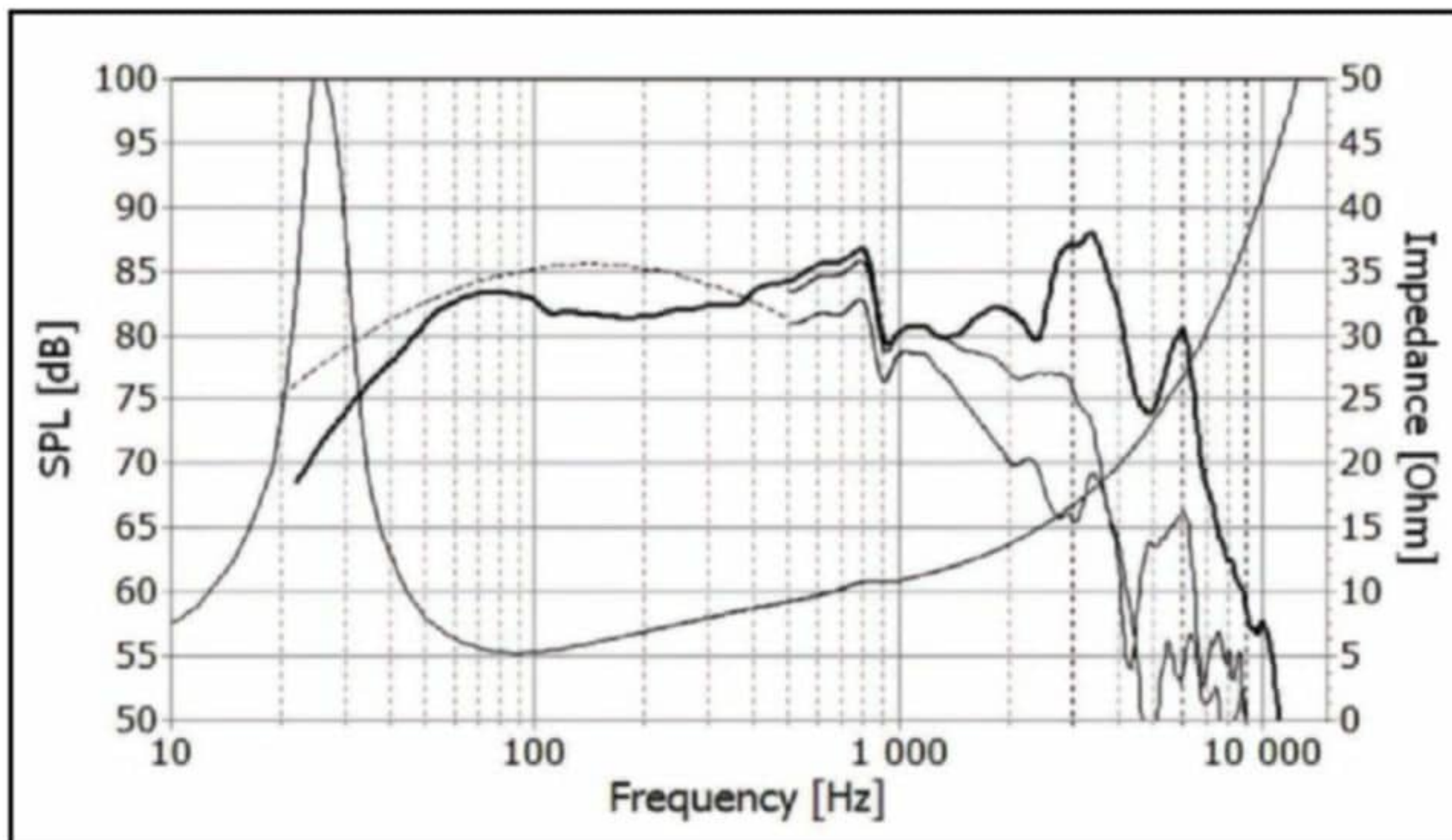


Figure 4: The Seas L26R04Y with a four-layer voice coil (Image source: The SEAS website, used with permission)

parameters. For a small midrange the coil mass can be optimized to less than 1g. In a slightly larger, 6.5" to 8", between 1g and 2g could be optimal, but that depends on the coil diameter as well. There is little reason to aim for a 3" voice coil in a midrange driver if the efficiency is not that low. It often just leads to a beamy driver with more unpredictable high frequency response.

3: The driver should have low mass, low loss suspension.

As previously described, the moving side of the surround is only acted upon by the forces of the cone. If the surround is heavy, it will decrease the integrity of the cone. A wide surround that is made light will be a very thin membrane that both tends to resonate very easily while at the same time representing a large radiating area adding to the total output. This addition can be distorted and unwanted. If we make the surround very narrow, we will get a surround that has naturally higher damping. It can be challenging to keep the damping

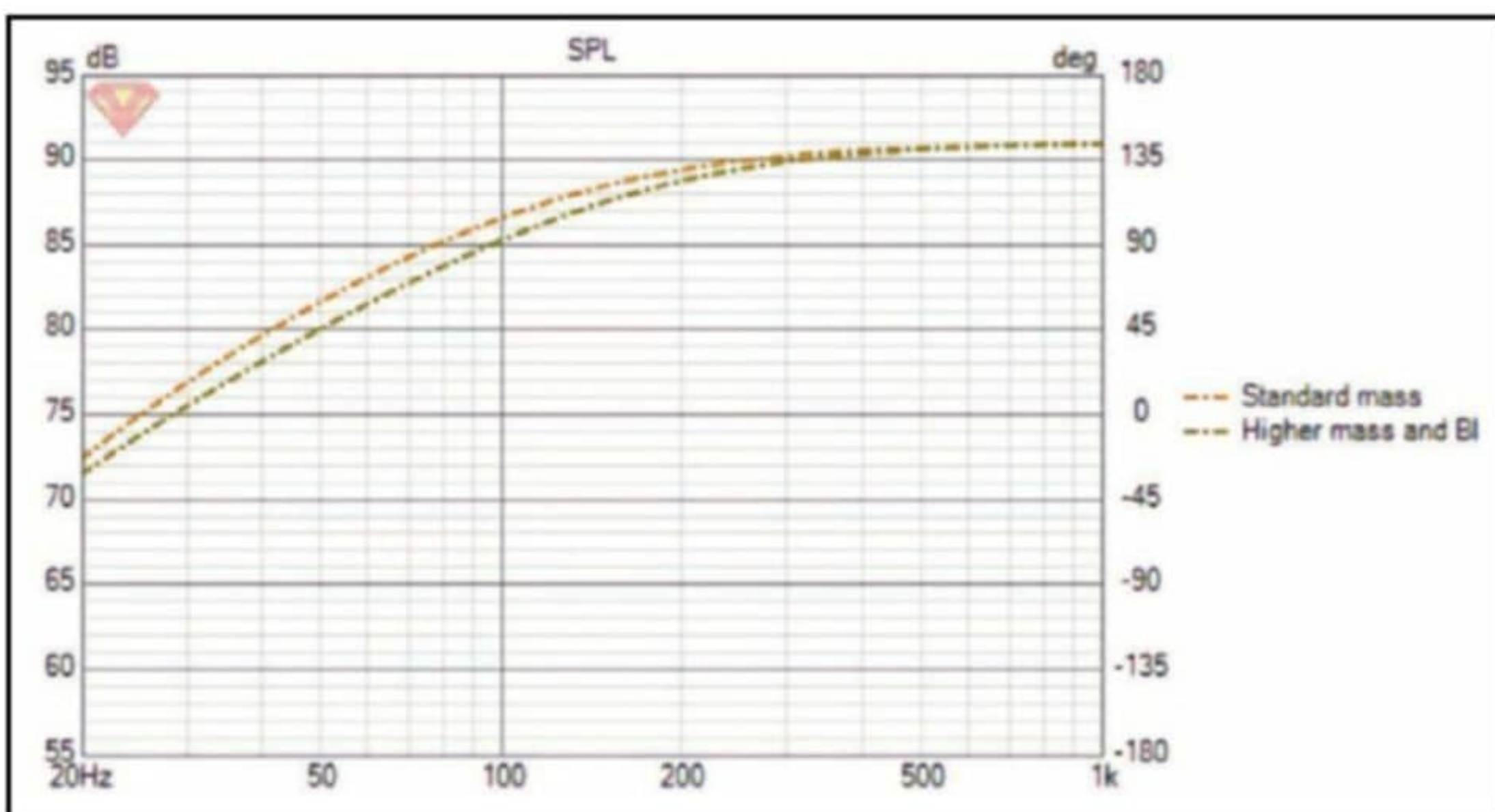


Figure 5: A simulation in Vituix Cad shows how a minor increase in mass, compensated with increased BL, still does not give the same low-end sensitivity as a lower mass option.

low while at the same time ensuring sufficient stiffness in the surround for it to fulfill its purpose as both a seal and a mechanical suspension. Narrow roll cloth or foam typically gives the best results. Surrounds inherited from woofers are rarely a good idea. Many of the same rules apply to spiders, but while the cone can be made a bit larger to fit a narrower surround, one needs the spider to cover the distance from the spider landing to the voice coil, unless an adapter is being used. The spider also is significantly better controlled by the coil and is not exposed to the same air pressure, so it is not as critical as the surround.

Troels Gravesen has experimented with edge coating on a 4.5" woofer in an attempt to improve its midrange performance [2]. He performed a blind listening test between the edge-coated and non-coated versions of the same driver. Although there was no clear winner, the listeners were surprised that the perceived sound characteristics were so different. This not only shows that adding loss made a huge difference to the sound, it also shows that achieving a smoother response by masking resonances does not equal increased sound quality.

4: The total moving mass should be less than 0.08g/cm².

Having low mass does not directly give advantages. However, it makes sense in at least two aspects. First, a mode in a high mass diaphragm has some significant energy to it. The diaphragm modes depend both on mass, compliance, and geometry. Part of this mass is already outside the diaphragm, and since we have already put down some ground rules for coil and suspension, this can be a significant mass for the diaphragm.

Second, if we are not talking about very large midrange chambers, the moving mass represents a trade-off between low mid extension and sensitivity. We could of course go even lower, but it would be hard to reach the low end and the low-end sensitivity would potentially go down instead of up. If we go higher, efficiency quickly drops and that is very bad if the driver uses a very compact coil to begin with. We could try to compensate by adding more force, but then we would also lose low-end sensitivity as the pass band moves further up in frequency. **Figure 5** illustrates this example.

5: Any steel present in the gap should be saturated to >1.9T.

The AC field from the voice coil should never distort the permanent field. The most important reason is the obvious deviation from ideal BL (x,i). When a strong force field from the coil pushes on



Improving our environments *with sound and vibration measurement solutions from Scantek.*

Scantek is a leader in sound and vibration measuring equipment sales, service, rental, and calibration.

At Scantek, we understand how important accurate sound reading and output data need to be in a professional setting. That is why we provide each customer with a thorough and personalized sales experience as well as unparalleled support when selecting sound and vibration measuring solutions that meet both their needs and budget.

Our mission is to provide expert advice and support on the selection and use of the products that we sell, service, rent, and calibrate. We offer a complete line of products known worldwide for being the best for sound and vibration measurement and analysis.

***Call us today to speak to an expert to select the best solution
for your sound and vibration measurement needs.***

Sound Level Meters | Vibration Level Meters | Acoustic Cameras | Multi-channel Analyzers | Data Recorders | Noise Sources | Sound Limiters
Dosimeters | Prediction & Calculation Software | Analysis and Reporting Software | Microphones and Preamplifiers | Accelerometers



800-224-3813
info@scantekinc.com
www.scantekinc.com

the permanent field we get a change in the force factor. This adds itself directly to the signal gain. Also, paramagnetic material has a complex set of properties. Translated to a lumped model, it would be represented by numerous different components. We have even been forced to invent some artificial components to fully represent the mathematical properties of a paramagnetic material. And with that we have not even included the hysteresis. All of this adds up to huge amounts of distortion, but as soon as we get past 1.9T, all steel types will suddenly approach the magnetic properties of vacuum (**Figure 6**). This means that all these non-linearities disappear, and with them, the distortion.

Any magnetic material present close to the coil therefore needs to be saturated. The level of saturation depends on the strength of the opposing field, but anything less than 1.9T for steel is not recommended. At the same time 2.4T is not likely to be any better than 2.1T, unless the paramagnetic material has a higher saturation point like some cobalt based materials. It is also crucial to point out that this is valid for magnetic materials, but not for the air gap itself. Air has basically the same magnetic properties as vacuum meaning it has no hysteresis and therefore does not behave any better with higher flux.

For a high efficiency midrange driver for home audio, you will rarely exceed 2A. Given the design guidelines for coil mass, you will not end up with a lot of turns in an 8Ω coil. The total magnetic field produced by the coil is the product of the current and the number of turns in the magnetic field. Few turns and low current means the magnetic circuit can easily handle the field from the coil. For low efficiency drivers with higher current density in the gap, the magnetic circuit may need to be saturated at a higher level.

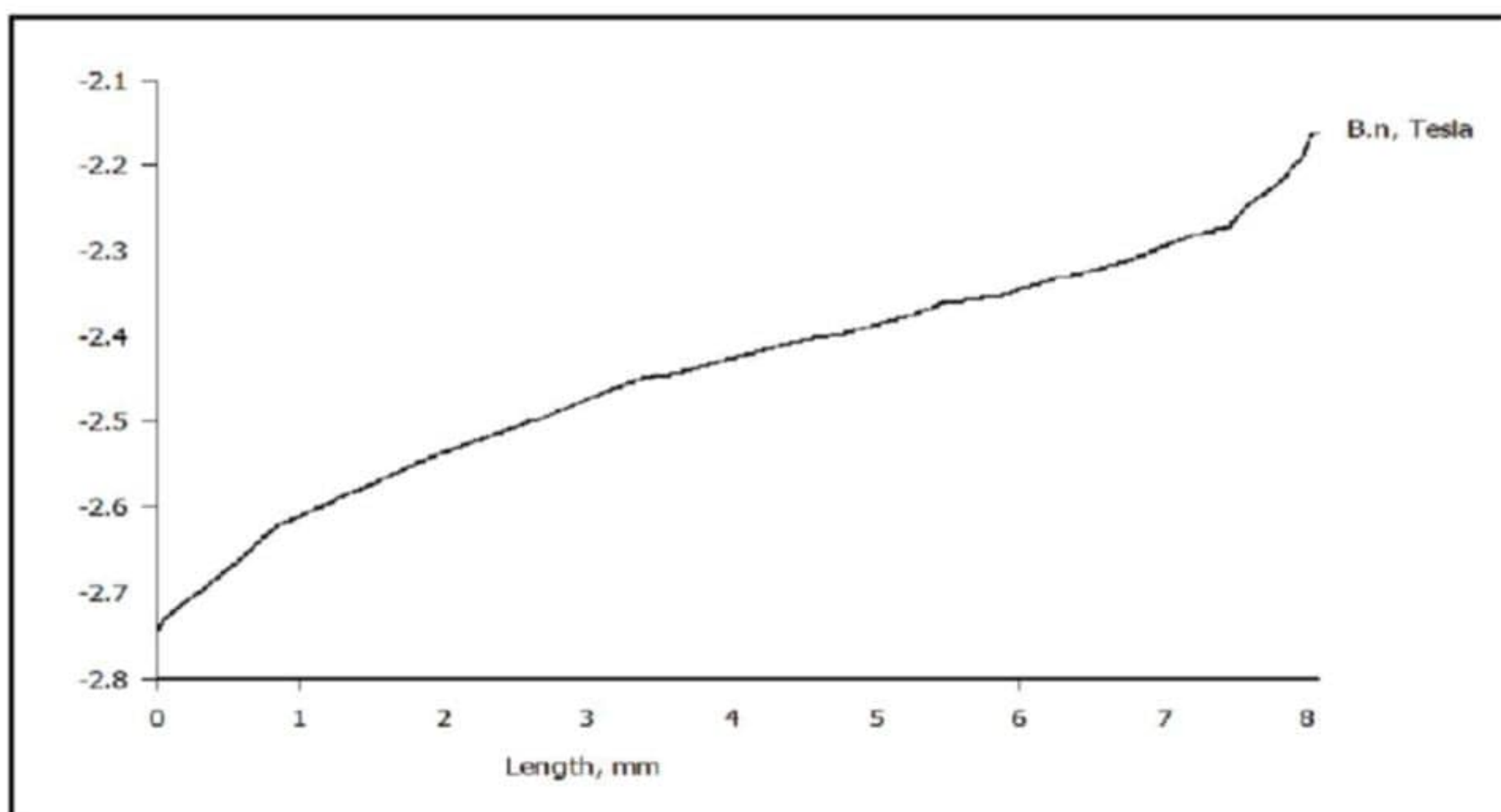


Figure 6: Saturation level simulated within steel core close to the gap in a saturated midrange driver motor. Simulation made with FEMM (Finite Element Method Magnetics).

6: The driver should use shorting rings inside the gap.

While the stability of the motor is taken care of by saturation it will in most cases be advantageous to reduce the inductance further. Saturation works well at low frequencies; at higher frequencies the addition of shorting rings will further improve the performance. The kind of effect we are looking for is to further reduce the inductance as well as to make it more linear as a function of coil position. Lower inductance means the driver can handle more current at higher frequencies with less distortion, but it also means that the inductive component of the driver's total impedance is lower.

Most amplifiers are voltage controlled, while the acoustic output of a driver is proportional to the current through the coil. If we double the impedance, we also reduce the current by 50% resulting in a 3dB reduction in acoustic output. This is most prominent when having long excursion and high frequencies at the same time. Therefore, this will typically be an intermodulation distortion issue. It is hard to reveal this effect just by measuring harmonic distortion in a driver.

When combined with a passive crossover, acting like a voltage divider between the driver and the crossover components, the effect of modulated impedance will further add to the intermodulation distortion, but slightly differently to what we see with drivers connected directly to a voltage amplifier. The addition of series components will make the driver see a transition from voltage drive to current drive.

Italian driver manufacturer Eighteen Sound has an interesting document showing the effect of shorting rings on current stability in non-saturated drivers [3].

7: The coil should be wound with aluminum ribbon wire on a bobbin no more conductive than titanium.

Using a bobbin with little or no conductivity helps keep the overall loss low. The frequency response of the driver will probably be less flat compared to using an aluminum bobbin, but that is due to lower loss. But loss should not be the tool for keeping resonances under control. They will still be there, just not as prominent on the frequency response curve.

The eddy currents in a conductive bobbin represent an actual force, altering the flux field in the motor core. This again will not disappear by itself, instead it ends up as a part of the flux field that drives the diaphragm. Distributing so much energy in the time domain takes away precision

from the driver. This may not affect harmonic distortion as it comes back at other frequencies, appearing like a totally random pattern. This is commonly filtered out in frequency response measurements. We simply do not want anything moving in the magnet gap that also interacts with the permanent field.

But the coil should be aluminum, and it should be ribbon. The reason is that we want the maximum field density possible in the coil while at the same time keeping it as light as possible. If we accept a bit more mass, we could use copper instead and get even higher current density. Silver will give about the same result as copper. Beryllium would give a lighter coil than aluminum, but with a bit less conductivity it would not have an advantage.

If we compare the same driver with aluminum, beryllium, copper and silver coil, aluminum and beryllium would give the highest sensitivity. The added resistivity of beryllium would require more material. In addition, it would require a wider gap. Therefore, the performance of the driver with beryllium coil will not be improved compared to aluminum.

8: The motor should allow for 1mm of excursion with less than 5% deviation of $Bl(x,i)$.

This can be done with ease on most drivers, until we start requiring a very light coil. It requires some FEM-modeling and is not hard to achieve. I wrote a previous article about this topic where I found that we should require 2mm of excursion with less than a 2.5% deviation. That is far more challenging to achieve and should certainly be a goal for everyone who wants to make a high-end midrange driver. Most midranges will live most of their life below

About the Author

Jørn Rune Kviserud has a background in physics, acoustics, and electronics with loudspeaker technology at the core. He started as an audio hobbyist in 1984 and for the past 20 years his work has mainly been R&D and product development within the audio industry. He founded Midgard Audio (later Tonalab, www.tonalab.com, Norway) in 2004 and has worked with numerous clients over the years. His focus in recent years has been to investigate what determines objective sound quality in order to further develop this field of knowledge and determine how it may lead to improved products.



- 5" SATORI MR13TX-4
- 6.5" SATORI MR16TX-8

NEW RELEASE!

Find us on:

- SB Acoustics
- sbacoustics
- sbacoustics_
- SB Acoustics
- sbacoustics.com

1mm of excursion. A good starting point would be to say that the driver rarely passes 1mm, and rarely exceeds 2A. All developers should simulate this very carefully to determine both the peak excursion and the peak current for a given system. The deviation from $Bl(x,i)$ simply determines the level of intermodulation distortion and is very much a design requirement. **Figure 7** illustrates simulated performance for a high-performance driver.

A $Bl(x,i)$ simulation will reveal how successful the motor design is. It also reveals the defining factors for intermodulation distortion in the driver and the driver's upper limit for usable output. A couple of things to really look for, to understand if the motor design is successful, is if the inductance modulates with current, and if you get asymmetric DC offset in the force factor). Both will reveal when the core falls below saturation. This can often be exaggerated at peak excursion so make sure to check that too.

9: The efficiency of the driver should be at least 86dB/1W/1m free space and transient capacity at +20dB, still at acceptable distortion levels.

This does not sound like much, but it is. We are talking about 100W in a very small coil, which is a challenge on its own. But given that we typically use two drivers mounted in their own baffles, we are looking at around 116dB just from the pair of midrange drivers mostly above baffle step. For a complete system, crossed over at 2Hz to 300Hz, and

with typical energy distribution in music, we are typically past 120dB at this point. A 15cm driver designed following the above criteria should be able to handle this within its excursion limit. But there are serious forces here, and that tiny coil will get seriously hot. The heat needs to escape somewhere. Any added sensitivity we can get will reduce the required power. If we can just get the efficiency up another 3dB we are down to 50W. The difference between an 84dB and a 90dB efficiency driver is very audible unless you use a lot of them.

Note that these are free space numbers, not to be confused with IEC or infinite baffle measurements. The latter will be 6dB higher, so this means we are talking about 92dB to 94dB efficient small drivers. And if we look at the sensitivity of 4Ω drivers, we are further 3dB up to 95dB to 98dB. So, if 86dB efficiency sounded a bit modest at first, I can assure you it is quite an ambitious number.

10: The cone should have no modes for at least 1 octave past its low-pass crossover point.

We do not always get detailed information about the actual internal damping of a cone material. A better sounding driver should achieve high bandwidth without adding additional damping to the cone. If this turns out to be impossible, consider a different cone material to push the modes higher. At the same time, materials that produce very hard break-up modes also tend to produce strong sub-harmonic components that are both hard to measure precisely, and to ultimately combat, in a crossover. This comes down to both shape and material or combination of materials.

We should also use a bit of common sense here. Using a metal cone can sometimes be a good idea, but in conjunction with a low-order crossover having a breakup only one octave from the crossover point could be a huge problem. You cannot fully fight this with a notch filter as it will also be triggered by other frequencies such as harmonic distortion. Also, it may produce lots of stored energy around one octave below, meaning it ends up directly at the crossover point.

A softer cone would normally work well if the breakup is around one octave away from the crossover point. For very small midrange drivers, metal can be quite unproblematic due to the high break-up point. The breakup pattern also depends on the cone profile. For midrange drivers the most desirable shape is almost always a curved cone shape. Some materials, like spread-tow carbon can have desirable modal properties, but do not allow the desired shape when weaved in a bidirectional plane.

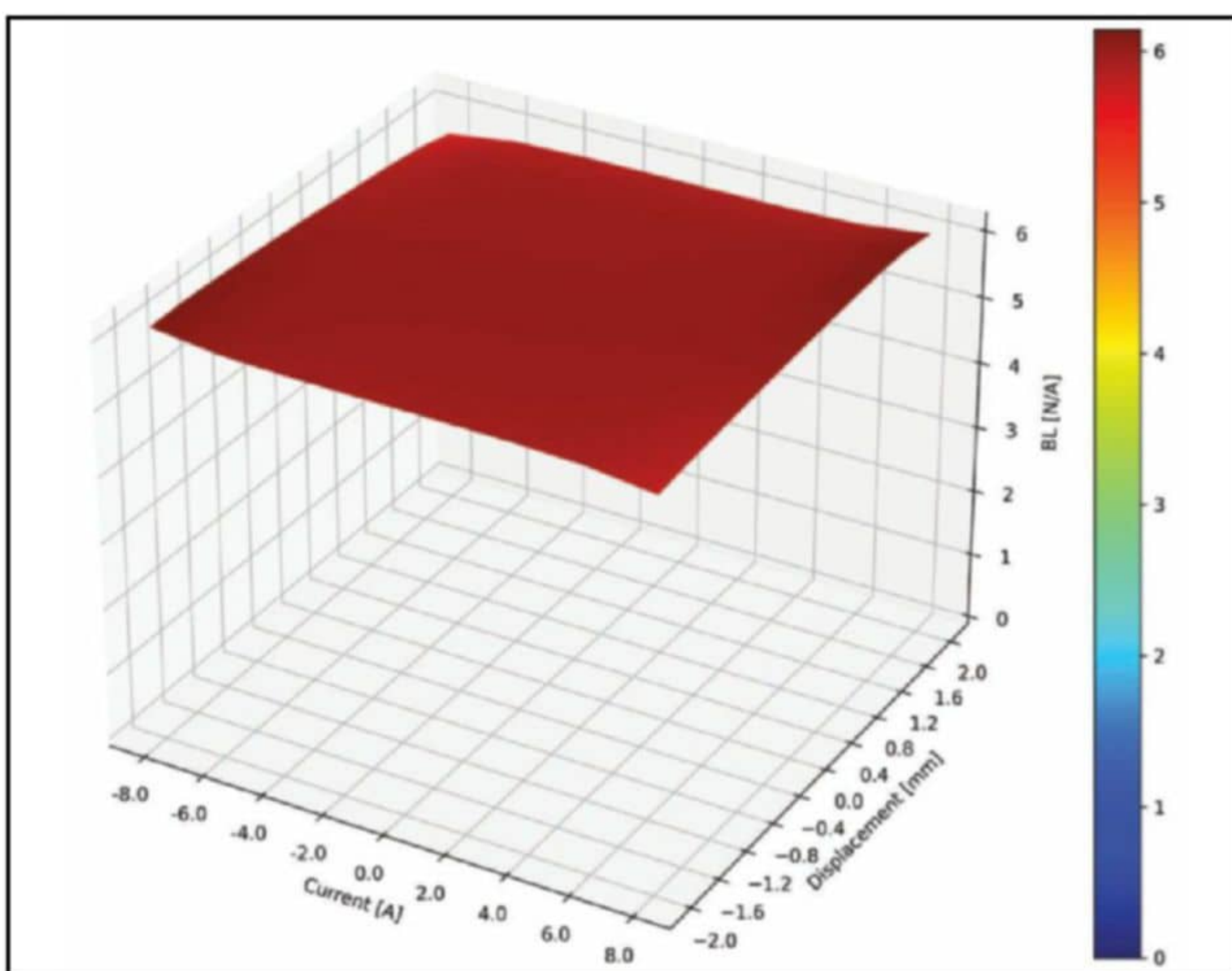


Figure 7: Simulation of a midrange motor showing an uncommonly perfect current stability as high as 6A. This will give the driver very low distortion. Simulation made with Tonalab ELMA.

What Else?

At this point, some questions may have popped up. I have deliberately avoided many of them that may lead the reader in the wrong direction by presenting a solution for achieving a particular goal as the goal itself. Many answers are given by the goals one aims to achieve.

One question may be the motor geometry. Should it be over hung? Under hung? Dual gap? Split coil? It does not matter in itself because what matters is covered in the list. You will, however, most likely end up with under hung as the only technology where you will be able to achieve the desired parameters.

Another question could be about cone shape and material. Again, it is not easy to achieve the listed requirements with any material. In a 6.5" midrange one could use metal if the crossover point to the tweeter is low, but for a 3kHz crossover point you most likely end up hitting the first mode just above the crossover point. This means that the bandwidth requirements also determine what might work. And let us not mislead ourselves by thinking that we can remove a resonance in the driver using some trick in the crossover.

Coil size is another question, but again, it is given by so many other parameters. Are you able to saturate a tall enough gap with a 20mm coil (given the small diameter pole piece)? Are you able to achieve the mass goal with a 50mm coil? While a diaphragm with

twice the diameter has four times the cross section, a coil with twice the diameter only has twice the surface area for cooling as well. The same goes for pole piece cross section vs gap diameter, so making the coil very big to improve cooling and at the same time trade efficiency, can be a very inefficient way to design a driver.

A True Midrange Driver

As far as I know, there are no drivers currently available in the market that can achieve all the above-listed parameters. Some drivers fit a few, but most midrange drivers are far from some of them too. A driver that fits all of them would be easy to work with and can be made to work well with pretty much any woofer and tweeter. Any driver that fulfills the complete list should be allowed to be labeled as a "True Midrange Driver."

References

[1] "M - Investigation of Energy Storage - Part 2," Linkwitz Lab, www.linkwitzlab.com/frontiers-2-M2.htm

[2] T. Gravesen, "W12CY003: Linear distortion and edge coating," January 2021, www.troelsgravesen.dk/W12CY003.htm

[3] "Eighteen Sound Single and Double Modulating Rings," Eighteen Sound, www.eighteensound.it/en/technologies

WHERE'S THAT RUB & BUZZ NOISE COMING FROM?

It could be the product you're testing or the instruments you're testing with.

Know for sure with low noise microphones.

SCAN TO LEARN

PCB PIEZOTRONICS
AN AMPHENOL COMPANY

1 800 828 8840 | pcb.com

DO IT YOUR WAY

Panels/Boxes/Cables

WITH REDCO AUDIO'S

"Build Your Own" web pages

Full control for design, editing and more.

WE'LL BUILD IT JUST THE WAY YOU WANT!

REDCO AUDIO

www.redco.com 203-502-7600

Bass Reflex Performance Envelope

Part 2 — System Analysis

By
Gary Gesellchen (Vanatoo)

This article takes a novel approach to estimating the performance envelope available from bass reflex loudspeakers. Part 2 expands on the system analysis, showing how driver and port sizes determine how loud and how low a system will play, and how that relates to generally available drivers.



The frequency response of a bass reflex speaker with a fourth-order high-pass Butterworth response and a f_3 (-3dB frequency) of 100Hz is shown as **Figure 2**. In this example the system is putting out 83dB, and you can see it drops to 80dB, or -3dB, at 100Hz.

Figure 2 is a Butterworth frequency response, but all bass reflex alignments responses are similar in nature. Figure 1 (*audioXpress*, July 2023) showed us the basic behavior at very low and very high frequencies. Figure 2 shows us how those behaviors add together in the frequency domain.

Below box resonance, the port and driver are moving out of phase (from Figure 1), which makes their two outputs partially cancel one another. Therefore, their sum, which is the system output, is less than either one individually. Individually each drop at a rate of 12dB/octave below the f_3 frequency. But since they are out of phase, they sum to a response that drops at 24dB/octave, which is characteristic of “fourth order” (each “order” is 6dB/octave).

At box resonance, we can see there is a sharp dip in the driver output, with nearly all the system output coming from the port. The port and driver migrate from being mostly out of phase an octave below the box resonance to mostly in phase an octave above the box resonance.

Above the box resonance, the port and the driver are moving in phase (from Figure 1), but the port output decreases rapidly with increasing frequency (at 12dB/octave), such that by the time you get to $2 \times f_3$ (200Hz) the driver is putting out most of the sound.

Figure 3 shows the displacement of the driver and the air velocity in the port for the system with the response shown in Figure 2. These two parameters were chosen because they are what limit the physical output of the system. All real-world drivers have a physical limit of how far the cone can move in and out without damage. The distance they can travel while still being under good control of both the magnetic motor and the cone’s suspension system will always be somewhat less

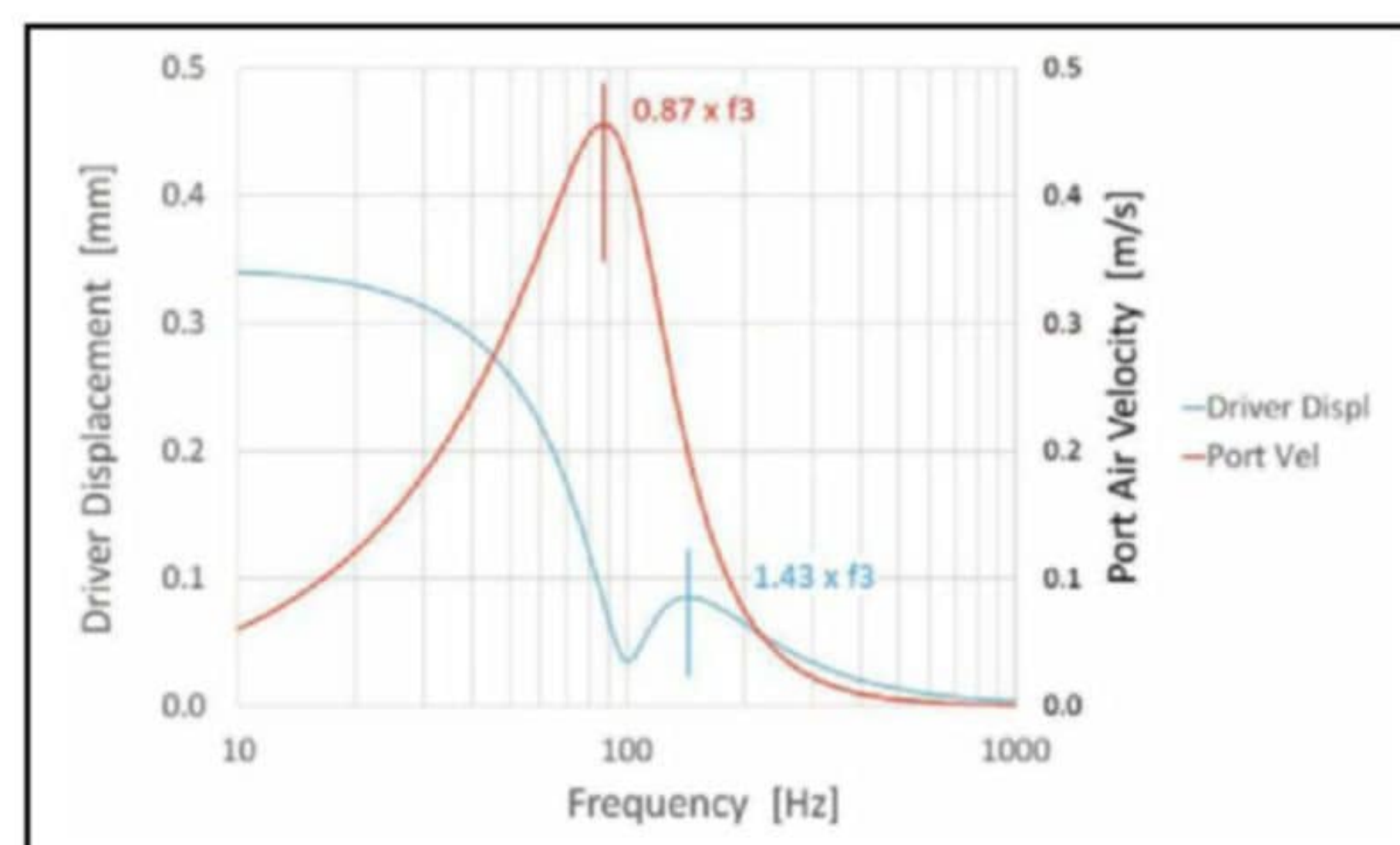
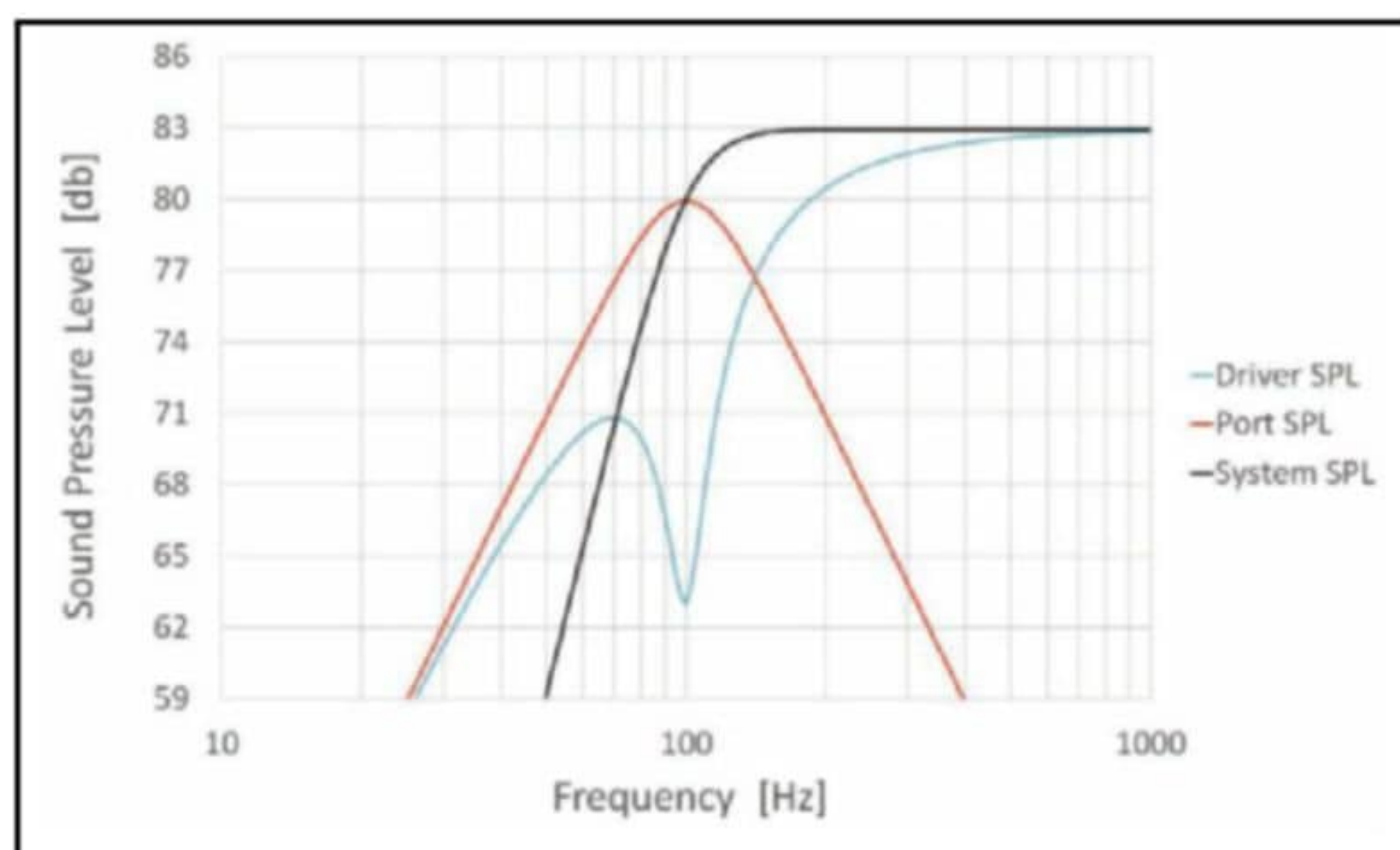


Figure 2: Fourth-order Butterworth speaker response

Figure 3: Fourth-order Butterworth driver displacement and port velocity

and is designated by the manufacturer as X_{max} . X_{max} is the farthest you can practically drive the cone before you run into excess distortion. Similarly, only so much air can flow through the port before it starts to resist the air flow and stops working properly. The variable that determines if the port will flow well is the velocity of the air in the port, which is why it is plotted in Figure 3.

Looking first at the port velocity in Figure 3, we see that it peaks at 87Hz. Looking next at the driver displacement (above the 100Hz f_3), we see that it peaks at 143Hz. The above trend will hold for all Butterworth ported designs: The port will max-out at $0.872 \times f_3$, and the driver will max-out at $1.432 \times f_3$. The curves will always look similar to Figure 3, with the frequency axis scaled by the actual f_3 frequency. This might not seem like much, but it's hugely important. It says that if we have a Butterworth response and we know the f_3 , we also know the two frequencies where the physical limits, driver displacement and port velocity, happen.

Note that the driver displacement increases radically below the f_3 frequency and is limited at DC only by the driver stiffness. This is why ported speakers can experience cone flapping displacements when fed signals below their passband. We're going to assume that the displacement below the system f_3 will be taken care of by some other means and that we only need to worry about the maxima above f_3 . Note also that the large decrease in driver displacement around the f_3 is one of the big advantages of the ported vs a sealed design. Most of the distortion producing mechanisms in a driver are related to displacement and producing more SPL for a given displacement generally means less distortion.

If we study Figure 2 at the above maxima frequencies, we will see that at $0.87 \times f_3$ the port is down by 3.63dB compared to the system's mid-band. Similarly, we see that the driver is down by 5.86dB at $1.43 \times f_3$. We now know the frequencies and something about the output levels that are going to represent the system limits.

$$f_{p \max}(f_3) = 0.872 \times f_3 \quad [1]$$

$$f_{d \max}(f_3) = 1.432 \times f_3 \quad [2]$$

$$SPL_{pm} = -3.63 \quad [3]$$

$$SPL_{dm} = -5.86 \quad [4]$$

Where: $f_{p \max}$ is the frequency of maximum port air velocity
 $f_{d \max}$ is the frequency of maximum driver displacement
 SPL_{pm} is decibels the port is down compared to mid-band
 SPL_{dm} is decibels the driver is down compared to mid-band

If we define the maximum system mid-band SPL as $SPLM$, we know that at the system limits:

$$\begin{aligned} SPL_p(f_{p \max}(f_3), SPLM) &= SPL_p(0.872 \times f_3, SPLM) = \\ SPLM + SPL_{pm} &= SPLM - 3.63 \end{aligned} \quad [5]$$

$$\begin{aligned} SPL_d(f_{d \max}(f_3), SPLM) &= SPL_d(1.432 \times f_3, SPLM) = \\ SPLM + SPL_{dm} &= SPLM - 5.86 \end{aligned} \quad [6]$$

Where: SPL_p is the SPL produced by the port at one meter
 SPL_d is the SPL produced by the driver at one meter

Equation 5 and Equation 6 are enough to fully constrain the system response based on the physical limits of the port and driver. They say what the port and driver must each be capable of doing to support a system with a given -3dB frequency (f_3) and given maximum SPL ($SPLM$). At a frequency of $0.873 \times f_3$ the port must be able to output an SPL of $SPLM - 3.63$ dB, and at frequency $1.432 \times f_3$ the driver must be able to output an SPL of $SPLM - 5.86$ dB. We have constrained what the port and driver must be able to do to meet a given B4 response at a given maximum SPL, and it resulted in two simple equations!

Driver and Port Output

To make use of Equation 5 and Equation 6, we need to know how the physical limits of the port and driver affect the SPL they can produce. The Thiele-Small (T-S) model assumes that the driver is mounted on an infinite plane. The equations for the sound radiated by a circular piston on an infinite plane are well established. The simplified version of the SPL produced at one meter on-axis from a circular piston on a plane is:

$$SPL(f, S, V) = 20 \times \log \left(\frac{\rho \times f \times S \times V}{2 \times 10^{-5}} \right) \quad [7]$$

Where: f is the frequency Hertz [Hz]

ρ is the air density of 1.204 [kg/m³]

S is the piston area [m²]

V is the piston velocity [m/s]

If we assume that the driver and port can be considered as flat circular pistons on an infinite plane, we can derive the SPL they generate at one meter on axis as shown.

$$SPL_p(f, S_p, V_p) = 95.59 + 20 \times \log(S_p \times V_p \times f) \quad [8]$$

$$SPL_d(f, S_d, X_d) = 111.56 + 20 \times \log(S_d \times X_d \times f^2) \quad [9]$$

Where: f is the frequency

SPL_p is the SPL at one meter from the port decibels [dB]

S_p is the port area [m²]

V_p is the port air velocity [m/s]

SPL_d is the SPL at one meter from the driver [dB]

S_d is the driver area [m²]

X_d is the driver displacement [m]

If we set Equation 8 equal to Equation 5, we have the port constraint. Setting Equation 9 equal to Equation 6 constrains the driver.

$$SPLM - 3.63 = 95.59 + 20 \times \log(S_p \times V_{p \max} \times f) \quad [10]$$

$$SPLM - 5.86 = 111.56 + 20 \times \log(Sd \times X_{d \max} \times f^2) \quad [11]$$

Where: $V_{p \max}$ is the maximum allowable port velocity
 $X_{d \max}$ is the driver's published $X_{\max}$

According to Equation 10 the port is limited by its area, the frequency, and the maximum velocity. We're going to assume without proof that a reasonable maximum velocity limit is 20m/sec. This is a number that floats around the Audio DIY forums and seems to correlate well with the author's own experience. If we assume $V_{p \max}$ is 20, we can rearrange Equation 10 to solve for the port area (S_p), substitute in Equation (1) for the frequency, and simplify as:

$$S_p(f_3, SPLM) = \frac{10^{\frac{SPLM - 124.05}{20}}}{f_3^2} \quad [12]$$

Equation 12 is the area of the port [m^2] required to produce the SPL necessary for a system with a -3dB frequency of f_3 to produce a maximum SPL of SPLM. The port is now adequately defined. Equation 12 will prove useful as shown, but can also be rearranged to solve for SPLM as a function of the port diameter (D_p) as shown in Equation 14.

$$D_p(S_p) = 44.424 \times \sqrt{S_p} \quad [13]$$

Where: S_p is port area in m^2
 D_p is port diameter in inches

$$SPLM(f_3, D_p) = 58.14 + 20 \times \log(f_3 \times D_p^2) \quad [14]$$

Where: SPLM is the system mid-band Max SPL in decibels
 f_3 is the system -3dB frequency in Hertz
 D_p is port diameter in inches

(Note the mish-mash of units in Equation 13. This article normally uses SI units, but port and driver sizes are typically specified

in inches. We need S_p and S_d in meters squared for the other calculations.)

Equation 12 and Equation 14 define the port requirement. Equation 11 will similarly be used to define the driver requirement to support f_3 and SPLM. Looking at Equation 11, we see the sum of $S_d \times X_{\max}$, which is defined as the driver volume displacement. We could leave S_d and $X_{\max}$ in the equations as they are both readily available for high quality drivers. But we are interested in making sweeping system-level statements. It would be more useful if we could make a reasonable approximation of the quantity $S_d \times X_{\max}$ as a function of the driver size. This would tell us how big of a driver is required for any combination of f_3 and SPLM.

Commercial Drivers

Loudspeakers drivers are marketed by their nominal size, specified as the driver frame outside diameter in inches. The diameter of the cone is always smaller than the advertised diameter, d . I looked at the published cone area (S_d) of a lot of different drivers (i.e., woofers) with different advertised sizes (d). While there is obviously some variation, the following relationship was found to closely match the parameters of commercially available drivers.

$$S_d(d) = 0.00035 \times d^2 \quad [15]$$

Where: d is published diameter in inches
 S_d is cone area in m^2

We can also make some generalizations about the driver $X_{\max}$ as a function of the advertised size. The drivers we're concerned with are relatively long travel woofers. A reasonable approximation for the $X_{\max}$ you can expect from a high-quality woofer as a function of its advertised size is:

$$X_{\max}(d) = 2 + \frac{d}{2} \quad [16]$$

Where: d is published diameter in inches
 $X_{\max}$ is the displacement in millimeters

Combining Equation 15 and Equation 16 gives us the volume displacement ($S_d \times X_{\max}$) as a function of the driver size:

$$VD(d) = 3.5 \times 10^{-7} \times \left(\frac{d^3}{2} + 2 \times d^2 \right) \quad [17]$$

Where: d is published diameter in inches
 VD is the volume displacement in m

Figure 4 shows Equation 17 plotted as hi-fi drivers. Subwoofer drivers, with their emphasis on low-frequency output, can typically have as much as twice the $X_{\max}$ we assumed in Equation 16. Pro Audio Drivers, with their emphasis on efficiency, will typically have about half the $X_{\max}$ we assumed in Equation 16. Both are plotted in Figure 4. They show a Subwoofer outputting 6db more and a Pro Audio woofer outputting 6db less than our target hi-fi solution.

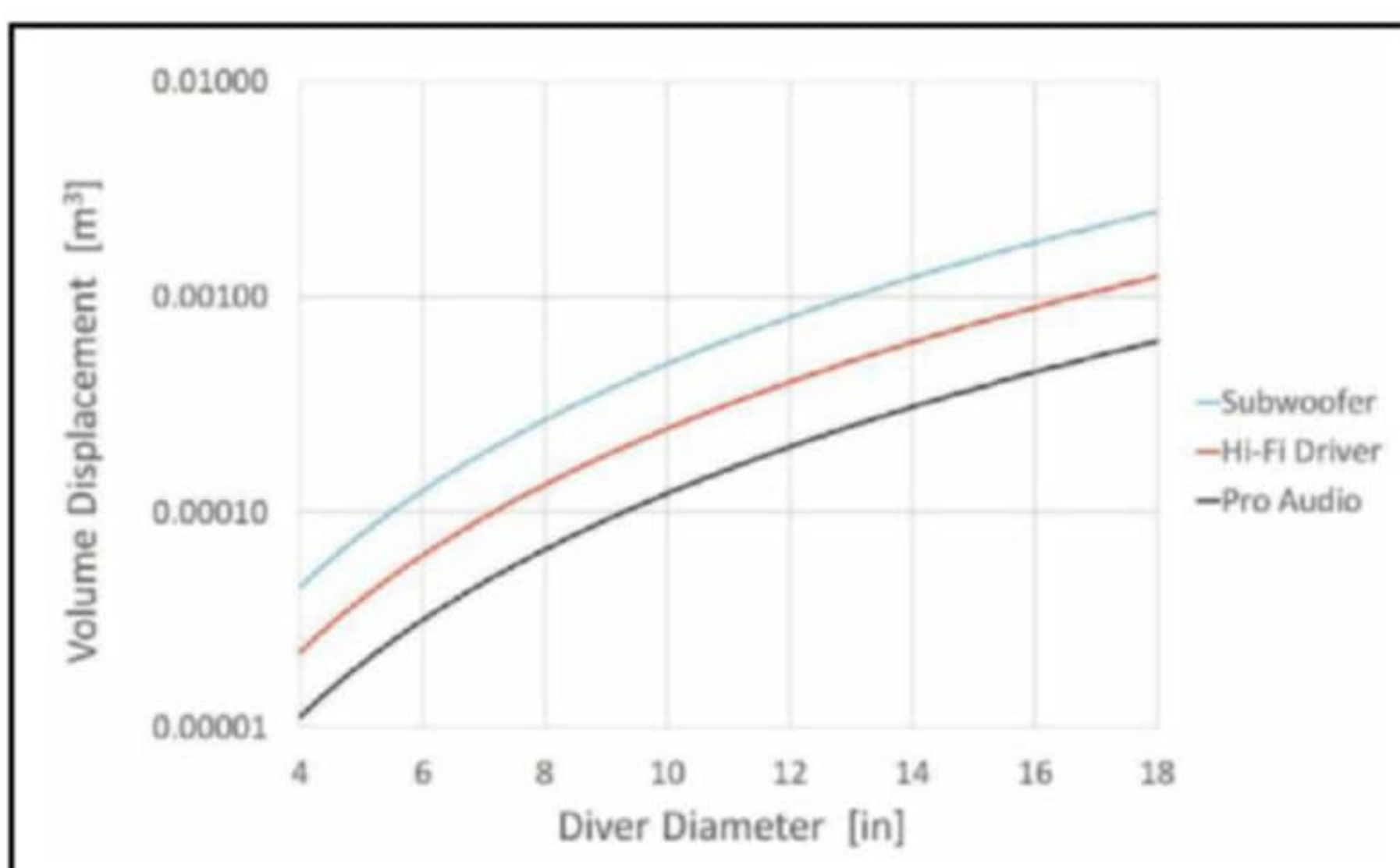


Figure 4: Driver volume displacement vs. size. Equation 17 is plotted for Subwoofer, Hi-Fi, and Pro Audio drivers.



Cine Gear EXPO 2023

Atlanta

October 6 & 7

Trilith Studios



cinegearexpo.com

We are now able to substitute Equation 17 for the volume displacement as a function of the driver size into Equation 11 for the driver requirement. Doing so gives the maximum system SPL driver requirement as:

$$SPLM(f_3, d) = -5.46 + 20 \times \log \left(f_3^2 \times \left(\frac{d^3}{2} + 2 \times d^2 \right) \right) \quad [18]$$

Equation 18 gives us our estimate of the maximum mid-band system SPL (SPLM) as a function of two basic things: 1) The system -3db point (f3); and 2) The advertised size of the driver. It makes perfect sense, yet it's still a great finding.

We now have two pretty simple equations for the maximum system mid-band SPL based on the system f3, the port diameter (Equation 14), and the driver size (Equation 18). Plotting Equation 14 and Equation 18 against the system -3dB frequency (f3) will show us how the maximum mid-band SPL (SPLM) varies with the port and driver size.

Figure 5 is a plot of Equation 14 with the port based maximum SPL plotted against the system f3. This is all of a sudden a lot of information. Speaker system f3 frequencies are widely, if inaccurately, reported. Figure 5 says that if you know the system

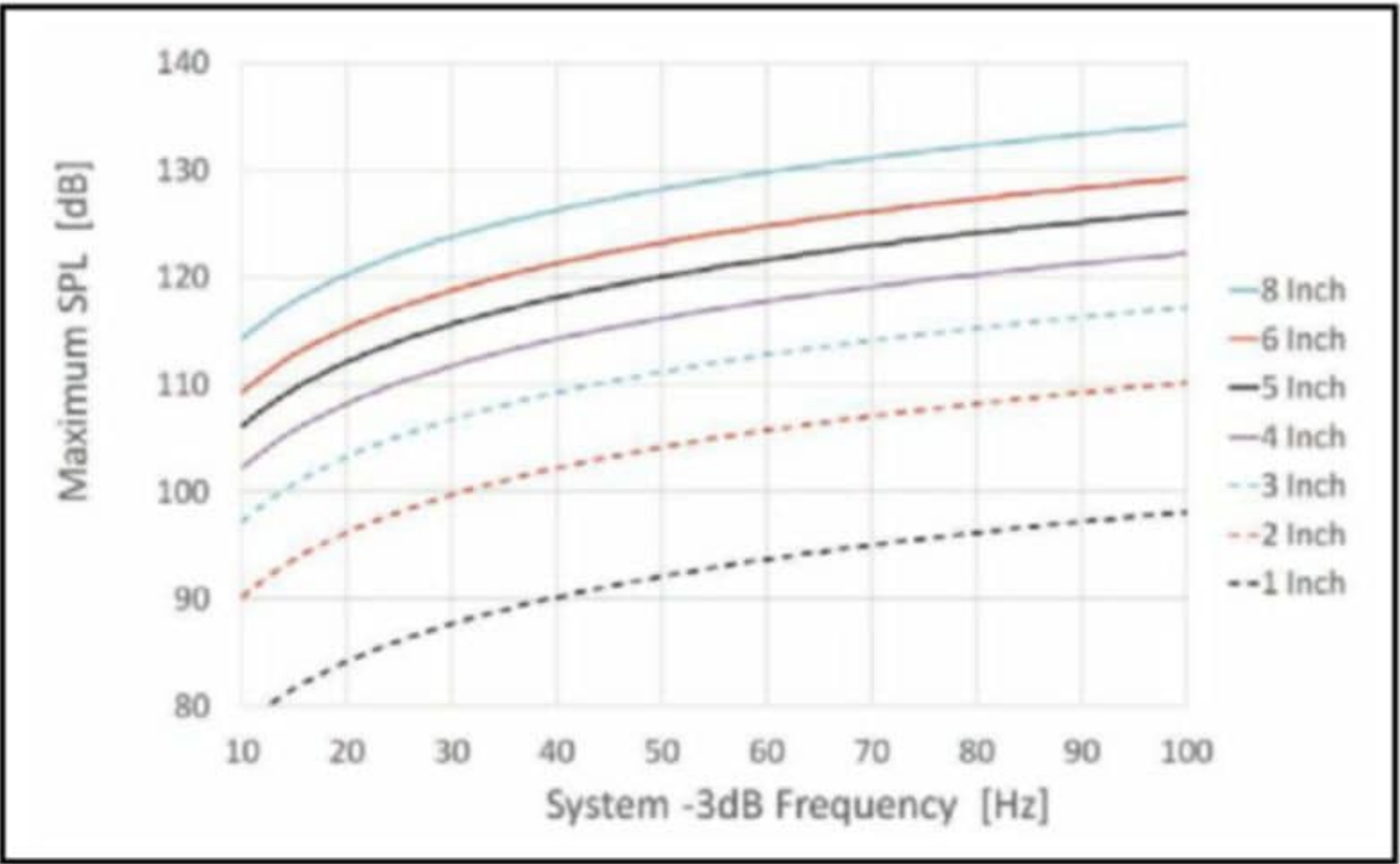


Figure 5: Maximum system SPL vs. system f3 for various port diameters (from Equation 14)

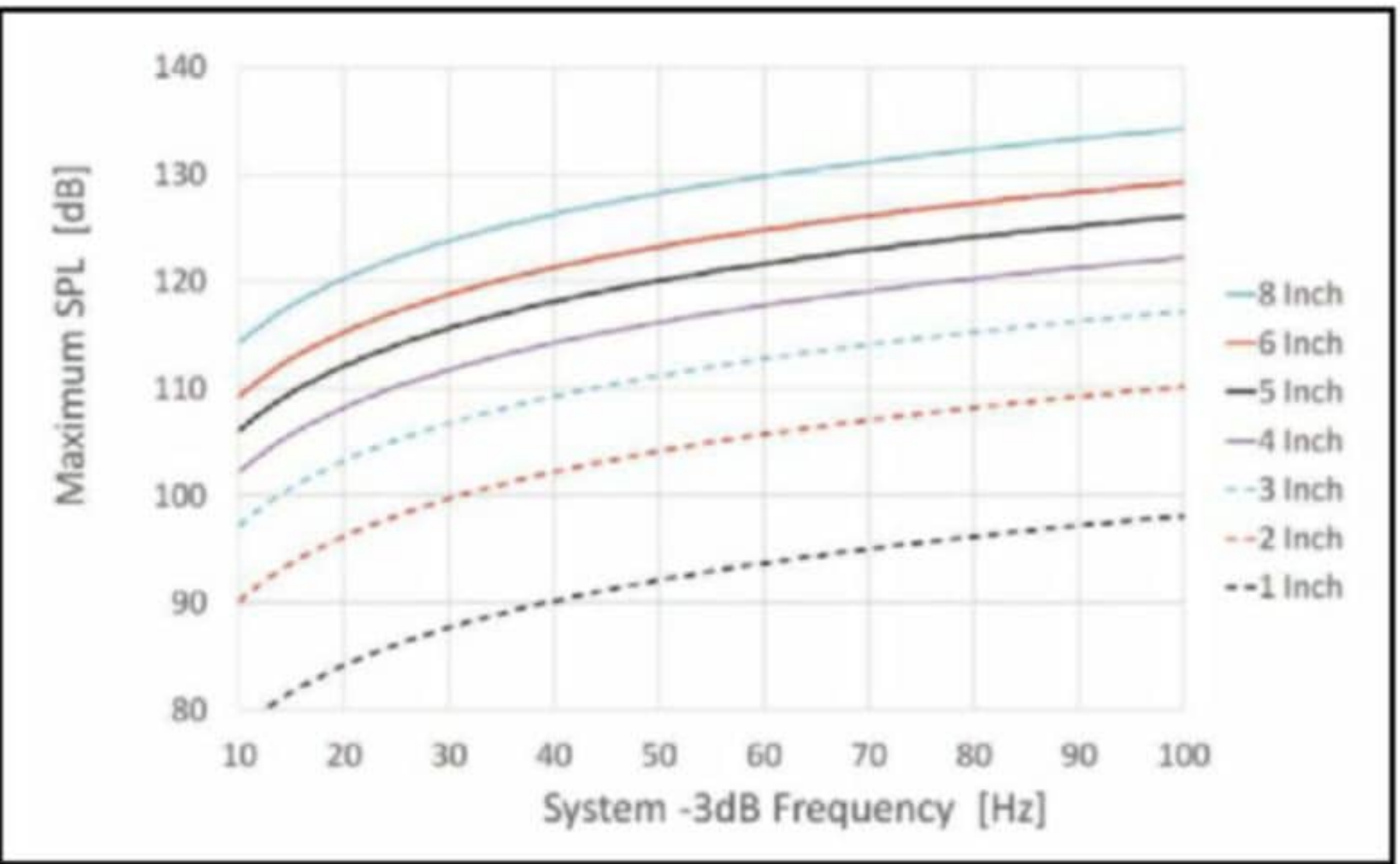


Figure 6: Maximum system SPL vs. system f3 for various driver sizes (from Equation 18)

f3 and the port diameter, you know the maximum SPL it is likely to play cleanly. There's no guarantee that any commercially available system is designed according to our assumptions. But our assumptions have all tended toward best case, and since Figure 5 represents the maximum system SPL for our hypothetically perfect system, it likely represents a reasonable upper limit of expectations from a commercially available system.

Figure 5 says that if you want to play low and loud, you need a really big port. If the system f3 is reliably known, Figure 5 will put a reasonable upper limit on the maximum SPL one should expect based on the port size.

Figure 6 is a plot of Equation 18 with the driver based maximum SPL plotted against the system f3. Once again, this is a lot of information. Figure 6 makes it really clear just how much driver size matters; if you want to play louder at a given frequency, you need to go bigger. Figure 6 gives the designer an idea of what to expect from different driver sizes, and how output level will vary with f3 for a given driver size. Per the logic of Figure 4 you can add about 6dB to Figure 6 for a subwoofer driver and take off about 6dB for a pro audio driver. The combination of Figure 5 and Figure 6 give the designer a very quick idea of the port and driver sizes needed to meet a given performance target.

Equation 14 and Equation 18 and their plots in Figure 5 and Figure 6 tell us a lot about a speaker system. If you know the driver and port size, you have a good idea of how loud/low it will play. What is a little unexpected is that Equation 14 and Equation 18 don't say anything about the size of the box or the required input power. They just tell you what to expect from given port and driver sizes, but the box size and input power don't show-up. They clearly matter, we just must go elsewhere to find them.

The Iron Law

Sorry, but I just love that term! It's so in your face. It leaves no doubt that, whatever it says, you are not going to get around it! The term "The Iron Law" dates to work in the 1950s by Josef Hofmann, the "H" in KLH. He observed that with the new "Acoustic Suspension" (i.e., sealed) designs they were developing there was a relationship between the size of the box, the lowest frequencies it could play, and the efficiency. If you want a small box to play low frequencies it will be inefficient: It's "The Iron Law"! Small quantified the Iron Law into a much more universal and useful form when he developed the equations for both sealed and ported systems. His equation for the acoustic efficiency of a ported system is:

$$\eta(f_s, V_{as}, Q_{es}) = \frac{4 \times \pi^2}{c^3} \times \frac{f_s^3 \times V_{as}}{Q_{es}} \quad [19]$$

Where: η is the system efficiency
 f_s is the driver resonant frequency
 V_{as} is the "equivalent volume" of the driver
 Q_{es} is the driver electrical Q
 c is the speed of sound (343.2 m/s)

Equation 19 is a quantifiable version of The Iron Law applied to a ported speaker. But it is not very useful for our purposes without a little modification. Here's another place where our assumption

of a B4 system starts to pay off. If we assume we have a modern low-loss system and choose a box loss of $Q_{bl}=7$ and a woofer mechanical damping of $Q_{ms}=3.0$, we will end-up with the B4 response requiring:

$Q_{bl} = 7.0$	Where: Q_{bl} is the Q of the box due to losses
$Q_{ms} = 3.0$	Q_{ms} is the mechanical Q of the driver
$f_b = f_3$	f_b = box tuning frequency [Hz]
$f_s = f_3$	f_s = driver resonant frequency [Hz]
$Q_{ts} = 0.4048$	Q_{ts} = Driver total Q
$Q_{es} = 0.4679$	Q_{es} = Driver electrical Q
$V_b = 0.942 \times V_{as}$	V_b = Box volume [m ³]
$L_e = 0$	L_e = Voice coil inductance [H]

If we substitute all of the above into Equation 19, we end up with the efficiency of our typical B4 system as:

$$\eta(f_3, V_b) = 2.2 \times 10^{-6} \times f_3^3 \times V_b \quad [20]$$

Equation 20 is the fraction of acoustical power put into the air by the system for 1W of electrical power input to the system. Put in 1W and you get the energy shown by Equation 20. Put in 100W and you get 100 times the acoustic energy shown by Equation 20. It scales linearly. Adding P_e as the input electrical power we therefore arrive at:

$$P_a(f_3, V_b, P_e) = 2.2 \times 10^{-6} \times f_3^3 \times V_b \times P_e \quad [21]$$

Where: P_a is the acoustic power [watts]

We now know the acoustic power generated but are really interested in SPL. The relationship between acoustic power and SPL at one meter from a piston on an infinite plane is:

$$SPL(P_a) = 20 \times \log \left(\frac{\sqrt{\frac{P_a \times \rho \times c}{2 \times \pi}}}{2 \times 10^{-5}} \right) \quad [22]$$

Combining Equation 21 and Equation 22, we get the SPL produced by our B4 system as:

$$SPL(f_3, V_b, P_e) = 55.62 + 20 \times \log \sqrt{f_3^3 \times V_b \times P_e} \quad [23]$$

We now have our sought-after relationship between SPL, box size, and power. But it is still not in its most useful form. We want to set Equation 23 equal to Equation 18 that defined the maximum SPL based on the driver. Doing so and then solving for the power gives us what we are looking for:

$$P_e(f_3, d, V_b) = 7.76 \times 10^{-7} \times \left(\frac{\left(\frac{d^3}{2} + 2 \times d^2 \right)^2 \times f_3^3}{V_b} \right) \quad [24]$$

Equation 24 is what we've been working toward. It is what is going to help us constrain our B4 design to a given box size and required input power.

Next Month

This concludes the main analysis portion of the article. Equations 14, 18, and 24 tell the story we set out to tell. We'll follow-up in Part 3 with an example to see how this all works together. We'll also address some practical considerations that will make it clear that not all theoretically allowable solutions to our equations are practicably possible. We will determine the cut-off where a desirable design that is not practicable with a port can be easily realized by using a PR. A set of Excel spreadsheets for you to automate these calculations will be available as supplementary material online. 

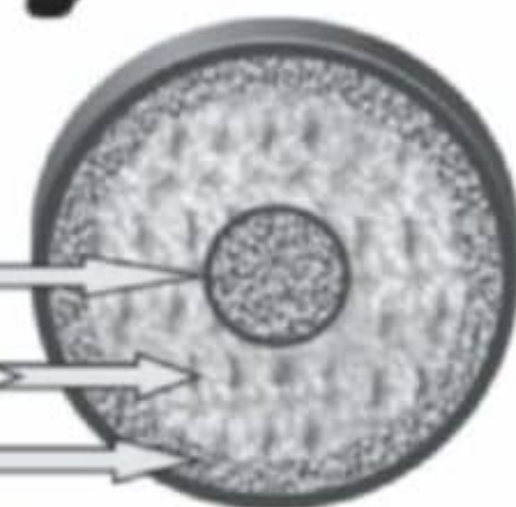
About the Author

Gary Gesellchen is a Mechanical Engineer with a background in design and manufacturing. He built his first pair of speakers in high school and has been hooked ever since. Other interests include being in the mountains with nature, travel, and time with family and friends.





Coax Is Cool Again!



Cardas Golden Strand
Air Filled Cotton Dielectric
Unique Spiral Litz Shield
ETI Tellurium Copper Connectors

Affordable Litz Interconnects and Bulk Wire
Tonearm - Phono - Line - Loudspeaker

KAB
Preserving The Sounds
Of A Lifetime

www.kabusa.com

Visiting the Western Electric Vacuum Tube Factory

Planned for some time, *audioXpress* was finally able to visit Western Electric in Rossville, GA. Officially established in 2018, this facility has been continuously producing 300B single-ended triode tubes since 2020 and is already expanding production to other vacuum tube models. Richard Honeycutt shares his guided tour.

By

Richard Honeycutt

Most of us learned in grade school that Lee de Forest (**Photo 1**) became the second person to apply the “Edison Effect” when he invented the Audion vacuum triode about 1906. (The first was John Ambrose Fleming, who invented the thermionic rectifier known as the “Fleming valve” in 1904.) Less well-known is that in 1912, Fritz Lowenstein and John Stone demonstrated to Bell Telephone Co. officials the Audion’s potential as a practical amplifier for telephone repeaters in long-distance transcontinental and intercontinental telephony. As a result, Western Electric bought de Forest’s US patent rights to the Audion in 1914, subsequently developing it into a more useful and reliable device.

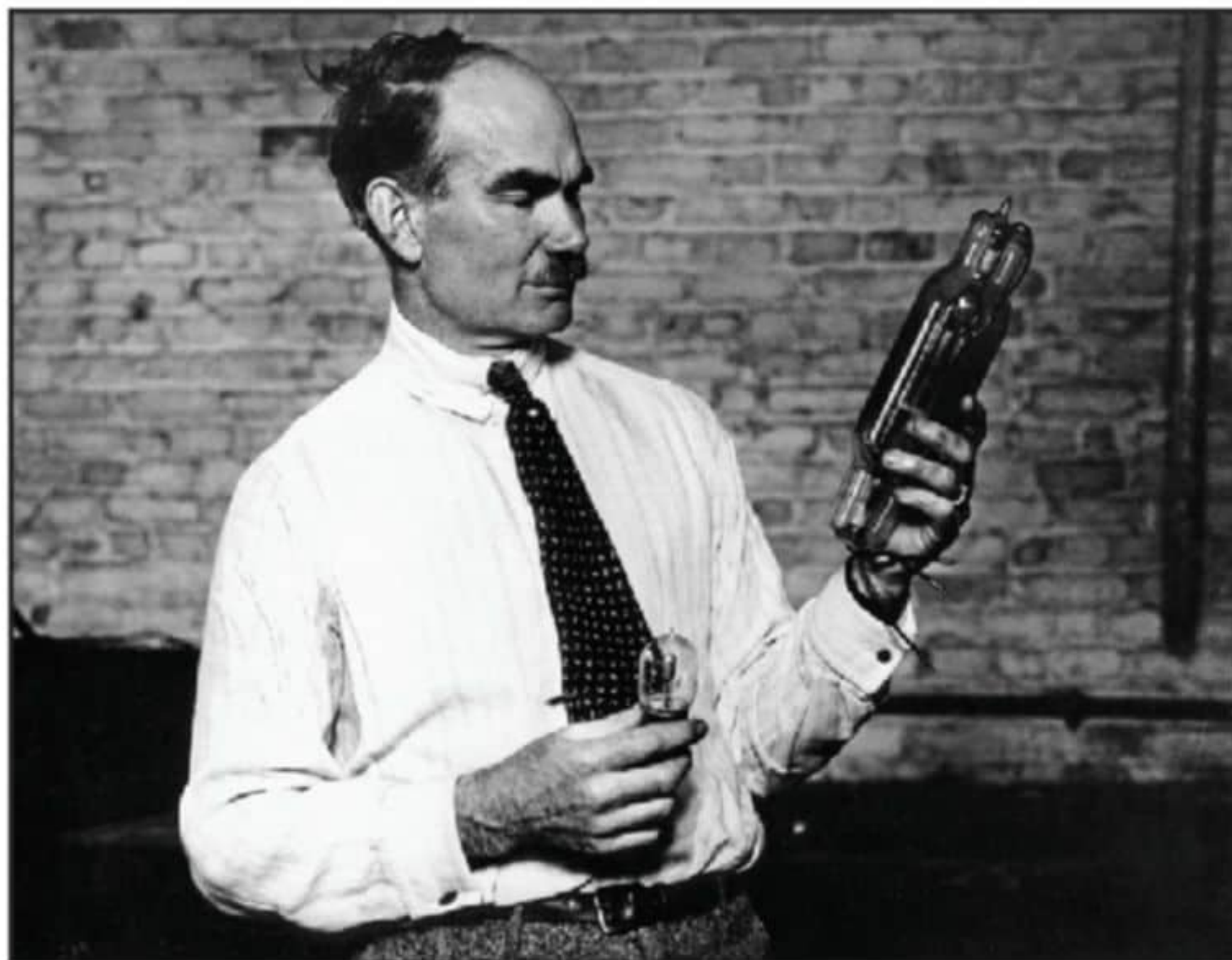


Photo 1: Lee de Forest holds two different models of Audion tubes.



The final 300B tube in its current packaging, as manufactured at the Rossville Works.

However, Audion amplifiers had erratic performance and provided insufficient output in their original form. H. D. Arnold of the Bell Telephone Laboratories noticed that attempting to increase the Audion’s output by raising the plate voltage caused a blue glow, suggesting that the vacuum was incomplete, allowing ionization of remaining gas molecules in the tube. By manufacturing the triode using a high-vacuum pump, Arnold improved its performance enough that it became the basis of amplifiers used in transcontinental telephony, which were placed into service in 1914-1915.

Further improvements pioneered by Arnold included the design of a high-emissivity filament made of platinum with a baked-on coating of alkaline-earth oxides. These improvements were incorporated into the Western Electric type 101 vacuum tube. As further improvements were made, letters were added to the type number. By 1926, the 101D had been in use in telephone repeaters for 12 years, and it was the most reliable and durable type of tube yet produced. Unlike today’s ubiquitous 12AX7 with its μ of 100, the 101D had a μ of 6.2. Complete data for the Western Electric 101D is available online at: <https://frank.pocnet.net/sheets/084/1/101D.pdf>). Western Electric’s development of other types of tubes is detailed in a *audioXpress*, December 2022 Hollow-State Electronics article.

The most famous Western Electric tube among audiophiles is the legendary 300B power triode, introduced in 1938. Used in the Model 91 and 92 power amplifiers for cinema theaters, the 300B was manufactured in the Western Electric Kansas City Works (**Photo 2**) from its opening in 1961 until 1988. However, demand for the 300B continued all around the world. New-old-stock (NOS) tubes could demand premium prices, and a few other manufacturers introduced clones of the famous tube. In 1992,

Charles G. Whitener of Westrex (the Western Electric name in film sound) entered into negotiations with AT&T to acquire intellectual property and a license to manufacture and sell vacuum tubes and audio equipment under the Western Electric name. After successful completion of the negotiations in 1995, AT&T granted Whitener the patent rights and rights to manufacture tubes and audio equipment under the Western Electric brand. Also, since 2016, the trademarks and all associated intellectual property are owned by Western Electric under Charles Whitener.

The KC Works plant was not sold to Whitener, but Western Electric shared the space with AT&T, and former employees of that plant were hired to resurrect the authentic Model 300B tube. The new Western Electric 300B was demonstrated to electronic equipment manufacturers at the 1996 Consumer Electronics Show, and new 300Bs began shipping in 1997. To meet customer demand, the new Western Electric Co. opened a new plant in Huntsville, AL, in 2003; much of the old KC Works machinery and tooling were moved to Huntsville. Besides 300Bs, the plant also produced 6EJ7, 577W, 576A tubes, as well as test sets for the U.S. Naval Surface Warfare Center. In 2004, Western Electric introduced new high-fidelity audio equipment including amplifiers and the 203C CD player.

The Rossville (GA) Works factory (**Photo 3**) was established in 2018 to modernize 300B manufacturing, introduce additional tube lines, and house research and design of new Western Electric products. Some highly specialized equipment from the KC Works plant is now in service in the Rossville Works.

The Rossville Tube Factory

Early in April of this year, I was granted the privilege of touring the Rossville works. There, I was able to see how the authentic 300B (with some improvements compared to the original) is made, using some of the latest manufacturing techniques along with some of the classic equipment mentioned above. We'll start by showing two workers making Western Electric tubes 80 years apart (**Photo 4**). The original specifications for the 300B are detailed in **Photo 5**.

The first critical operation is high-temperature cleaning of some parts in hydrogen furnaces. **Photo 6** shows a control grid being formed. The cathode and anode are die-stamped from nickel metal. Then the cathode is coated with graphene (a monatomic layer of carbon) to improve thermal and electrical emissivity and conductivity.

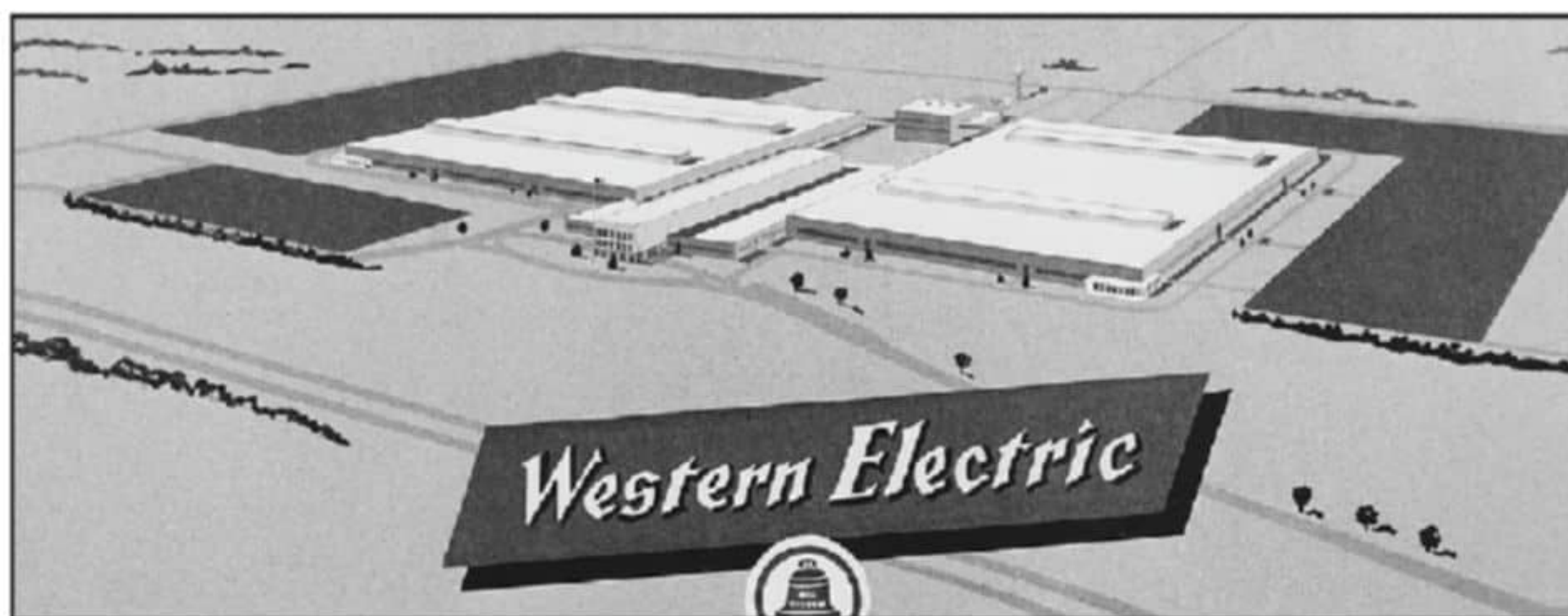


Photo 2: The Western Electronics Kansas City (KC Works) made tubes from 1961 until 1988.



Photo 3: The Rossville (GA) Works is now Western Electric's center for manufacturing, engineering, and development.



Photo 4: 300B manufacturing at Hawthorne Works around 1942 (a) and at Rossville Works in 2020 (b).

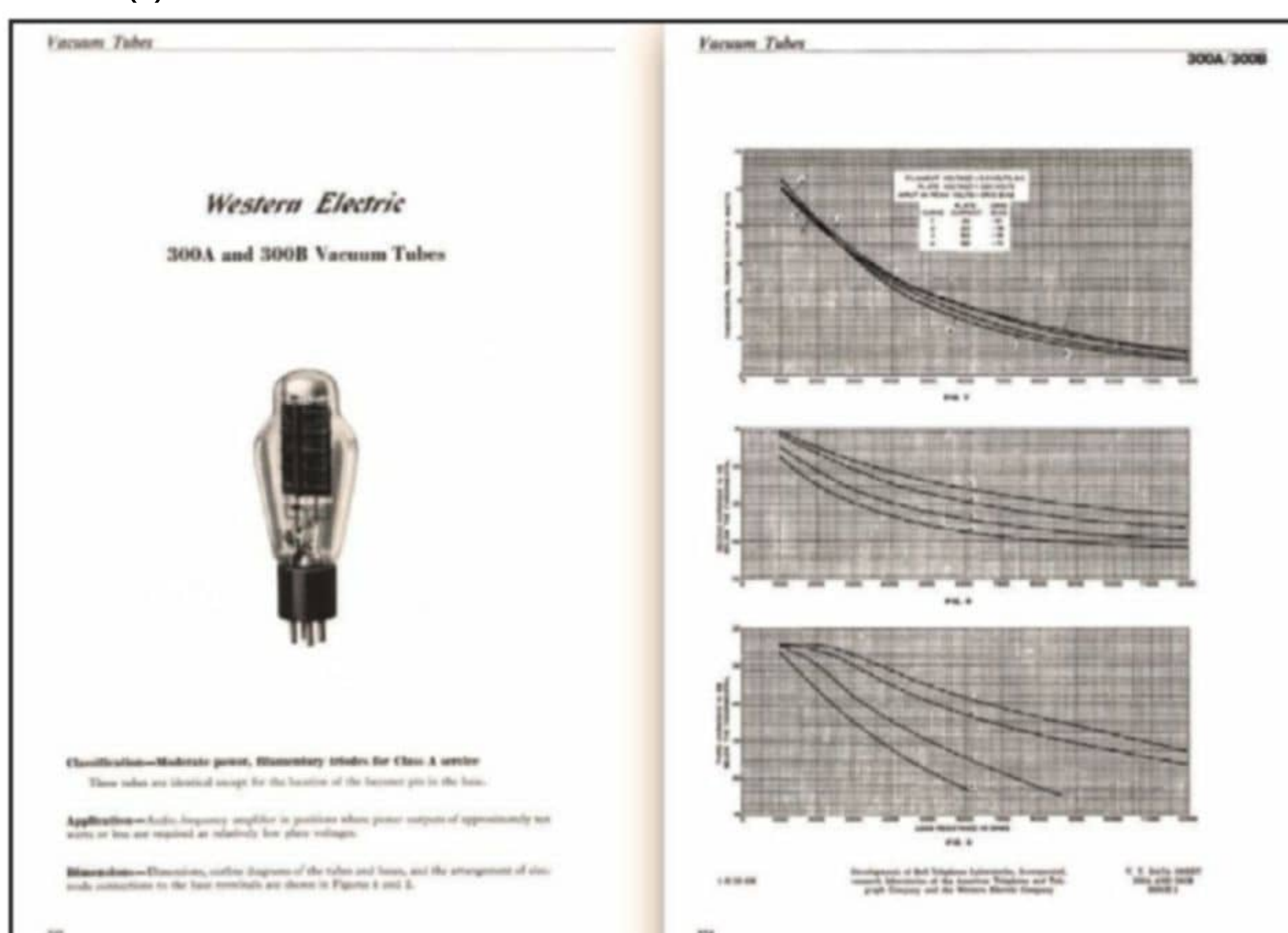


Photo 5: It all begins with the original drawings and documentation, which Western Electric has maintained since 1938.



Photo 6: The grid is formed using a special grid lathe machine that Rossville Works carefully preserves.

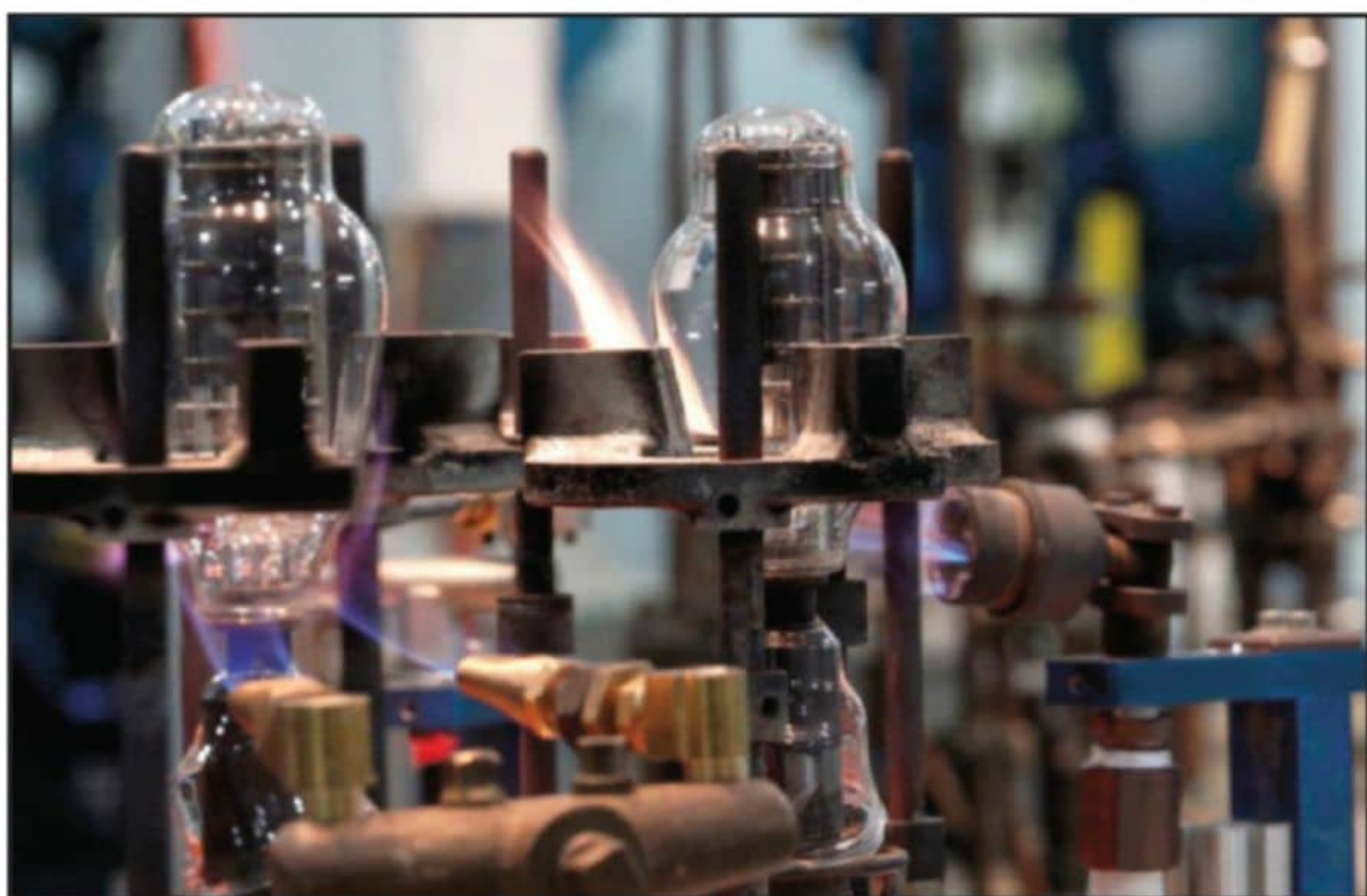


Photo 7: Glass is being extruded for the base of the tube mount.

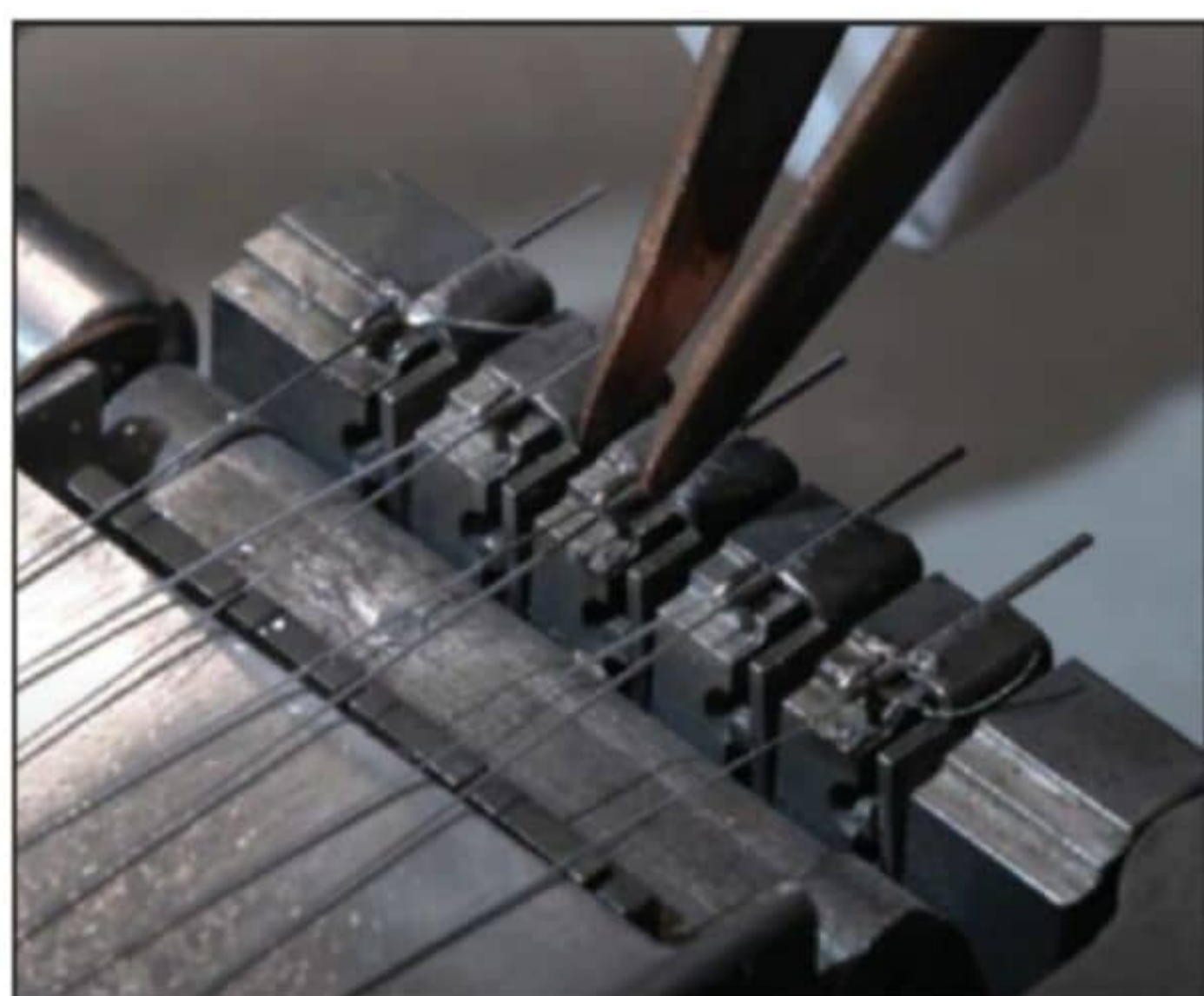


Photo 8: The filament connections are welded.

Resources

Western Electric 300B Factory, YouTube, <https://youtu.be/PA8G5zRjriI>

Each glass part is fitted on location. In **Photo 7**, we see glass tubing extruded in the flare machine. The glass is heated to melting point and the machine spins the flare out. This flare, along with the stem and other glass parts, will eventually become the base of the tube mount.

In a separate sequence of operations, the filament is produced. Filament production is a very special and carefully guarded process that gives Western Electric tubes their long life and reliability. The filament itself is coated with a proprietary chemical via a special system and taken up onto a large reel. In **Photo 8**, a tweezer welder is used to weld the filament connections.

The mount is the functional part of the tube, or the electronics of the tube. A completed mount is assembled and ready to be sealed into the glass envelope (**Photo 9**). **Photo 10** shows the completed sealed envelope.

The un-evacuated tubes are placed into proprietary induction furnaces to heat the internal parts to about 1000°C, to evacuate all gas from the glass envelope. As the vacuum is being pulled, the tube is “tipped-off,” meaning the straw-like tubulation is fused together to form a permanent vacuum seal. Western Electric vacuum pump technology is far superior to that of other manufacturers producing replica 300Bs. During the same evacuation process, the filament is activated so that it can begin to emit electrons. Evacuated tubes are then “flashed.” This is an induction process by which intense RF is run through copper coils around the tube. The getter ring is excited, coating a portion the glass envelope’s interior to form the silver getter. The getter’s purpose is to bind any extraneous gas molecules that may remain in the tube. Were these not bound, they would ionize during the tube’s operation and produce noise in the output signal.

The leads are inserted into the base, after which the leads are trimmed. Glue is applied to secure the base to the envelope, and then cured. After this operation, an experienced Western Electric employee visually inspects the tube, rejecting any found to have defects (**Photo 11**). Finally, to age the filament (for operational stability) and continue its activation, batches of tubes are placed on aging racks (**Photo 12**). Filaments are brought to various voltages and B+ voltage is applied to the plate.

The Future

The 300B is not the only tube Western Electric makes. In addition to the 12AT7s (ECC81s) used in the new Model 91E stereo amplifier, the Rossville Works produces several OEM tubes for highly respected audio manufacturers worldwide, as mentioned earlier. In addition, preparations are underway for manufacturing other high-demand audio tubes, beginning with the

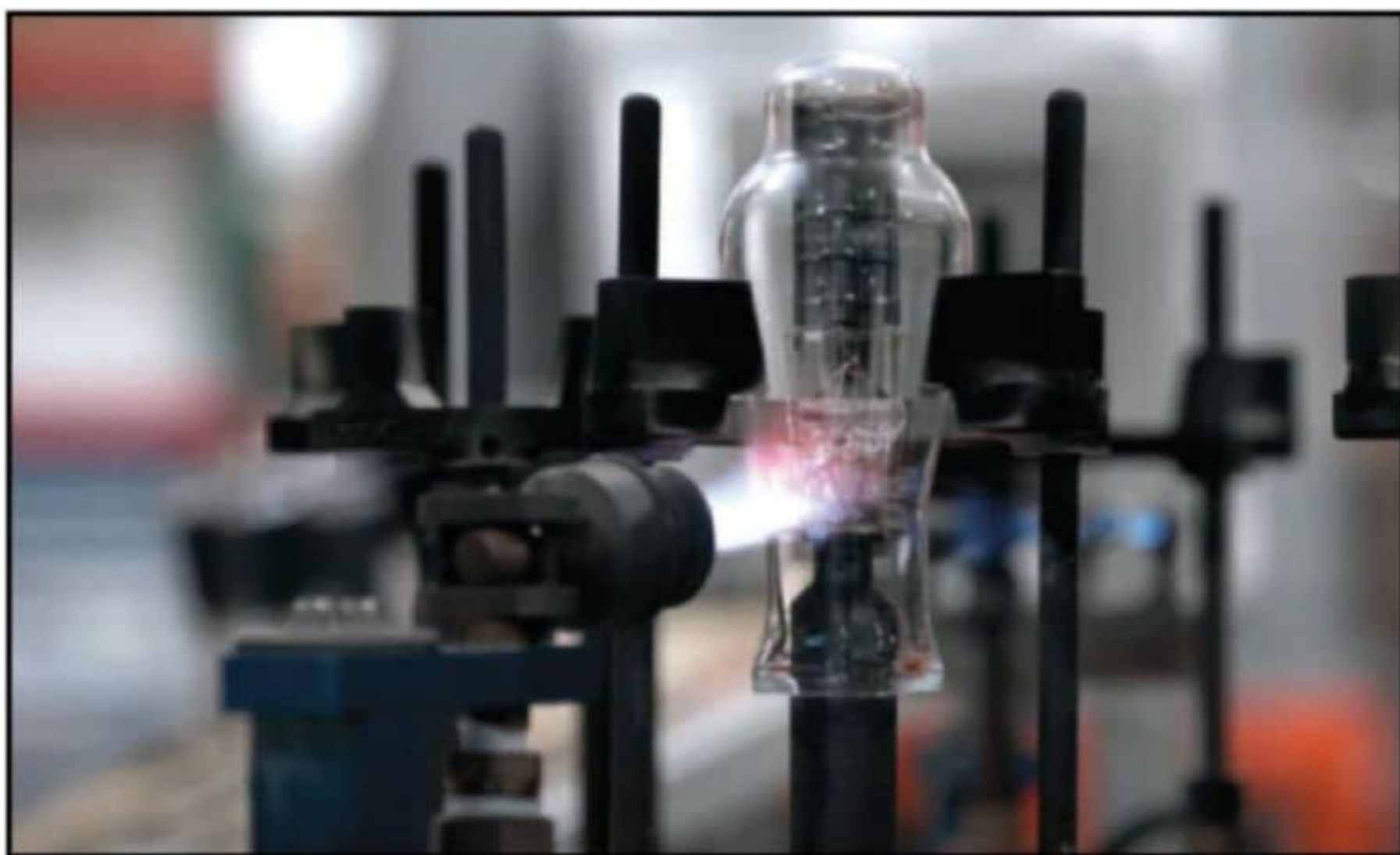


Photo 9: The elements are assembled into the glass envelope, ready to be sealed. The envelope is sealed, and the excess glass drops away. (Photo courtesy of Klipsch Museum of Audio History, Jim Hunter, Curator)



Photo 10: The sealed envelope is ready to be evacuated.

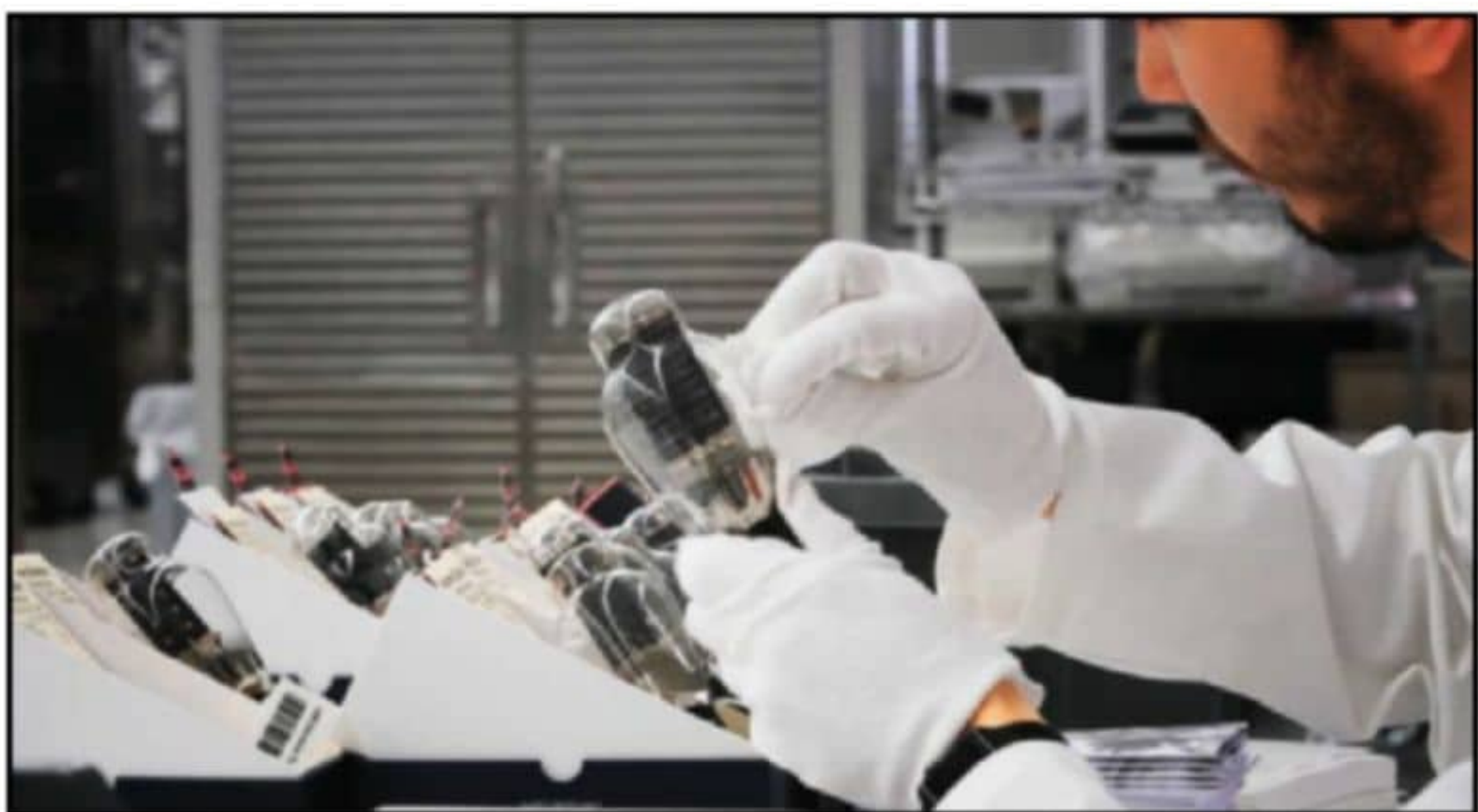


Photo 11: Each tube is visually inspected before aging.



Photo 12: Racks of tubes are aged as the final step. (Photo courtesy of Klipsch Museum of Audio History, Jim Hunter, Curator)

12AX7 and the 12AU7 models, with power pentodes and beam power tubes to be expected later.

Preparations are also underway to produce new Western Electric 308B power triodes. This 13 5/8"-long tube is rated at 50W undistorted power output per tube when used in a Class-A configuration. **Photo 13** shows Western Electric President Charles Whitener comparing a 308B and a current production 300B tube.

For many decades, Western Electric was the sole supplier of electronic equipment for AT&T, the world's largest telephone company. From the installation of the first undersea cable repeaters in 1914-1915, reliability was a prime concern for Western Electric engineers. (Replacing parts in an undersea amplifier is not an inexpensive proposition!) During the 1940s, when tube replacement for home radio receivers was so common that most drug stores and "dime stores" offered supplies of replacement tubes, and in the 1950s when half of the 20-ish tubes in a TV receiver would need replacement before the end of the set's (typically 10-year) life, Western Electric was designing telephone and military equipment for a minimum lifetime of 40 years. The same processes that gave Western Electric tubes long life and reliability are still performed with all Western Electric tube manufacturing today. Continuing the high-quality, high-reliability manufacturing standards is central to the company's continuing mission. 



Photo 13: Western Electric's president and CEO, Charles Whitener—seen here holding a 300B tube and a 308B tube—welcomed *audioXpress* Hollow-State Electronics columnist Dr. Richard Honeycutt on a tour of the modern Western Electric Rossville, GA facility.

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.



Instrumentation Amplifier-Based Preamplifier Line Stage

The traditional three op-amp instrumentation amplifier architecture is seldom used in home audio equipment, even though it has a lot of desirable characteristics: high-impedance, balanced inputs; high common mode rejection (CMR); stable gain (which can be very high, if desired); low noise; and low output impedance. This article describes a line stage preamplifier that also includes time-delay-matched, fully balanced (voltage and impedance) outputs and adjustable gain from -51dB to +12dB in precise 1dB steps.



By
Bruce Gillingham

Photo 1: Here is a front view of the PA2 Preamplifier.

My PA2 preamplifier (**Photo 1**) was completed in 2002 and remains in service today without modification other than the addition of two phono preamp modules (**Photo 2**).

The Input Line Stage

Figure 1 shows a simplified schematic of the line stage. Input buffers A1 and A2 are configured as in a standard instrumentation amplifier. A3 with resistors R3, R4, R7, and R8 form the standard difference amplifier output stage. The second

difference amplifier built around A4 with resistors R5, R6, R9, and R10 is connected in parallel with the standard one, but with the inputs switched. A4 provides the inverted output without incurring the additional delay and degradation associated with simply inverting the positive output.

Both outputs have full CMR and can be used independently. Placing the volume/balance control between the two stages provides both input and output buffering for them. It lowers output noise by improving the gain distribution. The input stage has a differential-to-differential gain of 6dB. The output stage has a differential-to-single-ended gain of 6dB and a differential-to-differential gain of 12dB. The attenuators are settable from 0dB to 63dB. Two attenuators are required as the signal is balanced at this point. They must be matched to maintain good CMR.

Figure 2 shows a simplified schematic for one attenuator. Six log-weighted attenuator stages are cascaded. Each stage is switched in and out by a military-grade, hermetically sealed relay with gold-plated contacts. The relays are double pole double throw (DPDT) and switch both the positive and negative path attenuators. Additional stages could be added for extended range and/or finer resolution. I found 1dB steps to be adequate for the volume and balance functions. A gain of -51dB



Photo 2: This is the removable dual stereo phono preamp module. This module also employs a differential instrumentation amplifier architecture.

is low enough for any reasonable listening level and the preamp has other means of fully muting its outputs.

The “secret” to cascading attenuators in any number or order is to design them with source and load impedances in mind. Home audio does not require matched transmission lines. The distances within a component and even between components is typically short and audio wavelengths are quite long. It is only necessary to match the impedance on one side of the attenuators to allow cascading. If one assumes a fixed load resistance (R_L in **Figure 3**), just two resistors (R_A and R_B) are required to provide the desired stage attenuation (A) and still present the value of R_L to the preceding stage. Equation 1 shows the calculation of attenuation (linear) for the circuit in Figure 3. Equation 2 shows the calculation for its input resistance, R_{IN} . It is desired that this equal R_L .

$$A = \frac{V_{OUT}}{V_{IN}} = \frac{\frac{R_B R_L}{R_B + R_L}}{R_A + \frac{R_B R_L}{R_B + R_L}} \tag{1}$$

$$R_{IN} = R_A + \frac{R_B R_L}{R_B + R_L} = R_L \tag{2}$$

Since the values of A and R_L may be selected by the designer, we have two unknowns (R_A and R_B) and two simultaneous equations, resulting in Equation 3 and Equation 4.

$$R_A = (1 - A) R_L \tag{3}$$

$$R_B = \frac{A}{(1 - A)} R_L \tag{4}$$

For decibel-weighted attenuation steps, we must convert the attenuation in decibels (dB) to its linear equivalent using Equation 5.

$$A = 10^{\frac{-dB}{20}} \tag{5}$$

In the PA2, the six attenuator sections are 1dB, 2dB, 4dB, 8dB, 16dB, and 32dB. Various combinations provide the desired attenuation.

It may not be obvious that attenuators designed for a specific load resistance can work with the single op-amp, four-resistor, differential amplifiers in the output stage. These amplifiers have a fixed input impedance to ground on the non-inverting input, but not on the inverting input. The inverting input impedance varies with the signal level on the non-inverting input. I made some simplifying assumptions. I did not include the input stage

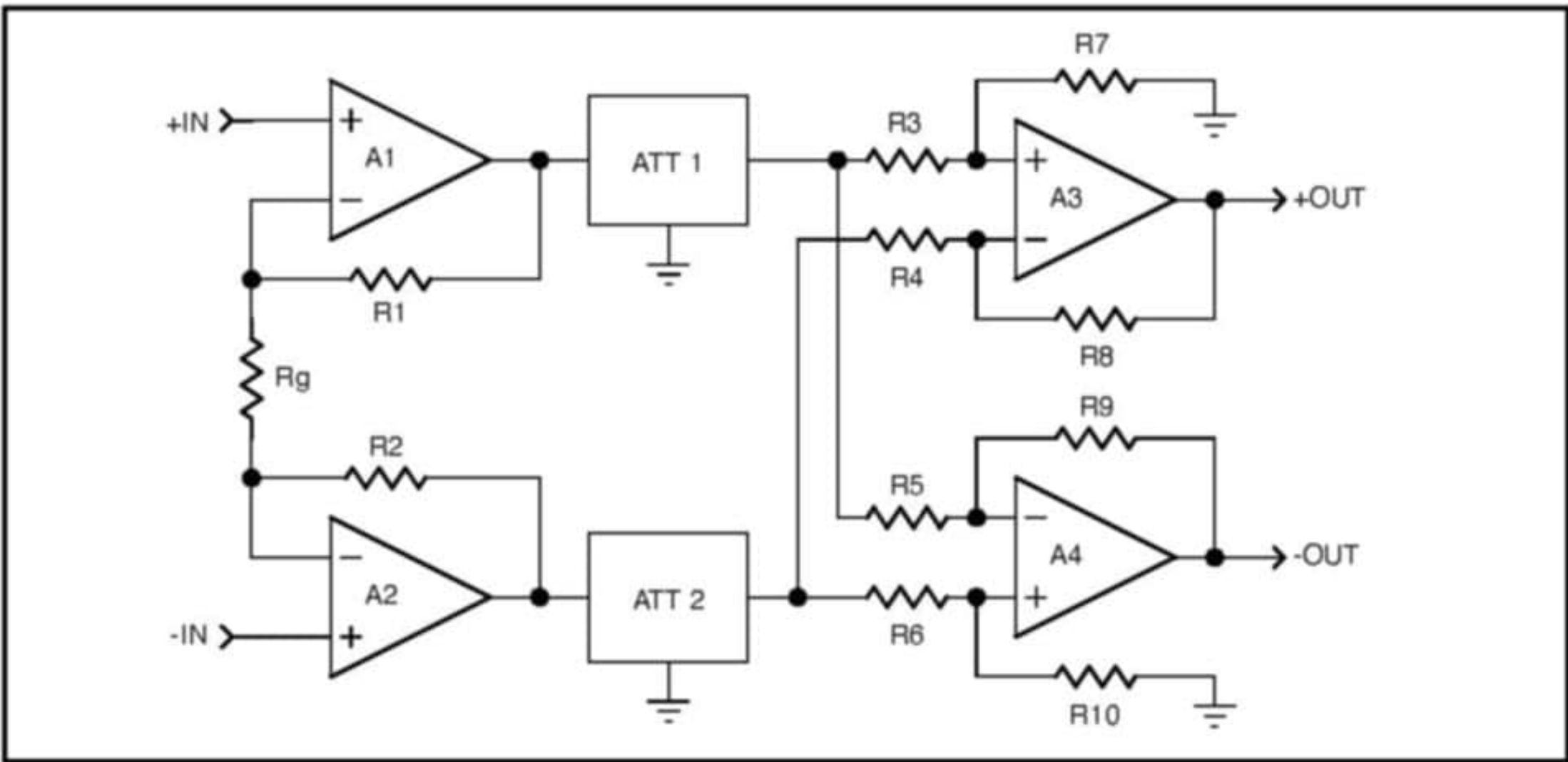


Figure 1: This simplified schematic shows the PA2’s line stage.

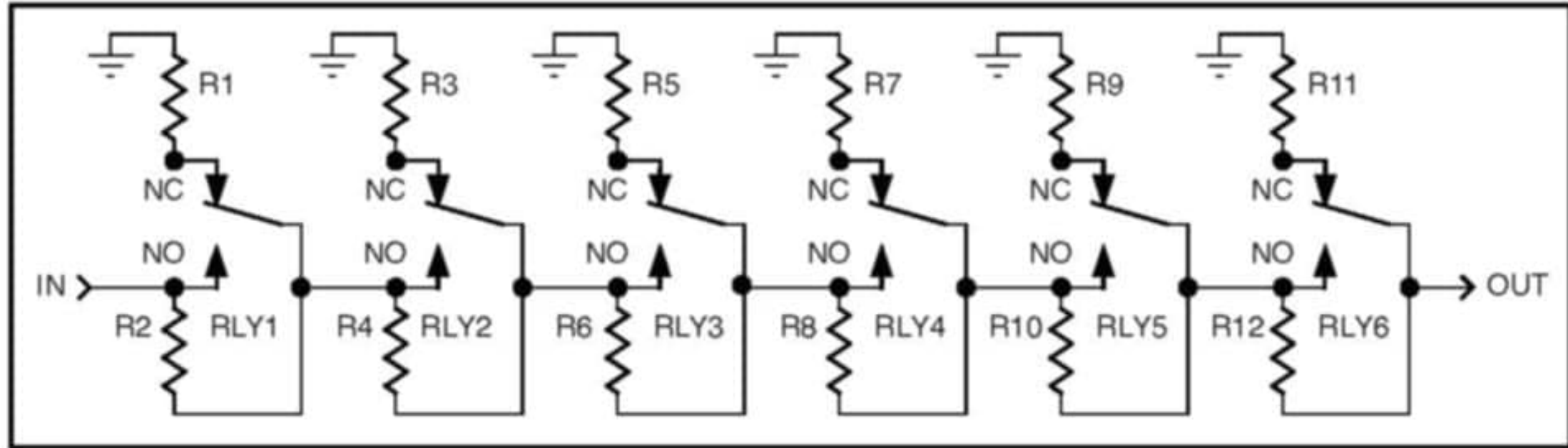


Figure 2: Here is the schematic for the digitally controlled attenuator.

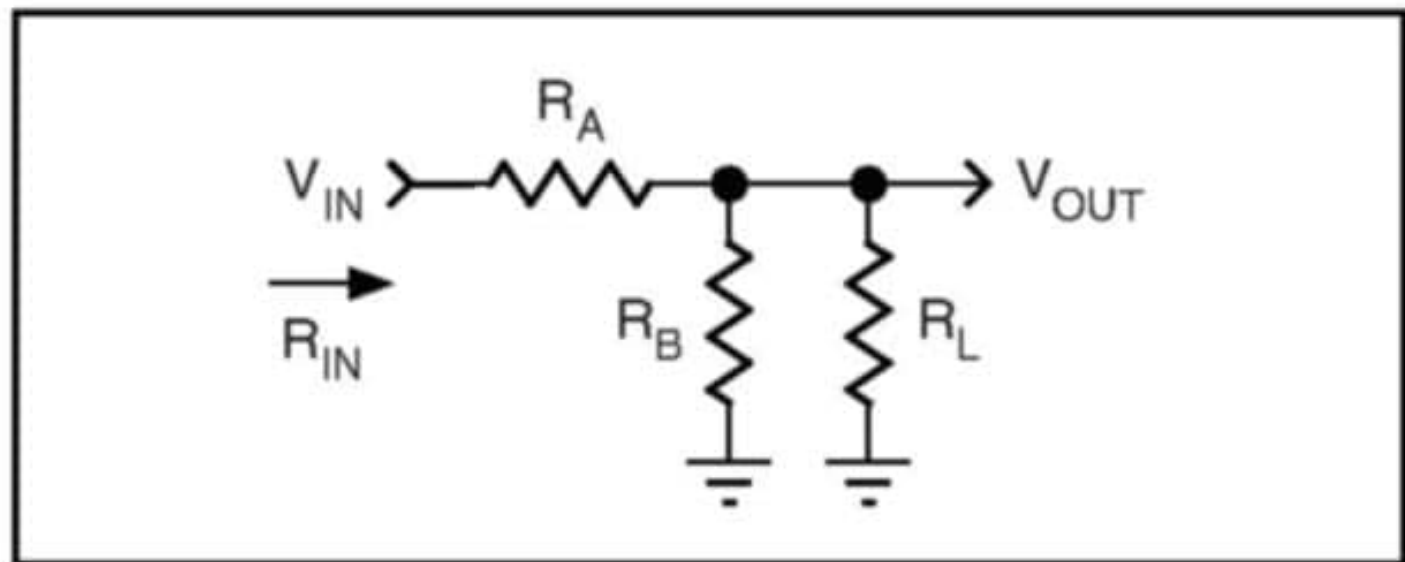


Figure 3: This schematic shows the attenuator stage.

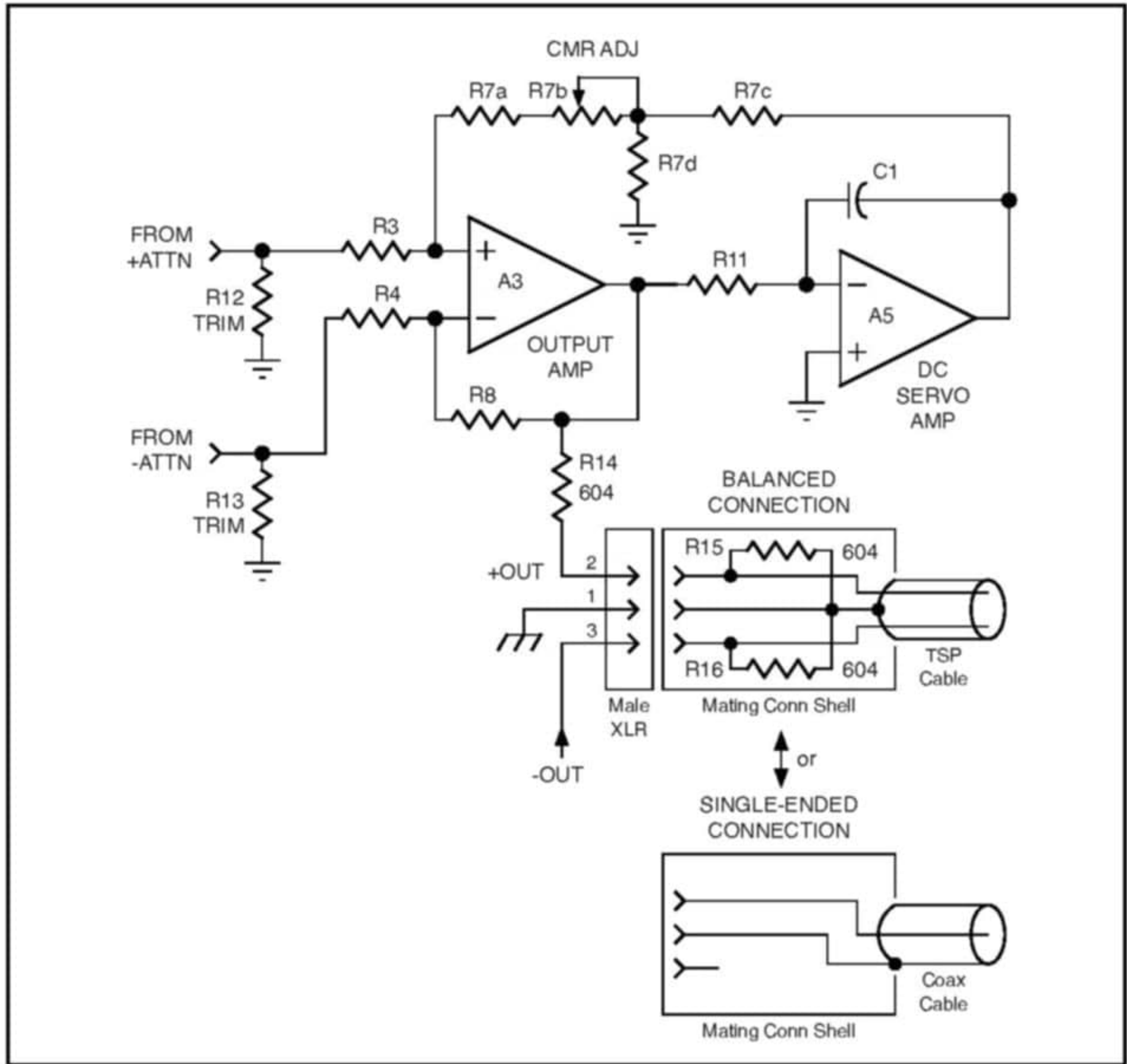


Figure 4: The schematic shown here details the non-inverting output amplifier.

as it can easily be added back in later. I set R3 through R10 = $2R_L$ and included only one -6dB attenuation stage. The analytical solution shows that the differential output voltage is equal to the differential input voltage with these assumptions. It works with any degree of imbalance between the non-inverting and inverting inputs. With a fully balanced input (impedance and $V_{-IN} = -V_{+IN}$), each attenuator “sees” the desired load impedance (R_L) to ground. This explains why it works for this special case. Why does it also work with partially balanced and single ended inputs? The answer is balance within the line stage. Although the dual attenuators present a variable source impedance to the output stage, this impedance is balanced. Since an op-amp forces the voltages on its inputs to be the same, there is a virtual short between them. A3 has a differential input impedance equal to $R_3 + R_4$. A4 has a differential input impedance equal to $R_5 + R_6$. These impedances are in parallel. With R3 through R6 all equal to twice the desired load impedance (R_L), the attenuators see a floating load of $2R_L$, which allows the math of equations (1) through (5) to work.

Attenuator Considerations

For safety, the normally connected relay contacts are used to implement attenuation. The relays must be energized to increase volume. To preclude momentary bursts of higher volume, the drivers enforce a slower pull-in time (20ms), then drop-out (10ms).

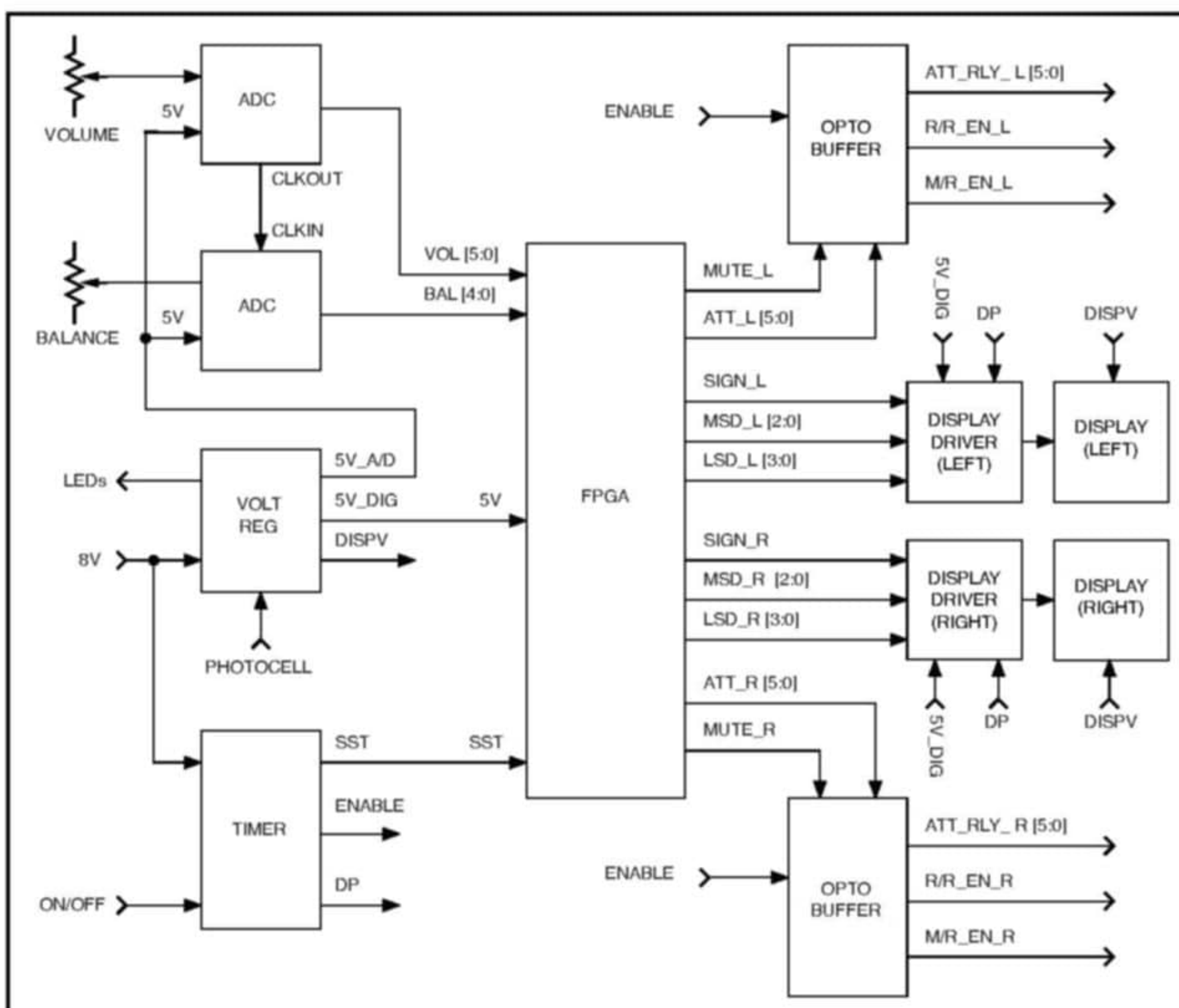


Figure 5: The block diagram for the PA2's controller is detailed here.

Whenever gain is switched in discrete steps, a transient is generated. The audibility of the transient is a function of step size, signal amplitude, and other factors. One mitigating characteristic of relay switches is that an intermediate level of attenuation is introduced as the armature moves from one contact to the other.

Another is that the switching of the relays is naturally staggered due to tolerances. With no input, there are no switching noises. With a constant sine wave input, volume changes introduce a mild, but obvious “zipper” noise that some could consider annoying. With music, the zipper noise is much less audible. In normal use, the switching noise did not merit introducing the complexity and possible distortion of a smooth muting function during switching.

Resistor Considerations

Trades must be made selecting the resistor values. To minimize noise, resistor values should be as low as possible. But the resistors should be large in comparison with the relay contact resistance. Also, the op-amps must be able to drive the resistors without significant distortion. I settled on selecting R_L to be 1.2kΩ. Resistors R3 through R6 should then be 2.4kΩ. The closest standard 1% value is 2.43kΩ. The slight error this introduces is trimmed by 93.1kΩ shunt resistors to ground on the output of each attenuator (186.2kΩ differential load). R1 and R2 are 1.21kΩ; R_g is 2.42kΩ for a differential-to-differential gain of +6dB in the input stage. R7 through R10 are 4.87kΩ to achieve a differential-to-single-ended gain of +6dB in the output stage.

Figure 4 shows the output stage amplifier design. Only the non-inverting amplifier (A3) is shown. R12 and R13 are the trim resistors. R7 (Figure 1) consists of four resistors (R7a–R7d) shown in Figure 4. R7b, a potentiometer, allows optimization of CMR. R7c and R7d attenuate the output of the DC servo amplifier A5 to reduce its noise contribution. With the selected op-amps, the servo is not strictly needed. It was included to remove any DC offset on the preamp's inputs.

For stability into capacitive loads, there is usually an output series resistor. This is R14 (Figure 4). Typically, the value of this resistor is in the 50Ω to 100Ω range. Larger values provide additional isolation from the load; smaller ones maximize bandwidth. I chose 604Ω; primarily, I wanted to be able to drive a matched load without over-taxing the amplifier's current capacity or generating excessive distortion. Another reason is nostalgia for the old 600Ω standard audio interfaces. The closest 1% standard value is 604Ω.

I had another reason for “standardizing” on 600Ω output interfaces for the PA2: I wanted the

preamp to present all loads with the same source resistance and unloaded gain. The advantages are consistent volume and bandwidth and flexibility.

All the PA2's inputs/outputs are fully balanced on standard XLR connectors. The balanced and single-ended/unbalanced load cases are accommodated by differences in external cabling. No additional amplifiers are required. I found this approach to be esthetically pleasing.

Single-ended loads are connected with a coaxial cable terminated with a female XLR connector on the preamp end. The cable's "hot" center conductor is connected to either pin 2 (non-inverting) or pin 3 (inverting) as desired. The cable shield is connected to pin 1, grounded to the preamp chassis as is safety ground. Preamp signal ground is connected internally to chassis ground at one point only. Single-ended loads "see" a 604Ω source with a maximum gain of +12dB.

Balanced differential input power amplifiers typically have a high input impedance and use both preamp output polarities. If one uses a standard balanced cable, the load will see a 1208Ω differential source resistance and a maximum gain of +18dB. This is undesirable. My solution was a 604Ω load resistor on each output phase within the shell of the connector that mates to the PA2. These are resistors R15 and R16 (Figure 4). They reduce both the gain and source resistance by a factor of 2, matching single-ended interfaces. The attenuator formed with the cable reduces both signal and noise and does not harm CMR when implemented at the source end.

Op-Amp Considerations

The choice of op-amps for the line stage is crucial for high performance. Many parameters are important (GBW product, slew rate, distortion, noise, bias current, offset current, offset voltage). Field-effect transistor (FET) input op-amps are desirable for A1 and A2. Low bias current reduces input noise current and DC offset; high input impedance improves CMR with less than perfectly balanced sources. The choices for A3 and A4 are less restricted as the source impedances seen by them are low. I did an extensive investigation at the time and the best available choices were the OPA627 for A1 and A2 and the AD797 for A3 and A4. Both devices have excellent AC and DC performance. Even 25 years later, these devices are no slouches.

Attenuator Control

The PA2 has rotary Volume and Balance controls and a six-character, seven-segment LED display that shows the gain (in decibels) of each channel. Balance is accomplished by reducing the gain in one

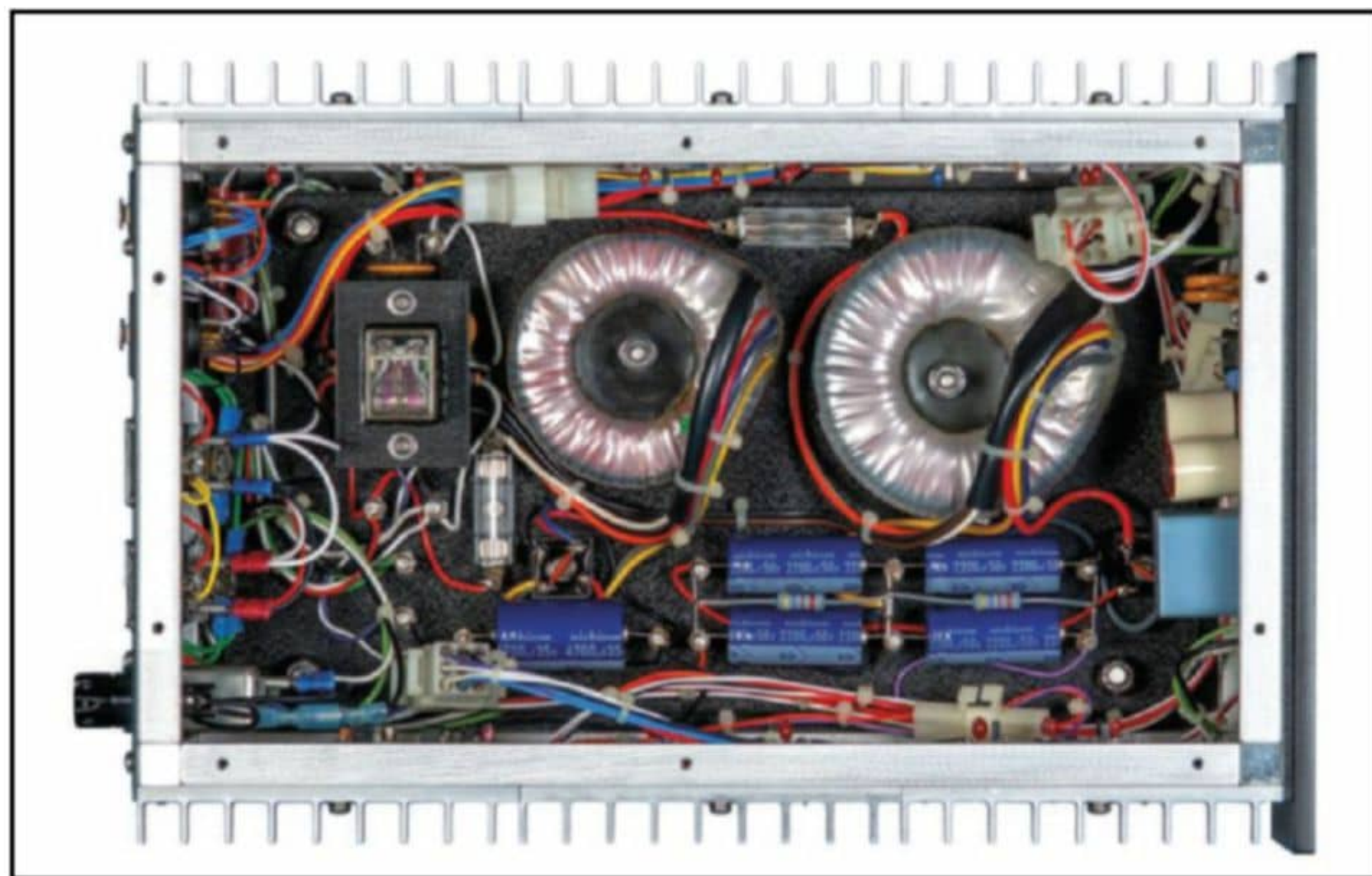


Photo 3: The remote power supply is pictured from the top with the top cover removed.

channel or the other. Turning the balance control to an extreme fully mutes the attenuated channel. All outputs are muted for ~12 seconds after turn-on to allow everything to stabilize. During this period, the display shows all decimal points illuminated to indicate the warm-up muting state.

To control the attenuators and the display, I chose a conventional linear taper potentiometer for the volume and balance controls. They are biased with a DC voltage and their wiper voltages are digitized by 8-bit analog-to-digital converters (ADCs). One issue with this approach must be addressed in the detailed design. If the ADC's input voltage is on, or very near a quantization threshold, the output word can switch back and forth between two values due to noise on the input. Filtering the ADC's input helps but does not resolve the problem.

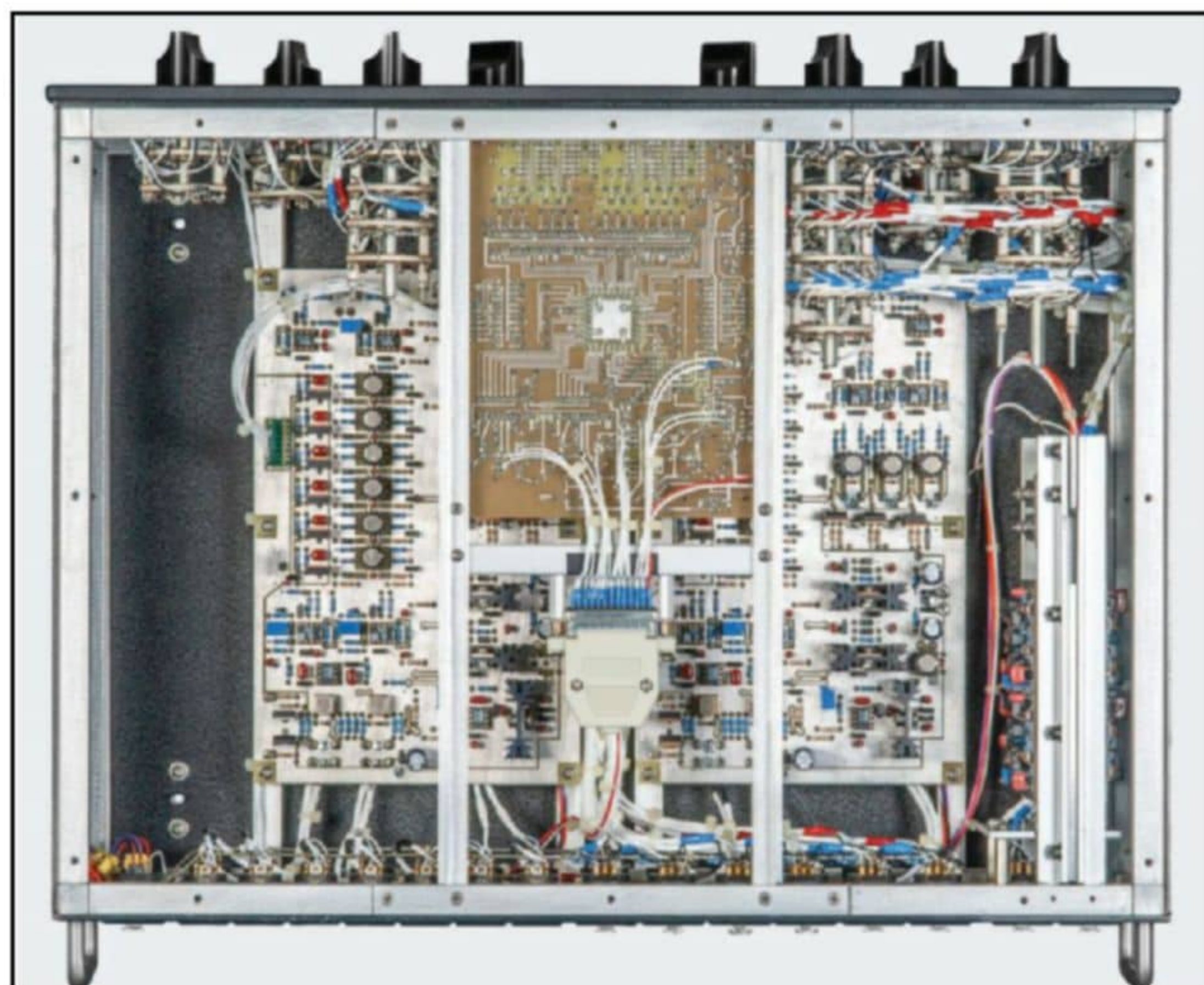


Photo 4: This is the PA2 with the top cover removed.

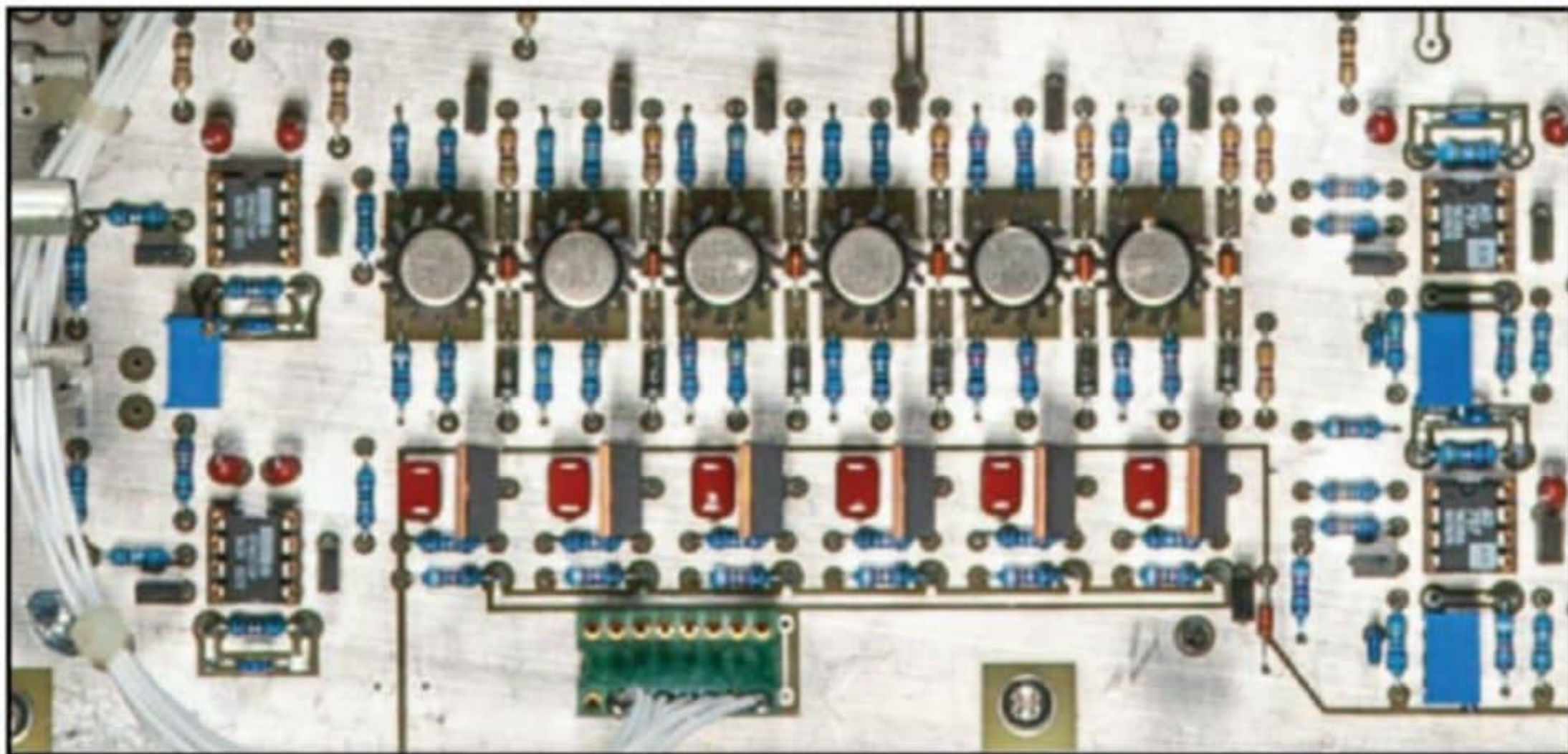


Photo 5: Here is the line amp detail. The upside-down Control Circuit Card Assembly (CCA), located top center, has the circuitry shown in Figure 5. Each channel has its own Amplifier CCA mounted on rails near the bottom of the chassis.

In the best case, this behavior is like contact bounce. In the worst case, there is gain oscillation. To avoid this, I introduced hysteresis (positive feedback) as is done with comparator circuits. The volume control is quantized with the 6MSBs of an 8-bit ADC. Bit 7 is applied to a simple 1-bit digital-to-analog (DAC). If the bit is 0, one-quarter of this bit value is added to the voltage from the potentiometer. If the bit is 1, one-quarter of the bit value is subtracted from the potentiometer voltage. This keeps the input voltage away from the threshold and eliminates instabilities on slowly changing input voltages. The balance control implementation is similar, but Bit 6 is used for hysteresis. My friend, Bob Smith, suggested this approach and I analyzed and implemented it.

Figure 5 shows the controller block diagram. Six bits of volume data and five bits of balance data are provided to a small field programmable gate array (FPGA). The FPGA computes the necessary control bits for the relay and display drivers. Another feature in the FPGA is “soft-start” (SST). An additional 16dB of attenuation is added to the volume setting for a few seconds after the warm-up muting relay kicks in. This provides a warning to the operator that the volume may be set too high and avoids unpleasant surprises. I sought help with the logic design from another friend, Bill Hartman, who quickly designed and programmed the FPGAs from an interface document that I generated. At the time, the Actel FPGAs were a standard way of implementing digital logic.

All relay control signals go through optoisolators to minimize the coupling of noise into the analog circuitry which is on a different CCA.

Also shown in Figure 5 are the voltage regulator and timer. The Controller CCA receives 8V DC input from an external Power Supply unit (**Photo 3**). The 8V is regulated locally down to 5V for the digital circuitry. The voltage for the displays and LEDs is variable and controlled by the room light level as sensed by a photocell on the front panel. The analog timer provides delays for output muting (“Enable”) and soft start (“SST”). It also drives the display decimal points.

Implementation

Photo 4 shows the PA2 preamplifier with the top cover removed. The upside-down Control Circuit Card Assembly (CCA), located top center, has the circuitry shown in Figure 5. Each channel has its own Amplifier CCA mounted on rails near the bottom of the chassis, which have the line amplifier, record/remote buffer, headphone amplifier, output switching relays, and local power regulation. An external $\pm 18V$ DC power supply is post-regulated to $\pm 15V$ DC in the main chassis. The assembly in the lower right corner is a plug-in module with two stereo phono preamplifier CCAs.

Photo 5 shows Amplifier CCA details on the left. The input buffer op-amps are on the left, followed by the attenuator relays. The output op-amps are on the right. Each relay is driven by a power FET. The FETs, in turn, receive optically isolated command signals from the Control CCA.

Performance

The line stage was implemented with $\pm 1\%$ tolerance metal film resistors. The worst-case measured gain error over the full 63dB range was 0.12dB including test equipment errors.

I included an adjustment to optimize CMRR at any one gain setting. Adjusted at maximum gain, measured CMRR at 1kHz is a very high 97dB. This degrades to 81dB at 20kHz and maximum gain. At other gain settings, the CMRR is not optimized by the adjustment, but is still high. The worst-case measured CMRR at 1kHz with gain settings 6dB, 12dB, 18dB, and 24dB below maximum gain is 67dB. Given the $\pm 1\%$ resistor tolerance and the gain of 2 in the input stage, the theoretical worst-case CMRR is approximately 40dB, but this was not observed. Even 40dB is more than adequate.

The noise and distortion performance of the line stage is dominated by the op-amps. With OPA627s and AD797s, the performance is excellent. With gain before and after the low impedance attenuators, the

About the Author

Bruce Gillingham retired in 2018 after a 45-year career with a major US defense contractor. With an education in electrical engineering, he began his career as an analog and RF designer and progressed to systems engineering, providing technical leadership for numerous development programs. Bruce has been designing, building, and listening to his own preamplifiers and power amplifiers since high school. Early efforts were with tubes; most have been solid state. His other hobbies include listening to live and recorded music (mostly classical), photography, and wood/metal working. Bruce is married and still lives near to where he worked on Central Florida’s East Coast.



output noise level is highest at full gain. Initially, it decreases with gain and eventually plateaus. Terminating the positive and negative inputs with 300Ω to ground and setting the gain to maximum (+12dB), the single-ended output noise measured in the 20Hz to 20kHz band is -103dBV. This is 109dB below a 2V RMS output. Reducing the gain to 0dB, the measurement improves to -108dBV.

Differential noise without an external cable attenuator is higher, but so is signal gain (+6dB). Noise contributed by the input gain stage is correlated on the two outputs so it will increase by 6dB. Noise contributed by the output stage amplifiers is uncorrelated and will increase by only 3dB. Worst-case, the differential signal-to-noise ratio (SNR) is no worse than when measured single-ended. Measured differentially at maximum gain (+18dB), the noise in the 20Hz to 20kHz band is -98dBV. Reducing the gain by 12dB, the measurement improves to -105dBV. Keep in mind that these are noise voltages and not SNRs. They do not show the +6dB signal gain advantage on the differential outputs.

Distortion is also very low and challenges my QuantAsylum QA403 Audio Analyzer. Total harmonic distortion (THD) at 1kHz with gain set to 0dB (single-ended) and 4V RMS across the differential outputs into a 200kΩ load is 0.00011%. The largest spur at 3kHz is 120dB below the fundamental (Figure 6). Distortion at 20kHz

is a still excellent 0.00023% with all spurs down more than 115dB (Figure 7). Figure 8 shows the THD vs. frequency at two different gain settings. Also plotted is the analyzer’s residual distortion (green trace). Note how close all traces are.

Figure 9 shows intermodulation distortion (IMD) performance with equal amplitude 19kHz and 20kHz tones. The 1kHz difference tone is more than 133dB below the RMS value of a sine wave having the same peak value as the sum of the two tones. This corresponds to 0.000022% distortion. The third-order products at 18kHz and 21kHz are about 2dB higher.

Summary

The enhanced instrumentation amplifier makes an excellent line stage. It provides high performance and interface flexibility at inputs and outputs. I enjoy having repeatable, accurate gain settings. This allows me to immediately return to my favorite settings, and you can easily calculate the remaining headroom before clipping when using a source like a CD player. The PA2 has served me well for 20 years and I have no complaints other than it perhaps has more gain than what I need. With its extremely low distortion and noise, I have not felt compelled to make any changes. If all my loads were balanced, I might reduce the gain by 6dB, change the 604Ω series resistors on the outputs to 300Ω, and forgo the attenuators in the cables. 

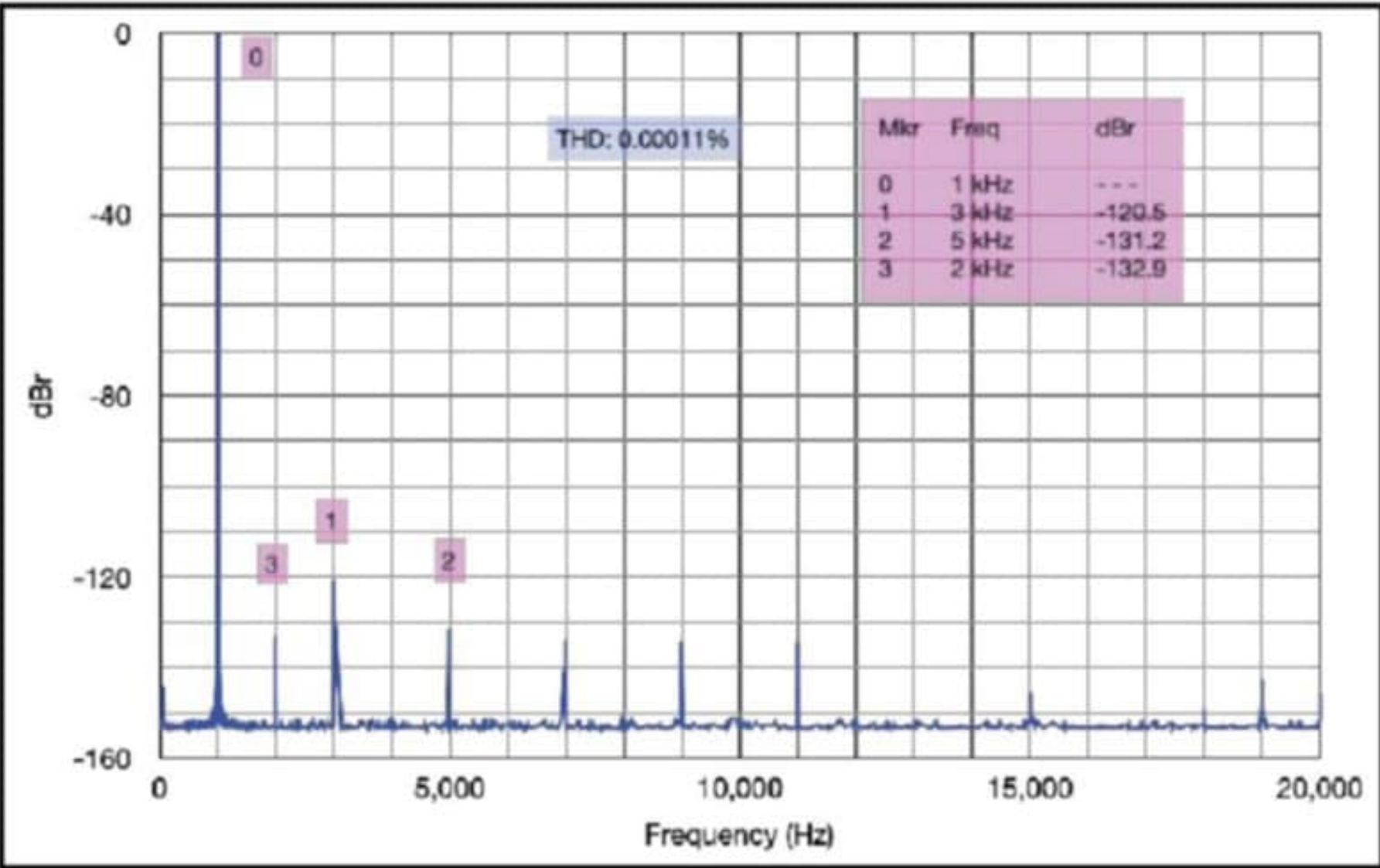


Figure 6: This graph shows the line stage total harmonic distortion—1kHz at 4V RMS differential output into 200kΩ.

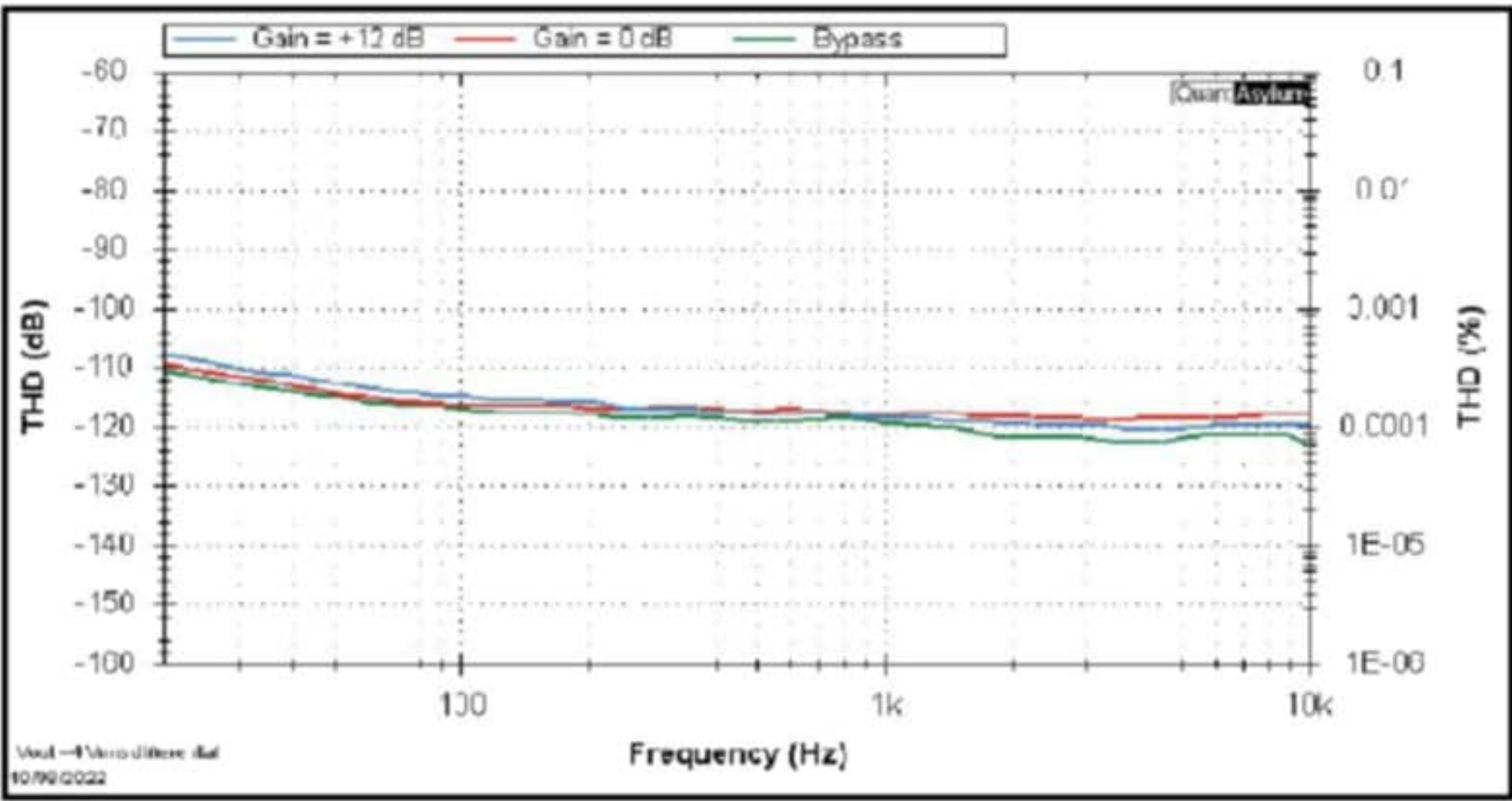


Figure 8: In this graph, the line stage THD vs. frequency is measured.

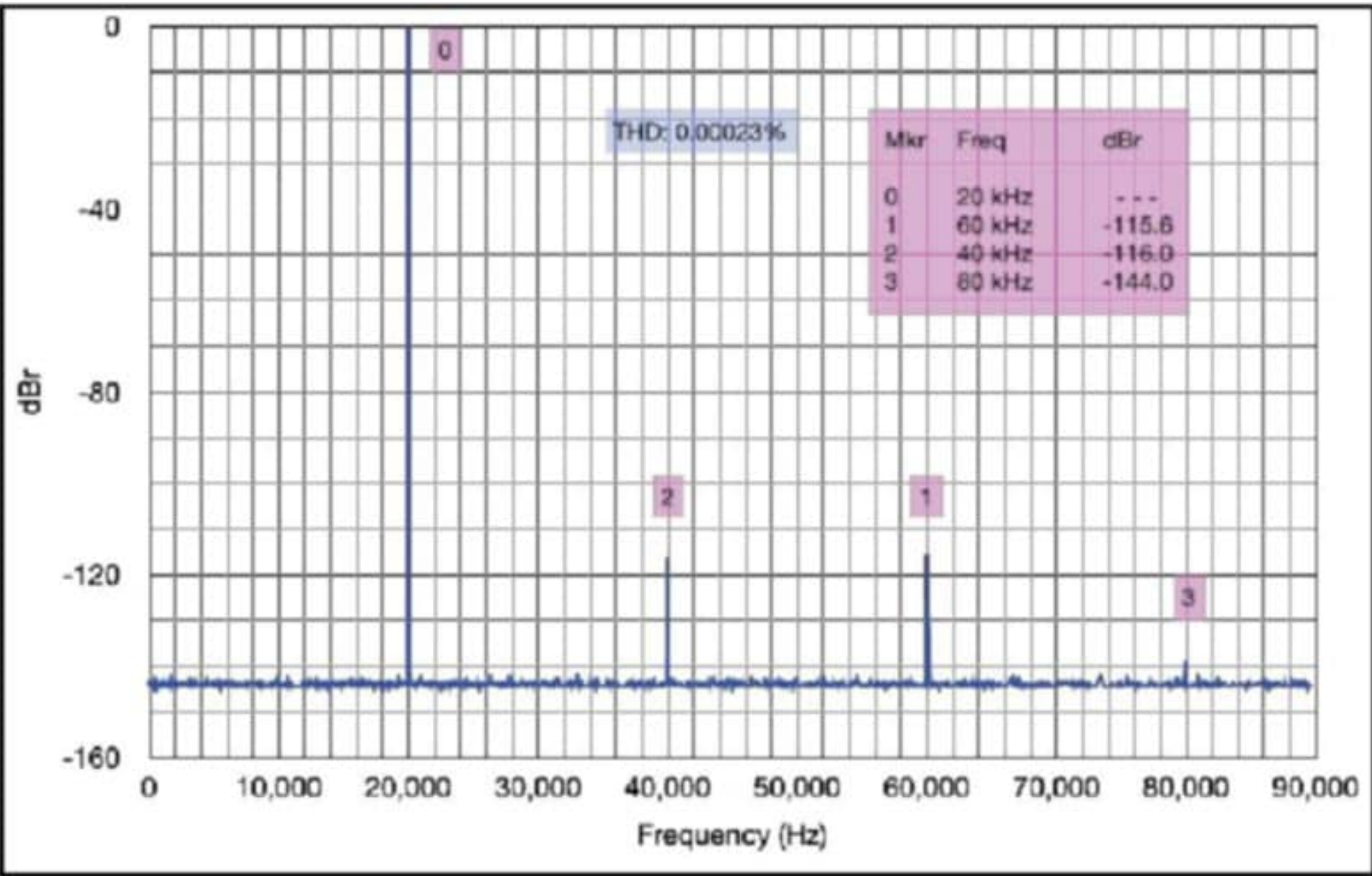


Figure 7: This graph shows the line stage THD—20kHz at 4V RMS differential output into 200kΩ.

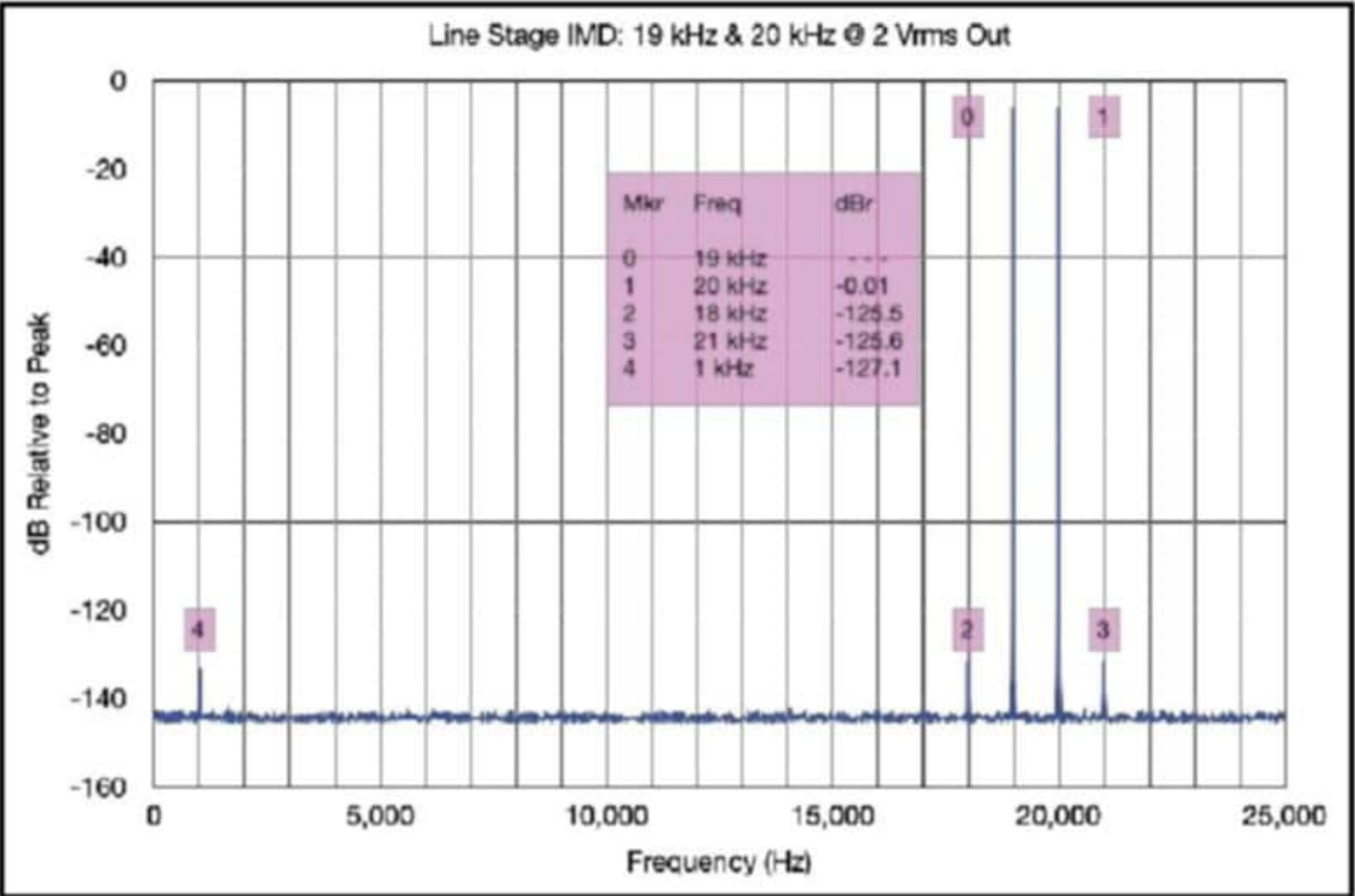


Figure 9: In this graph, the line stage intermodulation distortion (IMD) is shown at 19kHz and 20kHz at 2V RMS out.

Power Transformer Parameters, Selection, and Testing

Part 10 — More Transformer Tests

In this article, we continue to test different types of transformers for leakage flux and external magnetic shielding with both steel and aluminum, core loss sensitivity to the presence of these metals, and measurements of external magnetic leakage fields with distance, with and without shielding.

By
Chuck Hansen

Magnetic Field Evaluator Measurements for E-I Laminated Core Transformers

The first E-I core transformer I selected for this test was the Hammond 166G36, a 115V AC 18VA unit, which has bobbin-wound coils, a steel frame around three sides, and an open bottom with a fiber spacer that keeps it level with the mounting tabs (**Photo 28**). It weighs 400 gm (0.88 lbs.).

The second transformer was a RadioShack 273-1512B (**Photo 29**), a 120V AC 50.4VA with bobbin-wound coils, a steel frame with end bells over the windings, and a flat steel piece that covers the bottom. It weighs 828 gm (1.83 lbs.).



Photo 28: The first E-I core transformer I selected for this test is the Hammond 166G36.

The third transformer was a Triad F-264U (**Photo 30**). It is a 115V AC 120VA 50Hz/60Hz unit with layer-wound coils, an open frame (laminations exposed all around), and four steel strips rivetted to the laminations that act as mounting feet. It weighs 2.59 kg (5.71 lbs.).

This gave me three VA ratings and three different construction methods to evaluate for E-I transformer flux leakage.

The transformer primary voltages were measured at the terminal strip where the primary leads were connected to the AC power. The secondary voltages were measured at the terminal strip where the secondary leads were connected to the load resistors. The currents were measured with Arduino-type 5A AC current transformers.

I applied the rated full load for a 10-minute heat run to ensure the winding temperatures were stabilized, since winding resistance increases with temperature and reduces the efficiency.

Figure 34 shows the relative leakage flux density vs. distance from the AC Magnetic Field Evaluator (MFE) probe for the Hammond 166G36 EI transformer mounted on the 16-gauge (0.05") aluminum mounting plate. Note that the vertical flux density scale is logarithmic.

In this graph, no load (NL) is a dashed line; rated or full load (FL) is a solid line. The peak leakage flux density at NL is 1,582mG at the frame top (dashed green line). The peak leakage flux density at FL is 2,719mG, at the windings (solid red line).

The change in the secondary voltage from no-load to full-load with the primary voltage held at 115V AC was from 40.76V AC to 35.09V AC. The regulation factor (ϵ) was 0.1616, or 16.16%.

There was a 4.5 times leakage flux density increase between NL and FL for measurements at the windings (red lines), which was to be expected. The winding current generates the magnetic

field in the transformer core. However, I was surprised at the big difference between NL and FL measurements at the side of the frame (blue lines), with FL relative leakage flux density 8.9 times higher than NL with the probe against the center of the side of the transformer frame. The 18-gauge (0.048") steel frame of the transformer surrounds the top and both sides, and the steel sides are bent out 90 degrees at the bottom, forming the mounting brackets. The laminations are exposed at the open bottom.

There was a 13% increase in relative leakage flux density from NL to FL at with the MFE probe at the top of the transformer frame (green lines). There was a 17% decrease in relative leakage flux density from NL to FL with the probe and the exposed laminations at the bottom of the transformer (violet lines), another surprise.

Figure 35 shows the relative leakage flux density vs. distance from the MFE probe for the Hammond 166G36 EI transformer with a 20-gauge (0.037") carbon steel mounting plate.

Again, NL is a dashed line; rated or FL is a solid line. The peak leakage flux density at NL was 1,523mG at the frame top (dashed green line). The peak leakage flux density at FL was 2,929mG, at the windings (solid red line).

The change in the center-tapped secondary voltage from no-load to full-load with the primary voltage held at 115V AC was again from 40.76V AC to 35.09V AC. The regulation factor (ϵ) was 0.1616, or 16.16%. This shows that the regulation factor was essentially the same whether an aluminum or carbon steel mounting plate was used, even though the leakage flux densities are different.

Note that we are not talking about huge amounts of flux density here. The highest relative flux leakage measurement with either mounting plate was less than 3 gauss. CRGO steel FL core flux density was 15kG to 18kG. However, that is not to say leakage flux was insignificant with respect to susceptible low-level signal circuits in close proximity.

After these interesting results, I began to plan more experiments with the 166G36 transformer.

Wiring Design and EMI

There are two forms of EMI, Conducted and Radiated [7]. Both types can manifest themselves as either electric field (E field) or magnetic field (H field) emissions. There are differences between what is required for commercial equipment, commercial off-the-shelf (COTS) equipment and military equipment.

Tests with Parallel and Twisted-Pair Wires on Aluminum Mounting Plate

I didn't want to make any changes to my existing flux measurement fixture, so I cut new pieces of 16-gauge aluminum and 20-gauge carbon steel to 30cm x 14cm. Then I taped two 12-position barrier terminal blocks, one at each end of the metal plates, separated from each other by about 25cm. I made separate sets of primary and secondary test wires 30cm long. The final overall length was 25cm after making the twisted pair on the left side. I used standard PVC-covered tinned AWG 22 solid copper wire, which holds its shape better than Teflon stranded wire for this purpose. The two sets of wires were separated by about 9cm (**Figure 36**).

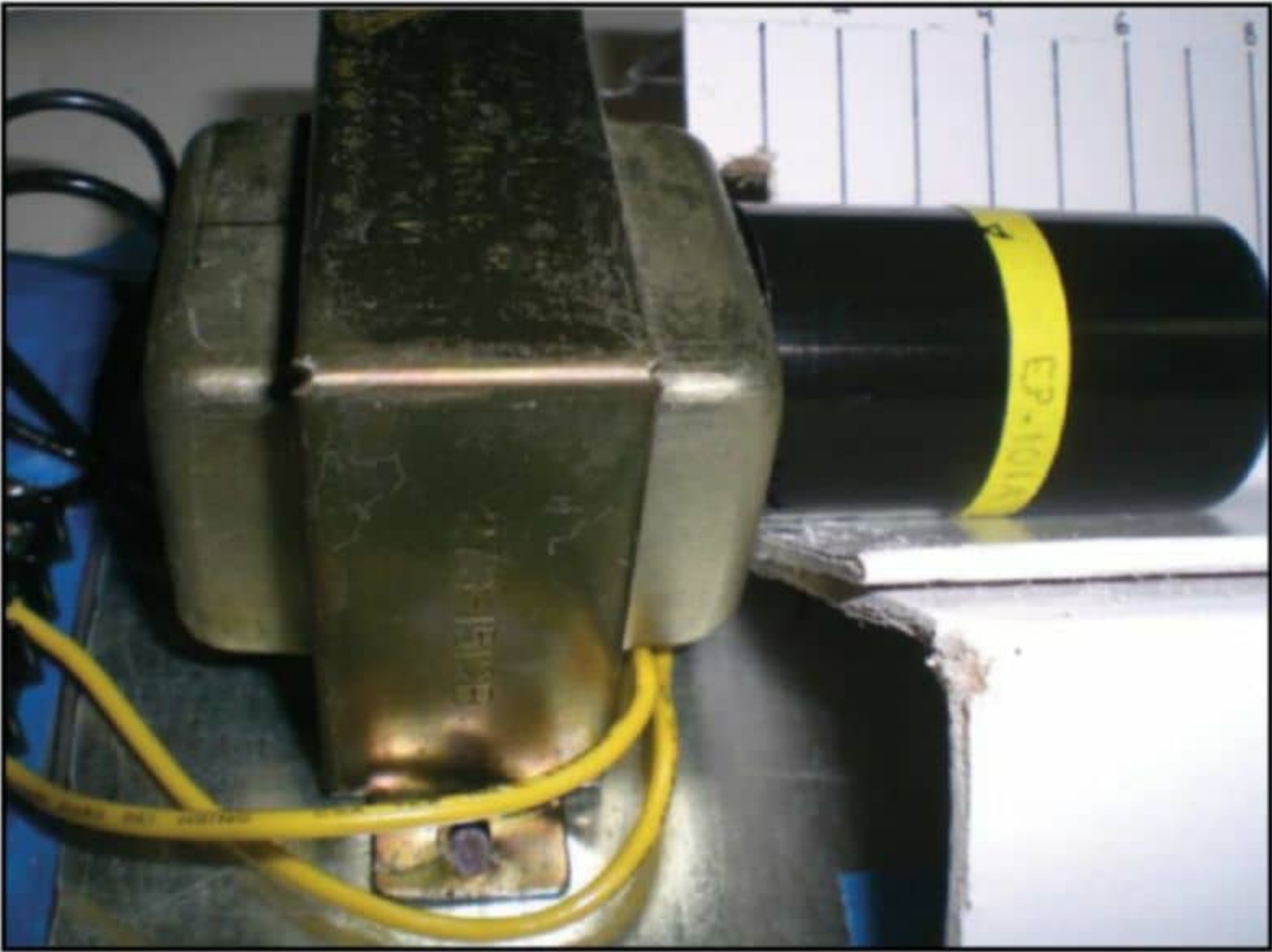


Photo 29: The second transformer I selected was a RadioShack 273-1512B.



Photo 30: The third transformer that I tested was a Triad F-264U.

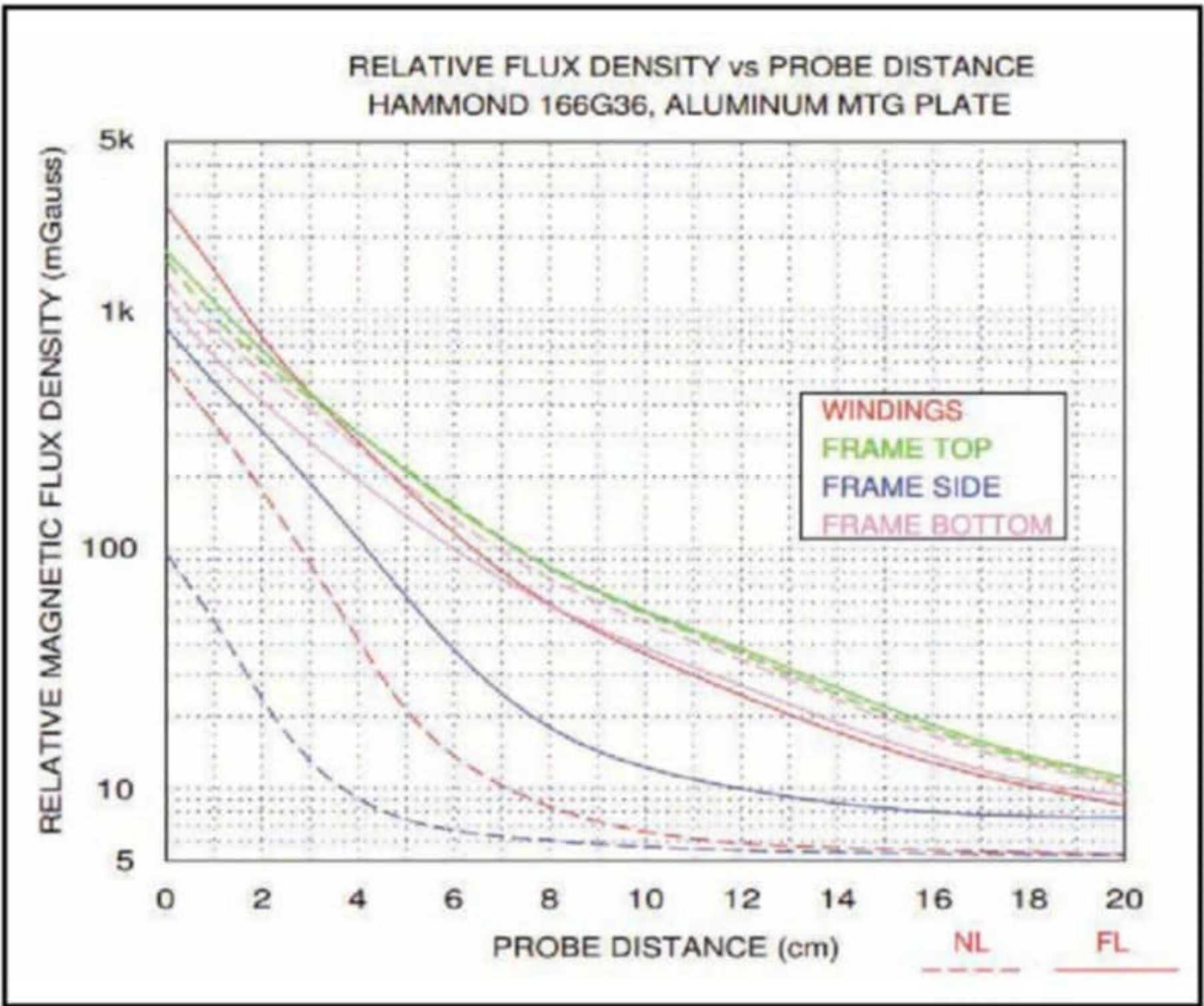


Figure 34: Leakage flux density vs. distance from MFE probe for Hammond 166G36 EI transformer with the aluminum mounting plate.

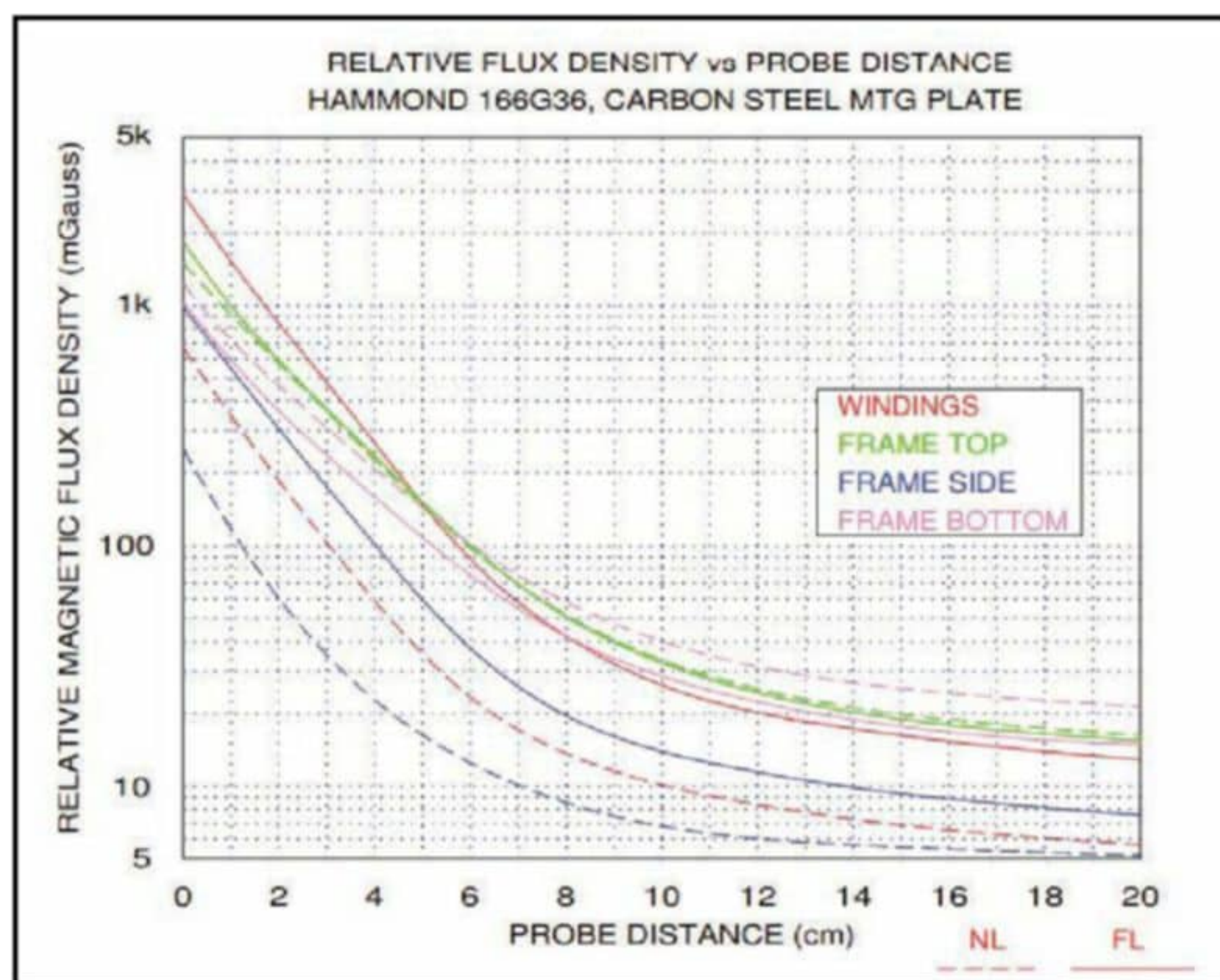


Figure 35: Leakage flux density vs. distance from MFE probe for Hammond 166G36 EI transformer with a carbon steel mounting plate.

The 12.5cm right half of each parallel wire pair had the two wire insulations touching each other. Next, I twisted the left half together about two turns-per-inch twisted-pair and cut these wires to their final 12.5 cm length. I stripped the ends and taped all the wires down flat against the aluminum plate with blue masking tape.

The wires in each parallel and twisted-pair group were designed to minimize the loop area of the wire runs, which is the product of the length of the wires carrying the source and return current and the physical distance between the conductors in each pair of wires. Twisting the wires had the added benefit of alternating the polarity of the adjacent loops, which cancels out the radiated field, especially when there are an even number of loops (Figure 36).

I connected the Hammond 166G36 transformer to the right-side terminal block and the power supply and load bank to the left-side terminal block. I put masking tape over the two terminal block areas that are connected to the 115V AC for safety.

I measured 332mG at NL and 414mG at FL with the MFE probe against the parallel primary wire pairs (see the upper section of **Table 4**). Then I measured 12.6mG at NL and 9.3mG at FL for the twisted-pair primary wires. I repeated the measurements again to make sure the leakage flux density readings really

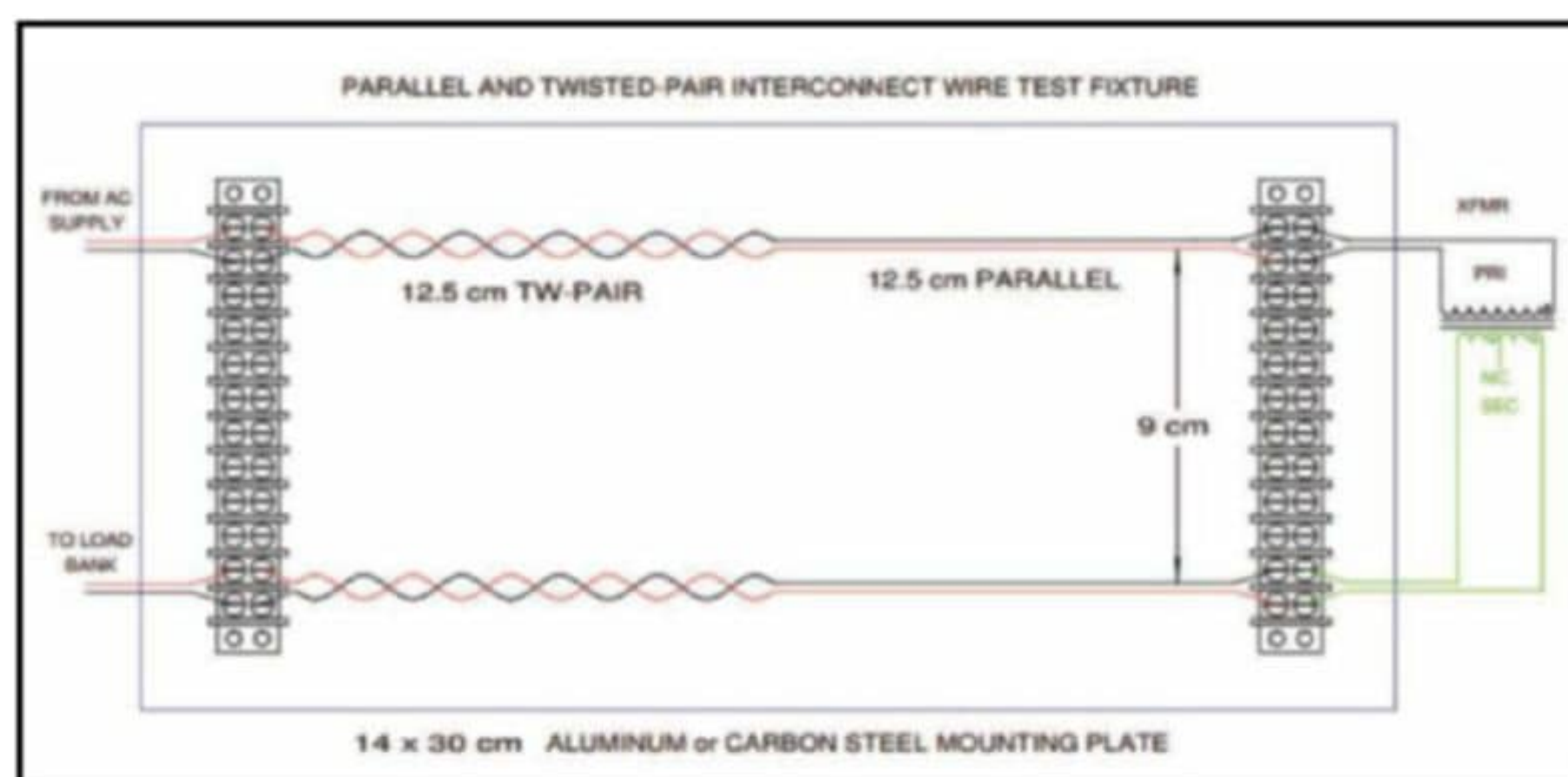


Figure 36: Parallel and twisted-pair interconnect wire test fixture.

did show that NL was higher than the FL for the twisted pair. I didn't expect that.

I also used 1cm and 2cm non-conductive spacers to measure the decreased in leakage flux density as the MFE probe moved vertically away from the parallel wires, also shown in Table 4.

This was comparable to the leakage flux density drop with distance measured back in Figure 34. However, there was no measurable change in leakage flux density with distance when I used the spacers over the twisted-pair section, hence it wasn't necessary to provide the data in the table.

Moving over to the secondary wires, I measured 146mG at NL and 218mG at FL with the MFE probe against the parallel primary wire pairs. Then I measured 8.3mG at NL and 7.6mG at FL for the twisted-pair primary wires. Again, the leakage flux density readings showed the NL was higher than the FL for the twisted-pair section.

I used the 1cm and 2cm non-magnetic spacers to measure the decreased in leakage flux density as the MFE probe moved vertically away from the parallel secondary wires, as shown in Table 4. This was also comparable to the leakage flux density drop with distance measured in Figure 34. As with the primary wires, there was no measurable change in leakage flux density with distance when I used the spacers over the twisted-pair section.

Tests with Parallel and Twisted-Pair Wires on Carbon Steel Mounting Plate

I removed the aluminum plate from under the extended wires and replaced it with the 20-gauge (0.037") carbon steel plate and taped the terminal blocks and wires back in the same locations.

I measured 210mG at NL and 306mG at FL with the MFE probe against the parallel primary wire pairs (lower section of Table 4). Then I measured 10.9mG at NL and 8.4mG at FL for the twisted-pair primary wires. I repeated the measurements again to make sure the leakage flux density readings really did show the NL was higher than the FL for the twisted pair. This seems to be true regardless of the type of metal mounting plate used.

I also used 1cm and 2cm non-magnetic spacers to measure the decrease in leakage flux density as the MFE probe moved vertically away from the parallel wires, also shown in Table 4. This was comparable to the leakage flux density drop with distance measured in Figure 35. Again, there was no measurable change in leakage flux density with distance when I used the spacers over the twisted-pair section.

Moving over to the secondary wires, I measured 200mG at NL and 242mG at FL with the MFE probe against the parallel primary wire pairs. Then I measured 10.1mG at NL and 9.3mG at FL for the twisted-pair primary wires. Again, the leakage flux density readings showed the NL was higher than the FL for the twisted pair.

I used the 1cm and 2cm non-magnetic spacers to measure the decrease in leakage flux density as the MFE probe moved vertically away from the parallel secondary wires, as shown in the bottom of Table 4. This was also comparable to the leakage flux density drop with distance measured in Figure 35. As with the primary wires, there was no measurable change in leakage flux density with distance when I used the spacers over the twisted-pair section.

MENLO SCIENTIFIC

Engineering
Success
since 1983

Engineering Consulting for the Audio Industry

We Know

- Product Development

We Provide

- Design and Technical Support

We Deliver

- Liaison, Sourcing, and Guidance

We've worked with the world's largest brands, supported large companies outsourcing production, advised manufacturing facilities on streamlining their process, and helped companies large and small to develop new products, and launch revolutionary technologies. We've helped startups find investors, and introduced entirely new concepts. Many companies learn their lesson by failing. Others have hired us first.

MENLO
SCIENTIFIC

Richmond, California, USA

- tel 510.758.9014
- mike@menloscientific.com
- www.menloscientific.com



These tests again verified that twisted-pair wiring does help cancel stray electromagnetic fields.

Magnetic Shielding Effect Tests with Aluminum, Copper and Carbon Steel

Continuing with my extra 166G36 transformer tests, I did a variation on the baseline AC magnetic field tests. See Photo 28 again for the baseline setup, except I used the 20-gauge (0.037") carbon steel mounting plate.

I inserted a piece of 16-gauge (0.05", 0.127 cm) aluminum between the MFE probe and the transformer windings and taped the MFE probe in that location. The NL flux dropped from 657mG to 557mG. Next, I removed the aluminum and recorded the NL flux. Then I inserted a piece of unetched FR4 double-sided 1-ounce (1.4 mils) copper PCB into the same 0.05" air gap space, and finally I inserted a piece of 20-gauge carbon steel into the same 0.05" air gap space and recorded the NL flux in those two cases (Table 5). I repeated the winding MFE probe measurements at FL.

The aluminum and the copper (Cu) PCB had negligible effects on the measured flux. The carbon steel caused a drop in flux to about 23% of the no gap flux value.

Impedance Measurements of Parallel and Twisted-Pair Wires

Using the MFE probe I determined that the flux from the transformer under test decrease with distance until the millivolt reading fell to a uniform low value 12.5cm from the terminal block in the extended wires (Figure 36).

The next set of tests for the Hammond 166G35 E-I transformer test fixture was to measure the impedance of the 12.5cm lengths

Aluminum Mounting Plate, 16-gauge (0.05")		
Load	NL	FL
Primary Parallel Wires	332mG	414mG
Primary Parallel, 1cm	141mG	179mG
Primary Parallel, 2cm	118mG	153mG
Primary Twisted Pair	12.6mG	9.3mG
Secondary Parallel Wires	146mG	218mG
Secondary Twisted Pair	8.3mG	7.6mG
Carbon Steel Mounting Plate, 20-gauge (0.037")		
Load	NL	FL
Primary Parallel Wires	210mG	306mG
Primary Parallel, 1cm	196mG	287mG
Primary Parallel, 2cm	163mG	238mG
Primary Twisted Pair	10.9mG	8.4mG
Secondary Parallel Wires	200mG	242mG
Secondary Twisted Pair	10.1mG	9.3mG

Table 4: Leakage flux density measurements for Hammond 166G36 transformer parallel and twisted-pair primary and secondary wires on aluminum and carbon steel mounting plates.

of parallel and twisted-pair wires that were in contact with the two different mounting plates. I cut the long wires into separate sections for parallel and twisted pair so I could test them individually without any interaction.

I created twisted-pair wires from longer lengths and trimmed it down to 12.5 cm after twisting the two wires together. This increased the finished resistance of the twisted pair 125% over the same finished length as the parallel wires.

I used my Keysight U1733C RLCZ tester for measurements using its 120Hz test frequency. I measured the resistance with the two wires in series, and capacitance with the two wires not connected to each other.

The final impedance measurements are shown in Table 6. The thing I found interesting was that the wire-to-mounting plate capacitance was lower on the aluminum plate than in free air. The carbon steel plate showed the highest capacitance. The inductance of the wires is down in the nanohenry (nH) range, where I can't really get useful readings with my Keysight RLCZ tester.

Waveforms of Secondary Current and Magnetic Flux

The final tests for the Hammond 166G35 E-I transformer were to look at the waveforms of the full load secondary current and magnetic leakage flux density. In the previous measurements the maximum leakage flux density was 2,929mG at the windings with the transformer mounted on the carbon steel sheet metal.

I used my Pico ADC-216 digital oscilloscope, which has 16-bit accuracy. I measured the secondary current using the current transformer inside my test fixture (blue trace in Figure 37).

MFE Probe at Windings			
Test Configuration	Gap	No Load	Full Load
No gap	0	657mG	2,929mG
16-gauge Aluminum	0.05"	557mG	2,614mG
Remove Aluminum	0.05"	557mG	2,614mG
Double-Sided Cu Blank PCB	0.05"	565mG	2,636mG
20-gauge Carbon Steel	0.05"	150mG	661mG

Table 5: Magnetic shielding effect tests with aluminum, copper PCB and carbon steel

12.5cm Parallel Wires	Air	Aluminum	C-Fe
R-Resistance	59mΩ	59mΩ	59mΩ
C-Capcitanace	5.36pF	3.86pF	5.46pF
L-Inductance	0.1μH	0.1μH	0.2μH
12cm Twisted Pair	Air	Aluminum	C-Fe
R-Resistance	74mΩ	74mΩ	74mΩ
C-Capcitanace	6.05pF	5.66pF	6.53pF
L-Inductance	0.2μH	0.2μH	0.2μH

Table 6: Wiring impedances for 12.5 cm parallel and twisted-pair wires in free-air and mounted on Aluminum and then Carbon Steel sheet metal.



I measured the leakage flux density with the MFE probe connected to my $\times 10$ gain amplifier (red trace).

The final test was the dBv vs. frequency spectrum of the Hammond 166G35 E-I transformer full-load secondary current with the carbon steel mounting plate. The bandwidth is 651Hz (**Figure 38**).

Harmonics are voltages or currents that occur at some multiple of the fundamental AC line frequency (60Hz in North America or 50Hz in Europe) caused by nonlinear loads on the power system, in this particular case, odd harmonics. Since harmonics are multiples of the fundamental, they are usually identified as $2f$, $3f$, $4f$, $5f$, etc.

The five peaks are the 60Hz fundamental frequency (0 dBv), as well as the first four odd harmonics at 180Hz (-29.2 dBv), 300Hz (-32.1 dBv), 420Hz (-38.9 dBv), and 540Hz (-42.1 dBv). The 660Hz harmonic (-44.9 dBv) was just off-screen to the right.

There was also a lot of low amplitude grunge from the AC power line just below -60dB. I expanded the bandwidth of the spectrum to its maximum and the grunge floor extends many tens of kilohertz.

There are no even harmonics shown in Figure 38 since there was half-wave symmetry and no DC content in the 120V AC 60Hz power we receive from the power companies.

Nonlinear loads can include:

- Magnetic core transformers and rotating machines (motors and generators) especially as they approach magnetic saturation.
- Switch-mode power supplies—AC to AC or DC to AC converters that provide AC power from solar panels or wind generators, or for high-voltage DC power transmission lines.
- System L-C resonance that occurs at the same frequency as the power line.

Institute of Electrical and Electronics Engineers (IEEE) Standard 141-1993, "IEEE Recommended Practice for Electric Power Distribution for Industrial Plants," includes lists of the types of nonlinear devices that might cause power line harmonics.

The calculated root-sum-square (RMS) total harmonic distortion (THD) in Figure 38 is 4.48%, while my HP-339A distortion test set measured 5.1% THD+N, where N is the non-harmonic noise content. Since the secondary full load current THD includes any distortion caused by the transformer under test, the AC power line THD may well be lower (I did not explore this since it is highly variable depending on the total nonlinear loading on the system, as well as in the B-K ac power supply isolation transformer and Variac as they approach saturation).

The THD shall not exceed 5.0% for 60Hz commercial AC voltage at the point of common contact (PCC), as specified in IEEE Standard 519-1992. The THD limit for individual odd harmonics is 3.0%.

The 240V AC 50Hz power in Europe is considered to be somewhat better than our 120-0-120V AC 60Hz power in that it is inherently balanced. If there is a significant difference in the load current on each 60Hz 120V AC bus in your home electric panel, there could be a voltage imbalance from side to side.

The three-phase 400Hz AC generators we make for aerospace applications are designed not to produce third harmonics or their triplen harmonics (9th, 15th, etc.), and minimal levels of the 5th, 7th, 11th, and 13th harmonics. When solid-state DC-to-AC 50Hz, 60Hz, or 400Hz converters are used, there is a MIL-Spec steady-state limit of $\pm 0.1V$ DC for DC content.

The peak level of a pure AC sine wave is the RMS value times $\sqrt{2}$. The third harmonic will reduce the peak level of the AC waveform vs. a pure sine wave. The 7th and 11th harmonics will do the same. The 5th, 9th and 13th harmonics will increase the peak level of the AC waveform vs. a pure sine wave.

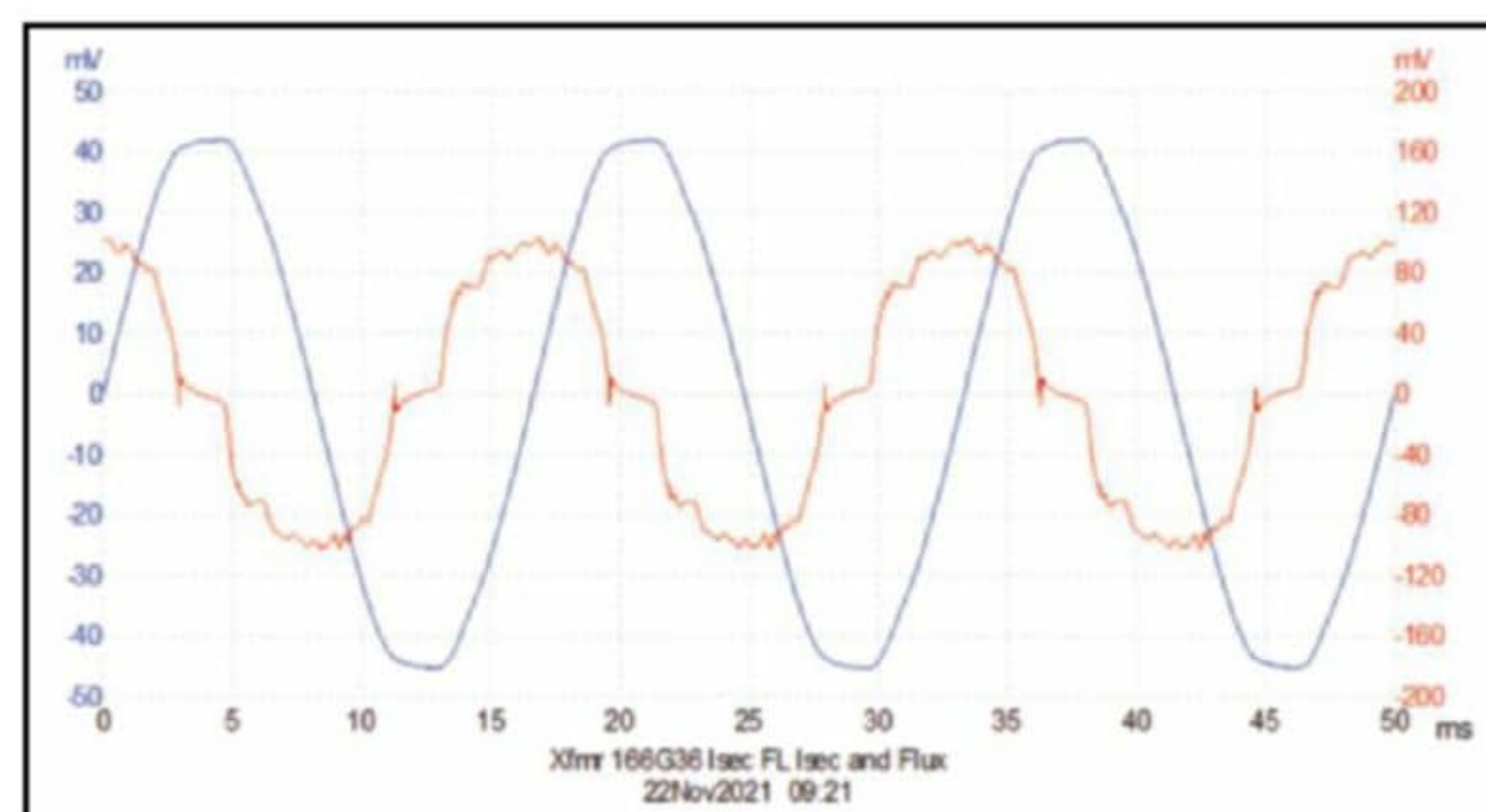


Figure 37: Waveforms of Hammond 166G35 E-I transformer full-load secondary current and magnetic leakage flux density with carbon steel mounting plate.

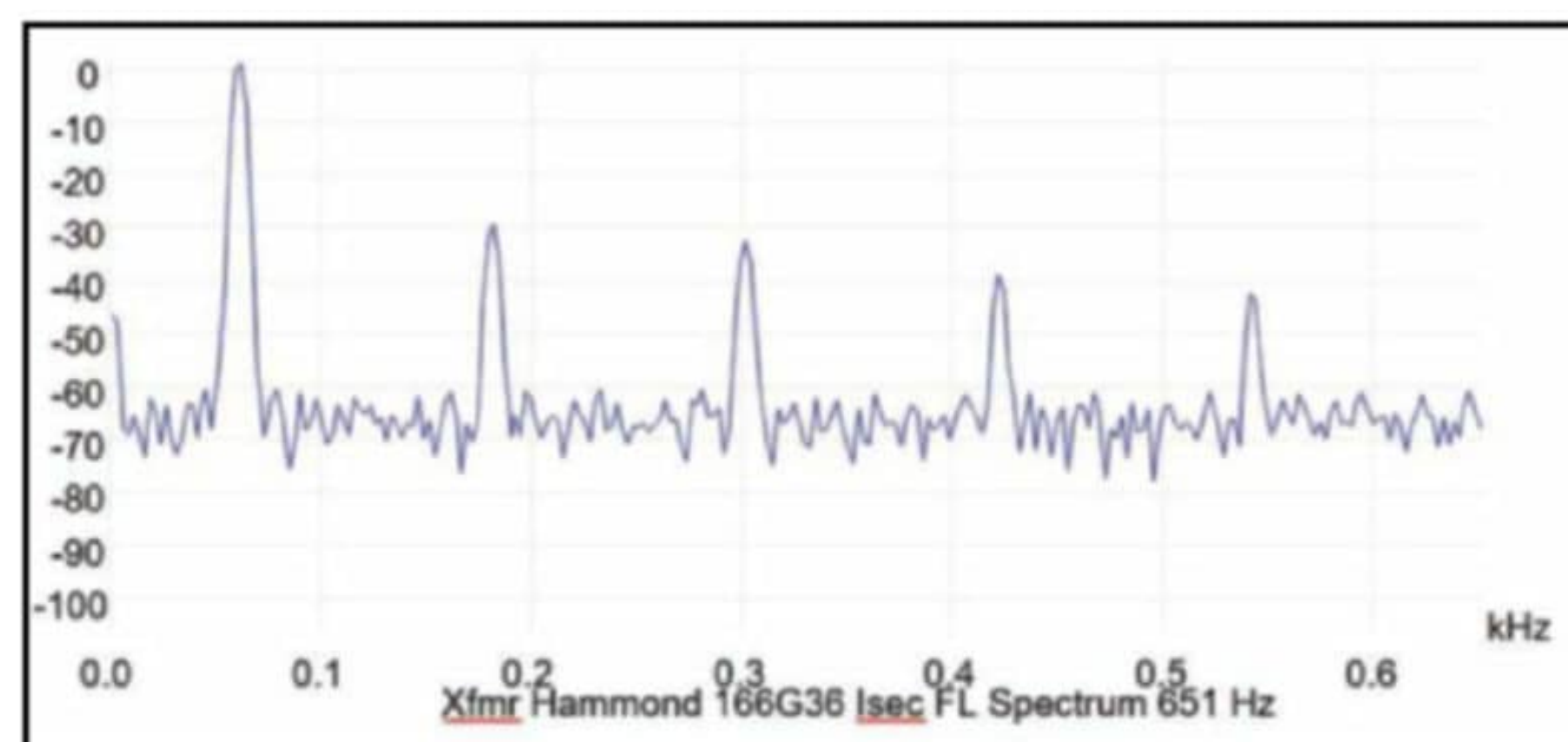


Figure 38: Spectrum of Hammond 166G35 E-I transformer full-load secondary current with carbon steel mounting plate.

About the Author

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





Figure 39: Waveforms for Hammond 166G36 due to addition of harmonics $3f$, $5f$, $7f$ and $11f$ to the 60Hz fundamental. Harmonics are three times the levels shown in Figure 38.

Figure 39 shows a SPICE transient analysis of the fundamental sine wave and first four odd harmonics over one full 60Hz AC cycle (there is no 9th triplen). The levels have been multiplied to three times the levels of the spectrum in Figure 38 to better show the harmonic effects on the fundamental. The waveform color codes are shown in the chart on the upper left. They are A – fundamental $1f$, 60Hz; B – third harmonic $3f$, 180Hz; C – 5th harmonic $5f$, 300Hz; D – 7th Harmonic $7f$, 420Hz; E – 11th harmonic $11f$, 660Hz. The letters correspond to the color bars on the upper left side of the Transient Analysis (Figure 39). The green trace F is the resultant of the sum of all the harmonic levels.

The effect on the fundamental can best be seen by the polarities of the harmonics at their combined amplitudes at 90° and 270° . Harmonics $5f$ (violet) adds to the fundamental peak, but $3f$ (dark blue), $7f$ (light blue) and $11f$ (yellow) subtract from the peak. Since $3f$ has the highest amplitude of all the harmonics, the net effect is that the peaks are depressed, while the two “shoulders” approaching zero crossing are wider than that of the fundamental. The fundamental and all the harmonics cross zero exactly at 0° and 180° .

Next Month

Next month, we will continue with tests on E-I lamination transformers and look at some R core tests of the same types as the E-I transformer tests.

References

[7] “Conducted Emission vs Radiated Emission—Difference between Conducted Emission and Radiated Emission,” RF Wireless World, www.rfwireless-world.com/Terminology/conducted-emission-vs-radiated-emission.html



Loudspeaker Design Cookbook 8th Edition

For the First Time Ever...

Available in
Full Color
and a
Hardcover Collector's Edition



Learn to Design Professional Sounding Loudspeakers for:

- **Residential**
- **Automotive**
- **Home Theater**

Vance Dickason's top-selling *Loudspeaker Design Cookbook* is an essential must-read for serious speaker designers, audio engineers, and anyone looking to master speaker-building technology. With more than 140,000 copies in print, author Vance Dickason's *Loudspeaker Design Cookbook* is not only regarded as the loudspeaker engineering “bible,” but the #1 resource on the construction of high-quality loudspeaker systems for two-channel, home theater, pro sound PA speakers, and studio monitors, as well as car audio applications.

Get your copy at: CC-Webshop.com/LDC8

J SERIES

THE LEGENDARY J SERIES IS BACK!



TRIAD'S CLASSIC J SERIES TRANSFORMER DESIGN COMES OUT OF THE VAULT WITH ITS LEGENDARY PERFORMANCE PEDIGREE AND MORE. BORN IN THE 1950S, THE J SERIES TRANSFORMERS FEATURES HIGH-NICKEL LAMINATIONS, PAPER SECTION WINDINGS AND A SPECIALLY TOOLED MU-METAL SHIELDING ENCLOSURE. LONG RECOGNIZED BY ENGINEERS FOR ITS QUALITY SOUND, TODAY'S J SERIES IS IDEAL FOR NEW OR VINTAGE AUDIO GEAR, SOUND PROCESSING AND ANALOG STUDIO RECORDING EQUIPMENT.

SHIELDED AUDIO IMPEDANCE MATCHING
DESIGN FOR A WARM SIGNAL.

SINGLE-HOLE MOUNTING ALLOWS ROTATION
FOR MAXIMUM HUM REDUCTION.

VINTAGE CIRCUITRY, PREMIUM
MATERIALS, HAND-CRAFTED QUALITY.

VISIT ONLINECOMPONENTS.COM TO
ORDER OR LEARN MORE.

MADE IN THE U.S.A.



onlinecomponents.com

THE ONLINE DISTRIBUTOR OF ELECTRONIC COMPONENTS

Dayton Audio BST-300EX **Extreme High-Power** with **6X** the Watts!

Dayton Audio BST-300EX Extreme High Power Pro Tactile Bass Shaker 300 Watts

The BST-300EX is a powerful bass shaker designed for serious audio enthusiasts. With an impressive 300W of power, it delivers extreme bass that will immerse you in your favorite movies, games, and music. The inaudible operation ensures a seamless experience, while the easy mounting makes installation a breeze.

#295-243



Dayton Audio BST-1 High Power Pro Tactile Bass Shaker 50 Watts

With 50 watts of power at 4 ohms, the Dayton Audio BST-1 Bass Shaker serves as an efficient exciter speaker, delivering an immersive tactile bass experience.

#295-244



Feel the
Tremors!



DAYTONAUDIO
daytonaudio.com

Distributed by:

PARTS EXPRESS
AUDIO COMPONENTS & SOLUTIONS
parts-express.com 1-800-338-0531
Custom OEM inquiries welcome



Scan for more
Information!
Dayton Audio
BST-300



With a Dayton Audio 5 year warranty, it's more than a guarantee, it's a promise. We stand behind our products and our warranty is a demonstration of our commitment to quality and reliability.