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Market Update

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By Michael Steffes

Audio Electronics
Powering Your Circuits: Mains Noise Filtering
By Ed Simon



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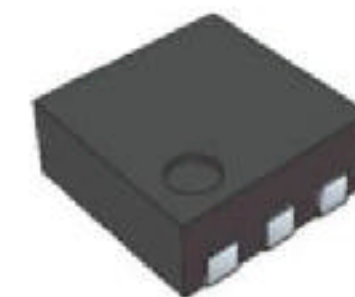
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Reshaping Automotive Audio

The pleasure of listening to music while driving is universally recognized and embedded in our collective imagination. Since the first car radios made their appearance in the 1930s, the connection between cars and music culture was cleverly explored in marketing. And as the automotive industry evolved, so did the applications for audio inside vehicles, including some of the very early experiments with voice commands and a never-ending number of attempts to transform car interiors with digital signal processing.

Today's automotive systems are highly integrated with audio, and the trend for that integration to expand is part of the reason why so many companies are betting heavily on the market's potential.

When we look at the cars of the (near) future, we know they will be predominantly electric, network-connected, and increasingly autonomous. And that also means that cars will have to be able to accommodate completely new requirements for drivers and passengers. For its part, the audio industry is working on it. In fact, there was never so much excitement about automotive applications in the audio industry.

But the automotive industry is a very tough environment. Not only because of the very specific challenges, requirements, and demanding specifications, but above all because the auto industry always had its own way of doing things. The industry works with much longer product cycles, typically likes to have control over what goes inside the car, and would prefer to destroy any opportunity for aftermarket products and sell their own optional, closed-systems instead.

And this clashes with the average consumer, in particular over the last two decades, with increasingly fast technology cycles. We've seen it happening with Bluetooth, GPS navigation, and telecommunications—all updated radically over the average lifecycle and ownership period of a car/model (6 to 10 years). The result was visible, with consumers bringing their own personal mobile devices to the car, and quickly turning factory-fitted systems obsolete. And the might of large technology companies such as Apple and Google, allowed them to sidetrack the automotive industry with operating system integrations—CarPlay and Android Auto—allowing their app and service ecosystems to invade the dashboards. Even Amazon is testing the market with its own Echo Auto hardware device and direct initiatives with automakers to build Alexa into their vehicles and create brand-specific Connected Car Alexa Skills.

All automakers and large technology companies know that cars will need to be 100% connected to be able to support advanced driver-assistance systems (ADAS—the strongest case for 5G networks) and to eventually support autonomous vehicles. And while cars will need the connectivity to move, passengers will need it to do other things, since they won't need to have eyes on the road.

In the middle of this intense technology transition, audio systems will be required to support vital features, from voice control to the car's own assistant interactions with passengers, noise, and acoustics control, or even enhancing the driving experience in silent-engine vehicles, or signaling their presence to the outside—all independently of music enjoyment.

Virtually everything the audio industry has developed for smart home applications, such as sophisticated smart speaker platforms, the most advanced acoustic compensation, and room correction features to the audio enhancement systems that we are still perfecting for hearables, all will be applied, transformed, and improved for automotive audio applications—while the business revenue generated will be significantly higher than any other audio market segment.

These are some starting points we thought to address in *audioXpress'* first Automotive Audio Market Update. In the end, we barely scratched the surface, and will return to the topic in the near future.

J. Martins
Editor-in-Chief

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COLUMNISTS

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

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

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

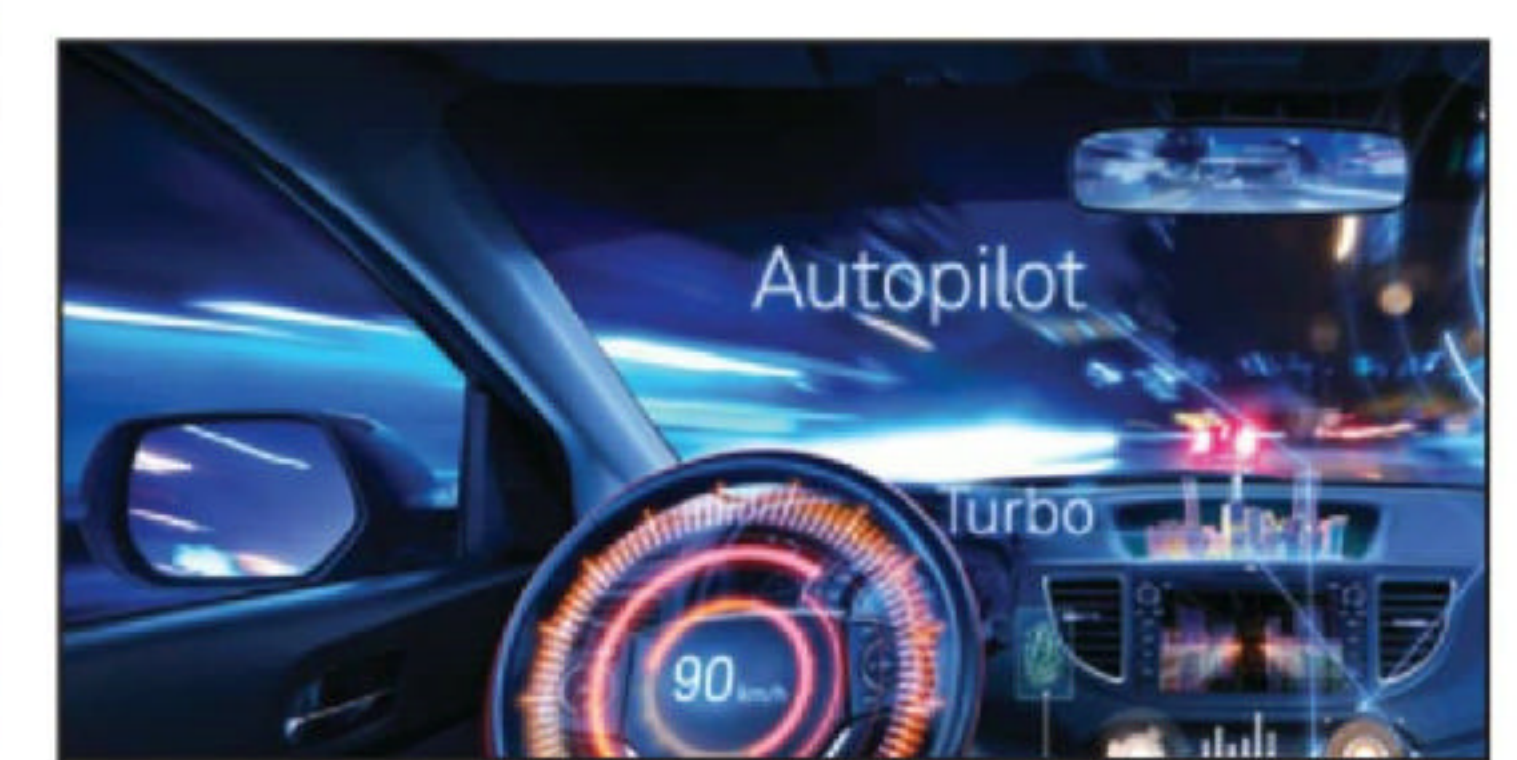
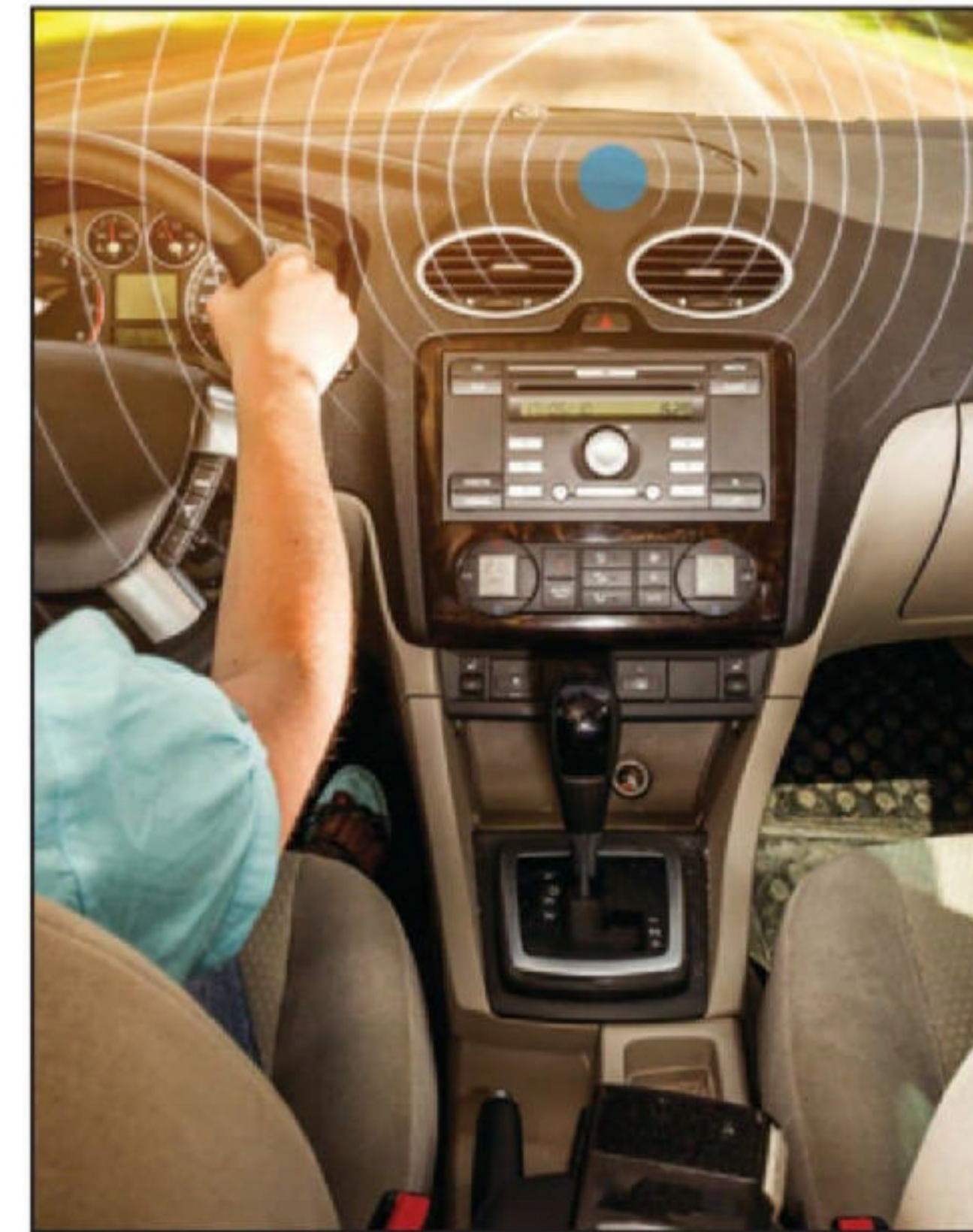
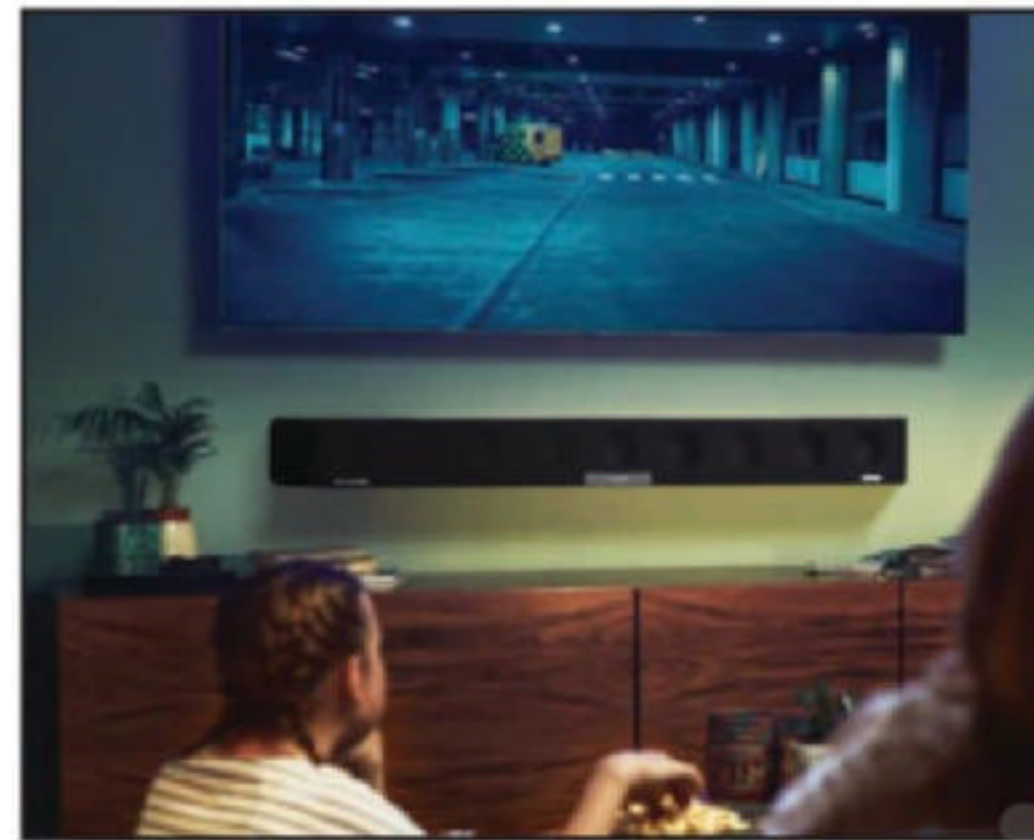
Michael Steffes has worked in high-speed signal path IC development, spanning IC design roles for the BurrBrown high-speed amplifier products, and product launch/application manager roles through Texas Instruments/Intersil. He continues to focus on high-performance ADC interface solutions, DSL/PLC line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

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Sennheiser AMBEO Immersive Soundbar

Raising the Bar



The Sennheiser AMBEO soundbar is a highly evolved system, able to deliver spatial audio that goes well beyond the typical home cinema experience.

By
Oliver A. Masciarotte
(United States)

The Sennheiser AMBEO soundbar is a sophisticated multi-speaker integrated home cinema sound system. It offers 13 high-end drivers, a frequency response down to 30 Hz, and even ships with a measurement microphone for room calibration. Now it is available for everyone to buy, so a review was in order.

Soundbars, a relatively new lifestyle category, serve many masters. On the one hand, they provide arguably the most important aspect of a home theater experience: the audio. On the other hand, they hide all that sound delivery in plain sight. At least, as unobtrusively as possible. As flat screen TVs have grown to gargantuan proportions and channel counts have increased into double digits, soundbars too have had to grow as well, to stay on par with ever greater expectations.

Why purchase a soundbar rather than the more common Home Theater in a Box (HTIB)? For one, clutter. Spousal approval often rests on a clean and unobtrusive install. Having speakers everywhere and wires to trip over, crush, or get mangled makes for a less than ideal installation.

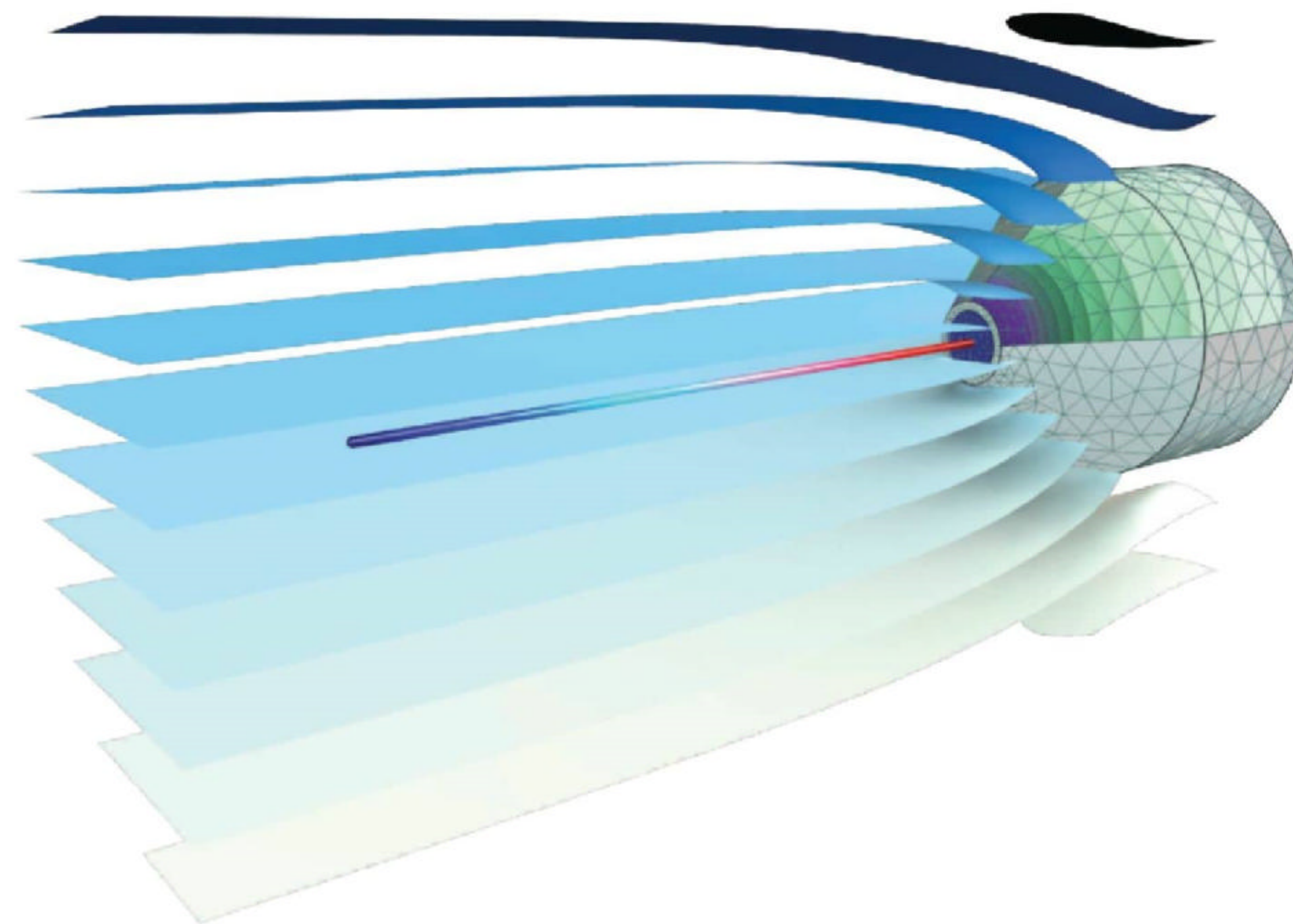
Another factor is the IKEA Syndrome, where the family takes home a shiny new item only to find its assembly and/or installation to be a profound puzzle. Frustration ensues. Over a decade ago, when HTIBs were new to many households, a UK study cited by Philips Research revealed that 47%

of consumers failed to set up their home theater systems correctly, often omitting a speaker, and consequently experiencing inferior sound. I doubt that outcome has changed. Simple and streamlined are two very strong arguments in favor of a soundbar.

Basic two or three channel examples can be had for an average of \$220. They will be about 33" in length, and weigh about 5 lbs. When it comes to soundbars, increased weight usually correlates to the number of drivers and size of their magnet structures. More drivers and bigger magnets usually mean improved low-frequency output and/or reduced distortion.

Entry-level products support Dolby Digital, the home theater equivalent of very low data rate MP3, and the front channels only at that. For a slight upcharge, fancier models support some semblance of true surround, with separate rear channel speakers but then you're verging on HTIB territory. Separate subwoofers are also common, since a diminutive and lightweight bar isn't going

Hearing aids can't solve the cocktail party problem...yet.



Visualization of the total acoustic pressure field around and inside an elastic probe tube extension attached to a microphone case.

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The sleek yet massive AMBEO soundbar performs well above its peer group.

to cut it, bass-wise. In this price range, distortion is high, frequency response is narrow, and so is the overall subjective sound for both bandwidth and soundstaging. One questionable advantage, even at this price point, is that many offerings are “hearable,” with Amazon or Google tech built in. Just say, “Alexa, this soundbar is anemic!”

For a major performance boost, some soundbars take on more weight, more drivers, and more powerful amplifiers to widen the frequency response and decrease distortion while increasing the channel count and sound pressure before clipping. For this



The room calibration process is essential to adapt the sound to the specifics of a living room. The supplied wand-mounted probe mic’s hinged base makes it easy to position at ear level.

category, prepare to add a cool thousand dollars to the equation. Higher-quality wireless soundbars, Bowers & Wilkins’ Formation fall into this category, are also usually more than \$1,000. The Formation subwoofer is also \$1,000, which doubles the price but enhances the bottom half octave.

AMBEO

Enter AMBEO, Sennheiser’s answer to soundbars for upscale video enthusiasts. Better known in the consumer audio community for its headphones, Sennheiser is a 70-year-old company manufacturing a wide range of professional microphones, pro and consumer headphones, and studio monitors. For its first foray into the already crowded soundbar market, Sennheiser wanted to make a statement, a rather substantial statement.

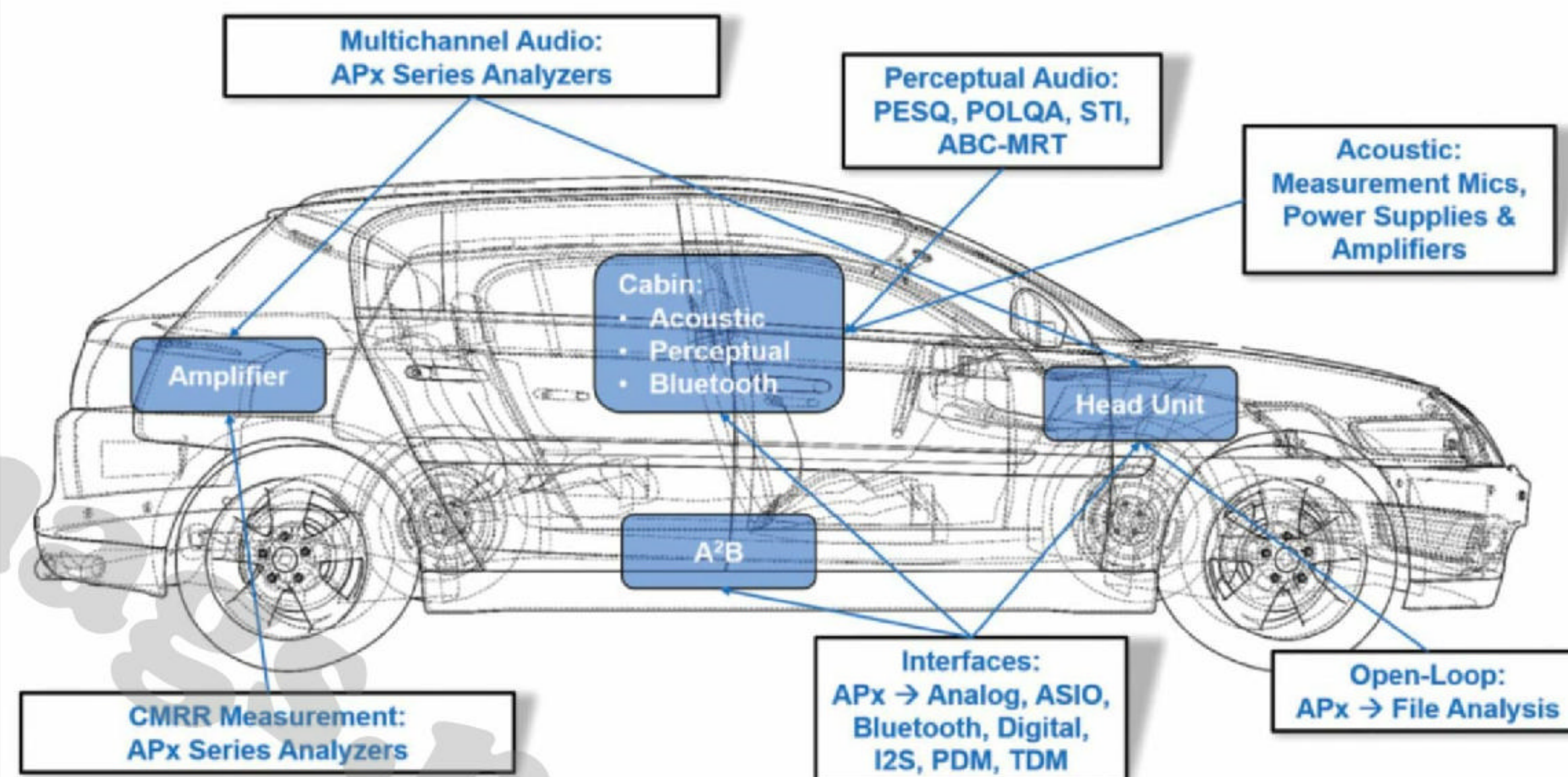
We are talking about a lifestyle product so separate cabinets, wires, and anything that smacks of clutter is a negative. While a traditional soundbar can provide the center, left, and right, how does one deliver those enveloping rear channels without unsightly infrastructure? In a word: beamforming.

Long used for radar and, more recently, larger audio installations particularly corporate conferencing systems and sound reinforcement, this implementation of phased transducer arrays combined with DSP enables a designer to create directivity through constructive and destructive interference. A phased array sidesteps lenses and horns to control directivity and, unlike a physical waveguide, beamforming is “steerable” in that DSP can alter the phase and amplitude of drivers to optimize directivity for a particular listening position. Tabletop hearables also use beamforming in combination with a microphone array to direct their “attention” to a particular talker in a room.

I first encountered a beamforming soundbar at the 2018 AXPONA show, where my geek buddy Mark Waldrep was demoing Comhear’s diminutive YARRA 3DX personal soundbar. Well before that, Philips published in 2007 and was granted a patent for audio beamforming technology in 2012. Five years later, Sony showed its HT-ST5000 beamforming soundbar at the 2017 CES, and JBL now offers a slim version with wireless subwoofer for \$500.

Because our hearing spans so many octaves, it’s a challenge to achieve wideband beamforming. At the high end, you need closely spaced transducers for those short wavelengths. At lower frequencies, you need a sizable array. The usual approach, to keep the cost and physical size reasonable, is to deploy many inexpensive drivers and throw a goodly amount of DSP at the problem. In the case of Sennheiser’s bar, there are 13 drivers total, of three

Automotive Audio Testing?



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varieties, starting with six 4" long throw woofers taken from their famed Neumann speaker division. For high-frequency duties, five 1" dome tweeters with fairly deep horns fit the bill, with three in front and two on the extreme left and right, plus two 3.5" drivers on top for height channels. Each transducer is driven from its own amplifier. As you would expect, they are all Class D with a combined power capability of 500 W. Although expecting a vented approach to aid in bass extension, the AMBEO



Position of the soundbar is important to maximize the immersive effect. By capturing the characteristics of a room and its reflective surfaces, combined with the Fraunhofer-developed virtualization technology, the AMBEO Soundbar actually leverages the acoustics to enhance the tridimensional sound experience.



In order to generate an immersive 5.1.4 soundfield, the AMBEO soundbar features 13 drivers, all purposely designed to enhance both multichannel information from surround formats, and render object-based sources with maximum impact.

About the Author

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

soundbar is an acoustic suspension design for better low-frequency transient response. There are two woofers in each chamber, while the tweeters also have their own enclosures.

The Setup

Upon unboxing, I thought I'd better check for firmware updates. The documentation that came with the product mentioned Sennheiser's Smart Control app for iOS and Android, so I went to the App Store and grabbed a copy. With the app on my iPhone, connected via Bluetooth and on the same Wi-Fi network, it showed a System Settings section in Preferences that includes a "Search for update" choice. Sure enough, my AMBEO needed just that. Though the app itself gave little or no user feedback, the soundbar's own bright and highly legible front panel OLED display showed that an update was in progress.

With that process completed, we almost settled in for listening. I say we since my whole family got into the act at one time or another. At this point, I digress regarding Nvidia Shield TV Pro, about which the youngsters in our family were salivating.

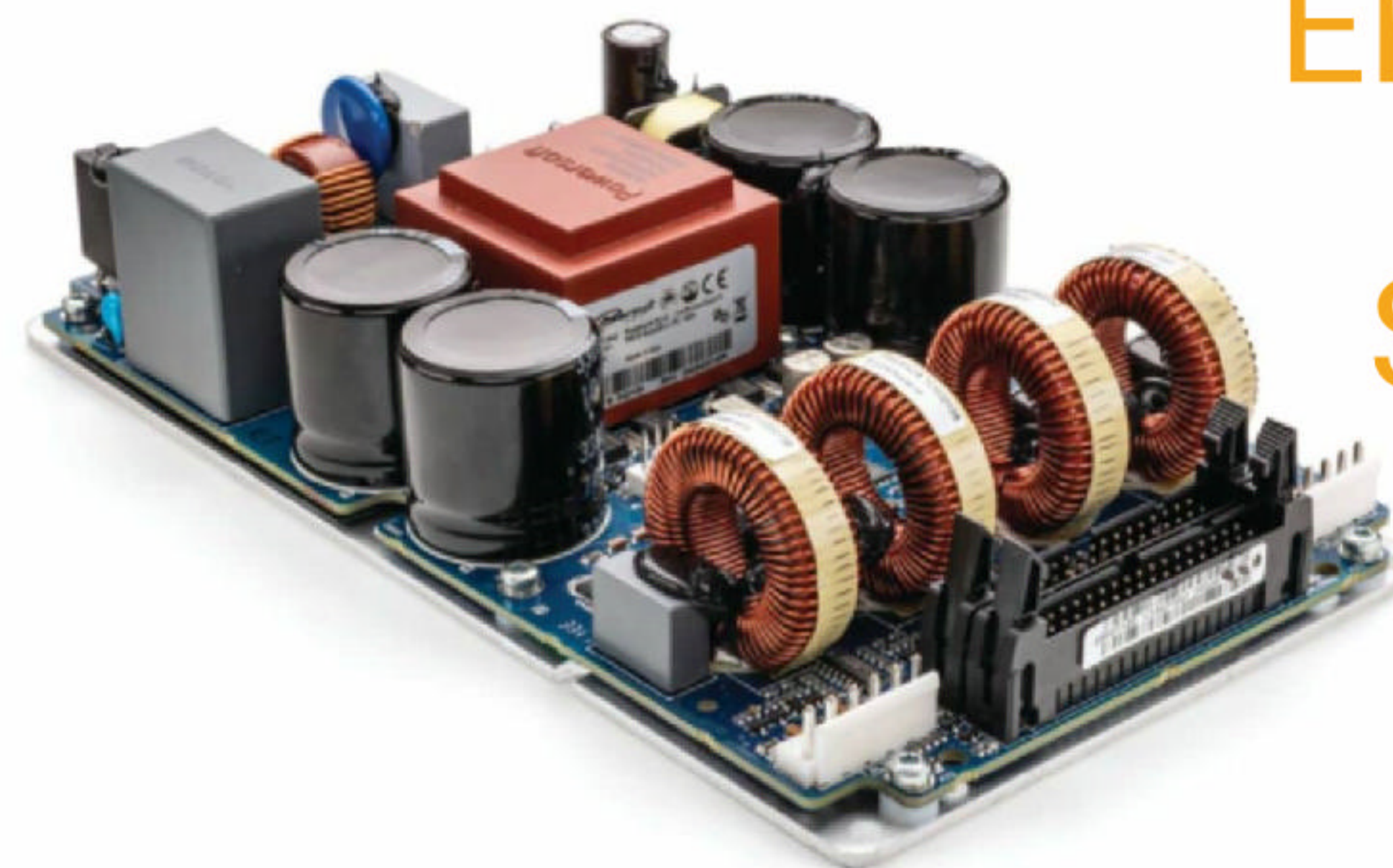
During the Audio Engineering Society (AES) New York conference last October, *audioXpress* editor J. Martins introduced me to Robert Bleidt, Fraunhofer USA's Division General Manager of Audio & Multimedia. Bleidt mentioned that he could provide content that would nicely demonstrate the capabilities of the Sennheiser product.

Fraunhofer was a development partner with Sennheiser, working on implementation and product refinement. Bleidt agreed to send me a piece of hardware to play back Fraunhofer's MPEG-H demo content and, after a fair bit of red tape, a Nvidia Shield TV Pro Home Media Server arrived at the house. Ani Maria, my in-house gaming expert, helped me set up the sleek little black box and, we finally did settle in to listen.

I first arranged the unit in our living room, a decidedly asymmetrical arrangement. Though an optional wall mounting bracket is available for the AMBEO Soundbar, it was not included in the demo package so I simply placed the product on the horizontal surface nearest to the television. The living room's left wall is 7' from the sweet spot while the right wall is more than 2.5' farther.

The product comes equipped with a standalone measuring microphone mounted on a 28" wand, making it very easy to position the mic at ear level. After a couple of calibration passes followed by listening, I found that placing the microphone about 3" in front of where my ears would normally be positioned produced the most satisfying results.

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Application Examples



AMBEO's Modes

Once the very simple calibration process was completed, we broke out the streaming content and Blu-ray movies. In that environment, I heard a surprisingly rich envelopment, though discrete localization from behind was lacking as one would expect from discrete ITU-positioned rear-channel speakers. To be fair, my ears were a mere 11" from the back wall so, not much opportunity for AMBEO to properly bounce that rear information.

The product has five listening modes or "profiles." These are more than voicing settings in that they combine timbral shifts with changes to the

AMBEO algorithm. Subjectively, the Neutral setting is, as you would expect—balanced and uncolored. When compared to my other hi-fi tabletop speaker, Naim's Mu-so Qb, the overall voicing was similar, though the Qb has a more extended high end and sounds a good bit punchier.

The second listening mode is Music, which adds a slight upper low-frequency lift. Sports mode adds a slight high-frequency lift to Music's low-frequency boost, while News mode exhibits a slight upper low-frequency suppression plus that high-frequency lift. Finally, Movie mode offers a pronounced low-frequency lift along with enhanced L-R steering. Think of it as Fun mode, as it was the most popular mode for the family regardless of what flavor of entertainment we were consuming.

The app has one control that I found particularly useful: the three position AMBEO intensity control. Depending on content and number of listeners, we found that we could always find an appropriate and satisfying setting that would spread the soundfield around to every listener or concentrate it for solo viewing.

The AMBEO Soundbar also has a Night mode, which can be enabled for almost any listening mode. As one would expect from the name, Night Mode rolls off low frequencies, and seems to dial in compression about a rotation point. Along with the low-frequency suppression to keep the material from traveling too far through the house, this setting seems to reduce the overall dynamic range while lowering the perceived Loudness.

A "Dolby Dialog Normalizer" can be activated or deactivated in the Smart Control app. According to the user manual, "...this function makes dialogs (sic) easier to understand by reducing loud music/background noise and amplifying softer dialog audio." Dolby DRC or "Dynamic Range Compression" is also included and, again, controllable by the app. "This function compresses the dynamics of the audio signal, thus changing differences from soft to loud passages." Setting choices include fully off with no change to dynamic range, "Auto" with automatic reduction of dynamics, "Normal" with slight compression of dynamics, and "Heavy" with brute force compression. If that's not enough, a four band EQ is also included in the app, so extensive personalization is possible.

The Remote

As with any lifestyle product, a remote is expected and, in this case, included. Unlike its companion bar, the all-metal remote is small, with a satisfying heft and easy-to-understand layout. The bar also has a five-color Status LED that visually

confirms many of the remote's functions. All of those functions of the remote are replicated by the app, along with configuration controls and three extra features. One is the above mentioned ± 10 dB, four band graphic equalizer. The second is a "subwoofer" enable/disable switch for an optional external sub, while the third is a three-position intensity setting for AMBEO. With those three controls, the user is able to modify and store the five presets to taste. The stored configurations replace the factory setting until the user resets any individual mode. The soundbar has an "ECO" standby mode, and you can set the mode via the app. Booting the soundbar takes quite a bit longer when ECO standby mode is enabled.

The Soundstage

With AMBEO off, the soundstage collapses to a conventional stereo presentation. Imaging is reasonable as long as you're on the speaker's centerline, while timbre is neutral if a bit bass-shy. Considering that the outer, front-facing speakers are only 39" apart and the typical listening distance is 5' to 8', in stereo mode the speaker subtends a very narrow angle. You have to sit very close to the speaker to hear "normal" stereo imaging in stereo mode.

Enabling AMBEO mode drastically enlarges the perceived acoustical environment, a bit like the difference between what a dipole paints compared to a box loudspeaker. The algorithm jacks up the interchannel difference information while activating the upward-facing Atmos height speakers along with the beamforming. In short: bigger, wetter, and much mo' better. The timbre also shifts, adding a dram of upper bass heft. Overall, *mucho más* excitement. Movie mode plus AMBEO is crazy good fun. We pretty much kept AMBEO enabled all the time, since going back to plain old stereo is a colossal letdown.

Soundbar Doubles as a Hot Spot

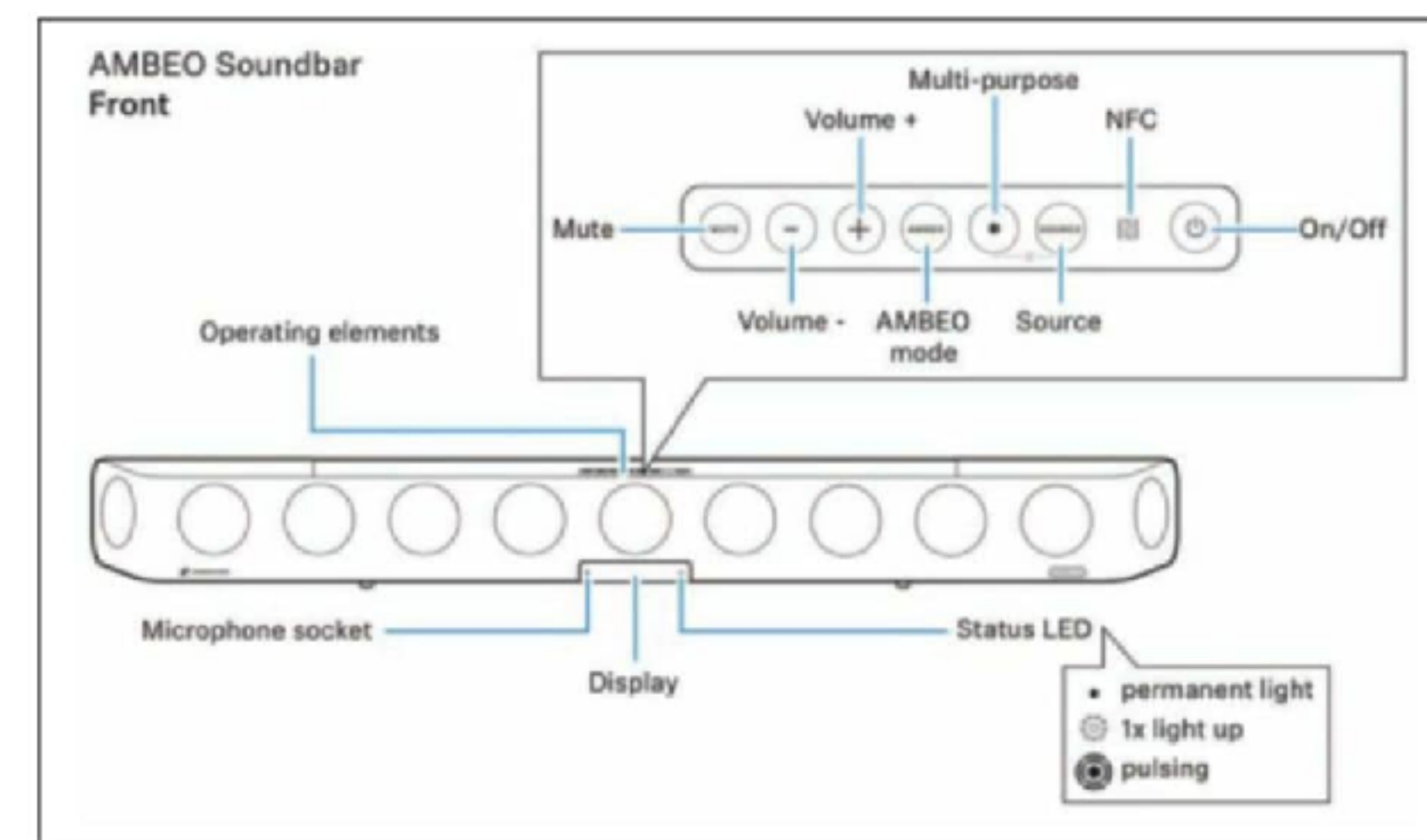
The soundbar is an 802.11 hot spot. Used for Google Home setup, Chromecast moves configuration data to the soundbar during user configuration. The AMBEO Soundbar also includes wired Ethernet, a digital optical input, an analog in on RCA, three HDMI inputs and one eARC-enable HDMI output. Bluetooth Low Energy (BLE) plus NFC connects the smart phone app to the soundbar. A USB spigot is dedicated to firmware updates. The soundbar and remote have a multi-function button that allows you to control playback of a Bluetooth source. Two DSPs are hidden in the soundbar, one for up to 10 channels of Atmos & MPEG-H decoding, and a second for frequency contouring.



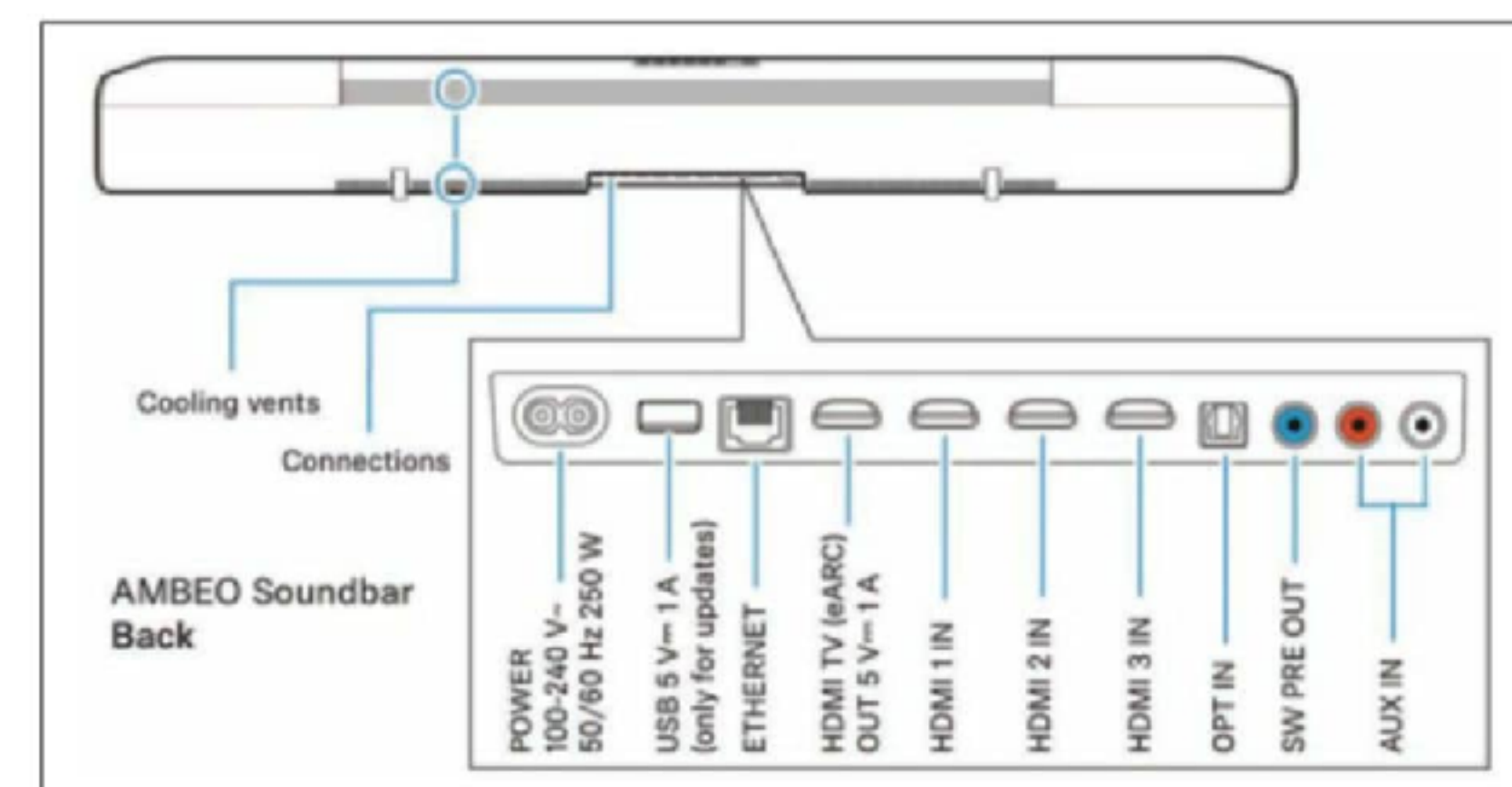
The five dedicated high-fidelity tweeters are strategically positioned to generate the widest projection, including two side-firing drivers. On the top extremities of the soundbar, two additional up-firing drivers generate the extra dimension.



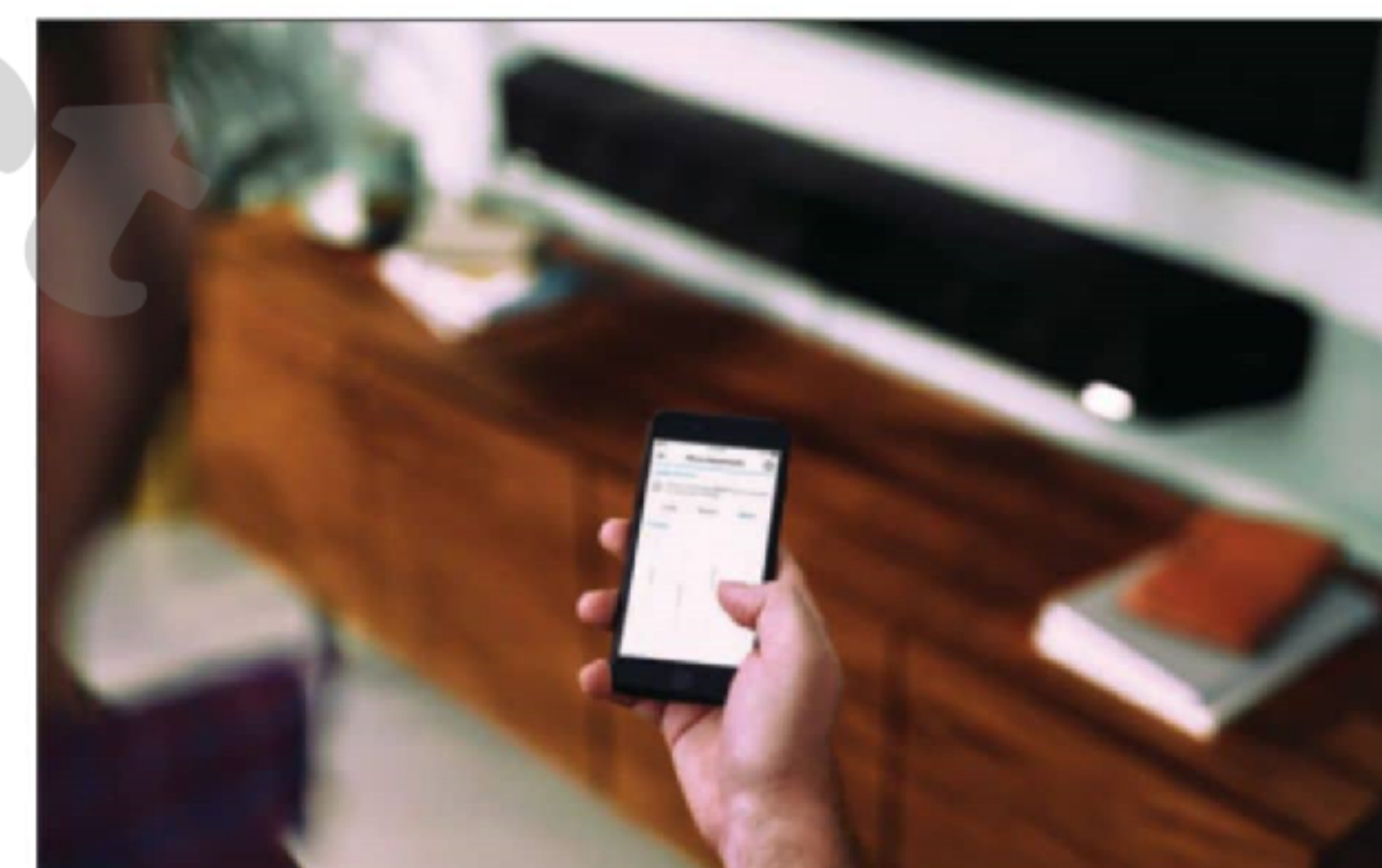
The Nvidia Shield TV PRO includes Dolby Vision HDR and HDR10 as well as Atmos, MPEG-H, and DTS-X.



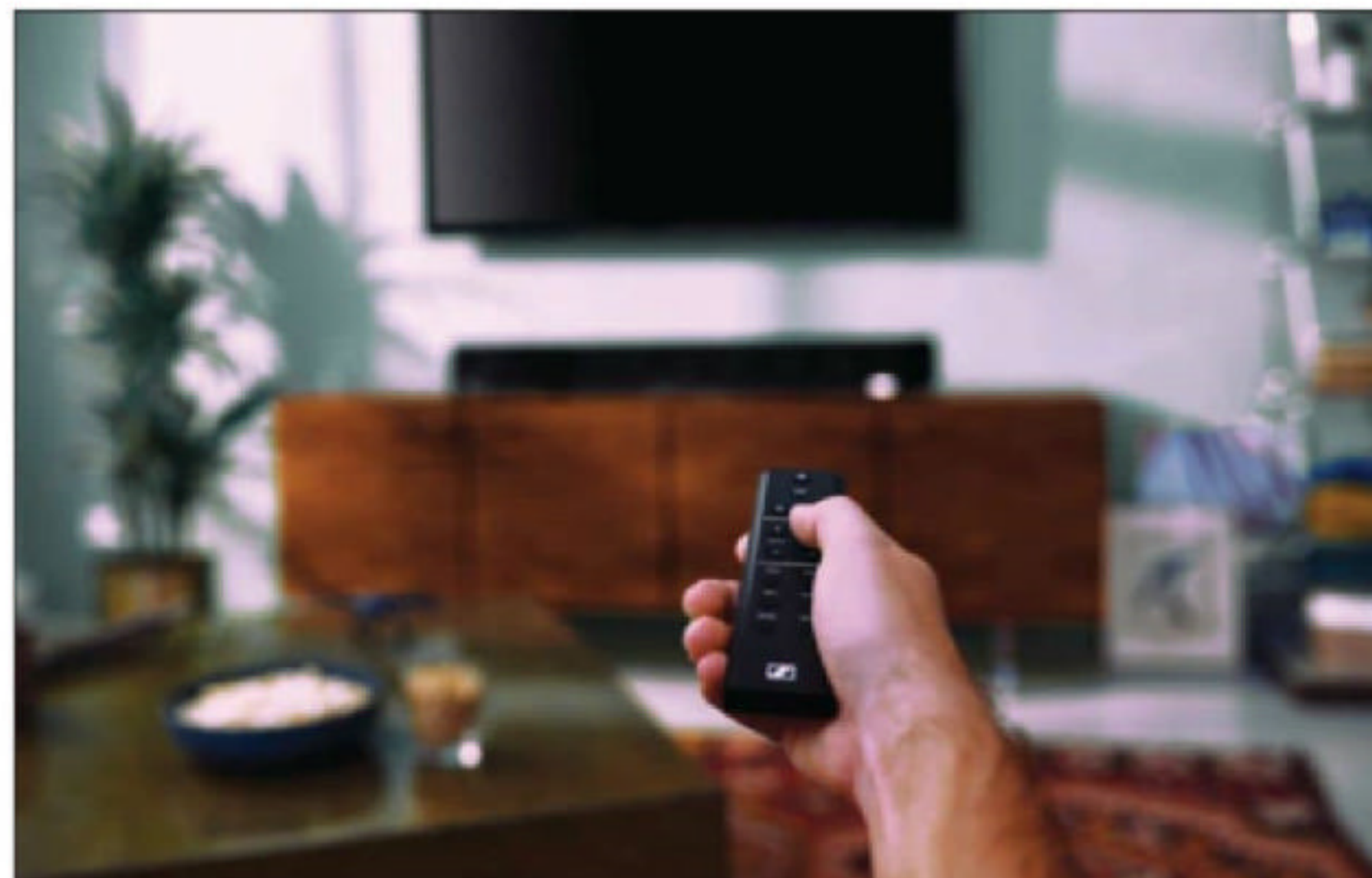
The controls are clearly marked and their operation straightforward, no fishing through nested menus.



The nine I/O ports take care of most any installation situation. There's support for the latest HDMI standards—eARC and Dolby Vision—plus 4K passthrough compatibility, Bluetooth and Chromecast is built-in.



With the Smart Control App, users can adjust the acoustic settings and the built-in equalizer, and tailor the sound to personal taste.



Alongside Dolby Atmos, DTS:X, and MPEG-H, the AMBEO Soundbar's Upmix Technology can turn stereo or 5.1 audio into a 3D sound experience.



The minimalistic look and brushed aluminum surface of the soundbar doesn't do justice to its power.

For stereo content and with AMBEO off, a downmix to stereo is produced; front left and right tweeters plus all six woofers. Enabling AMBEO initiates upmixing from stereo. As Max Voigt, Product Manager for Sennheiser's CE Division described it, "You can hear what immersive sound can bring." Truly an understatement, I came to love AMBEO!

Overall Impressions

For a more typical listening experience, we moved the bar to our basement rumpus room, where

my discreet 5.1 system usually serves yeoman audio duties. Our downstairs has a 7' popcorn ceiling, while the TV room's rear wall is largely covered by a floor-to-ceiling glass front bookcase. Placed in a symmetrical position on the centerline of the room, with the rear wall some 8' from the listening position and the bar 6' in front of the listener, I expected better rear and height localization but was again a bit disappointed.

Voigt was not surprised, telling me that there is only so much that can be done with virtual speakers. As he said, it is "...a disadvantage of not having dedicated rear speakers." Given that caveat, the overall perceived envelopment was outstanding, bested in the horizontal plane only by a discreet ITU setup.

The AMBEO Soundbar adds a somewhat vague but satisfying vertical dimension to the surround experience. No, you most likely will not hear a "Voice of God" channel as you would in an Atmos theater, but you will be swathed in a rich quarter sphere of clean, exciting sound.

One aspect of the AMBEO Soundbar experience that needs to be emphasized is that this is made by Sennheiser. This is a Teutonic product, built by engineers who usually create top-shelf audio equipment. A characteristically thorough design process has produced voicing and performance that reflects the company's 75-year history of producing products without compromise or euphonious distortion.

When asked what the development group was most proud of, Voigt focused on project administration and the result of a distributed team effort. "We were able to manage the complexity, to get a really fine product into the market." He stressed the "really great design" that the acoustical team produced, and also the cooperative interaction between Sennheiser and Fraunhofer, especially for AMBEO.

With three years of development and over a million lines of code, Sennheiser's AMBEO Soundbar brings well over a half century of combined wisdom and craft to the home theater market. The result is a category-defining product, with a wealth of useful features and outstanding dynamics and crowd-pleasing excitement along with exceptional clarity and linearity.

The Upside: exceptional fidelity, legendary build quality, peerless spatialization of mono and stereo sources, versatile input and format support, and the all-in-one form factor.

The Downside: relatively high cost, no hard "rear" localization, and the mic placement critical to optimal setup. 

Resources

Sennheiser AMBEO
<https://audioxpress.com/tags/AMBEO>

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www.iis.fraunhofer.de/audio

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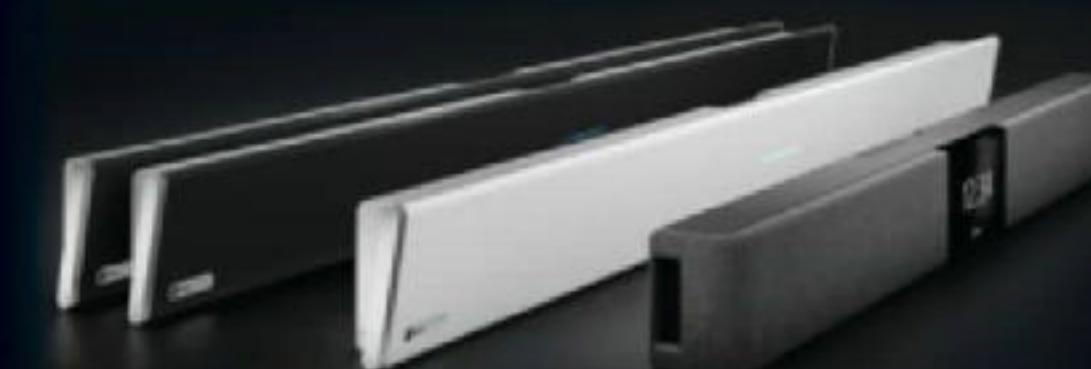
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A Review of *Hi-Fi: The History of High-End Audio Design*

For those audio enthusiasts looking for a good read showcasing some wonderful examples of design and craftsmanship, Mark Driedger suggests taking a look at Gideon Schwartz's new large-format book.

Review by
Mark Driedger
(United States)

Hi-fi audio appeals to us in many ways. The fidelity of the music to our ears, the emotion of it to our hearts, the constantly changing technology to our intellect, the craftsmanship and design to our eyes and sense of touch, with the stories of the people behind the products adding a touch of humanity.

I recently read Gideon Schwartz's new large-format book, *Hi-Fi – The History of High End Audio Design*, with hopes that it would appeal to all but my ears.

Schwartz's book is an 8.5" x 11" hardcover "coffee table book" format with 272 heavy glossy pages of text and nicely produced photographs. Text is only two-thirds of the page width, with lots of white space. The Introduction takes us from Edison through to 1950s. Thereafter, each chapter covers a decade. Specifically, "1950s - Industrial Stereo Utopia," "1960s - Stereos Get Sexy," "1970s - The Birth of High-End Audio," "1980s - High-End Audio in Full Bloom," "1990s - Return of the Vacuum Tube," and "2000s - Post-Digital Analog Renaissance." Despite

Hi-Fi: The History of High-End Audio Design
By Gideon Schwartz
Hardcover: 272 pages
Publisher: Phaidon Press
October 30, 2019



Gideon Schwartz recently wrote a tome discussing "The History of High-End Audio Design."

the book's October 2019 publication date, the past two decades get lumped together.

Inside the Book

Introduction: The introduction starts with Thomas Edison and focuses mainly on acoustical recording and the first "streaming" service—French inventor Clément Ader's "Théâtrophone," which transmitted sound on a pay-per-use basis from theaters over phone lines. Ader describes the concept of stereo in an 1890s patent (two mics, two speakers, and two ears). The narrative between 1890s and 1950 is sparse and jumps around.

Topics covered include the Bell recording experiments of the 1930s, a one sentence mention of the gramophone, Zellaton speakers, early Nordic design ethos, and a sampling of Art Deco tabletop radios. The invention and application of the vacuum tube is not covered, although Schwartz does give us a photo of nice pair of Western Electric 49C amplifiers. While one can argue there was little in that era that ranks as "Hi-Fi" today, there were a few attempts, such as the Zenith Radio Stratosphere and Crosley Radio Corporation's Colossus (circa 1936) which had three-way speakers, 37 tubes, 75 W of output power, and weighed 475 lbs.

Technology: Schwartz covers the biggest technology innovations in Hi-Fi: acoustic to electrical recording, magnetic recording, stereo, vacuum tube

to solid state, MP3, and both acoustic suspension electrostatic speakers. That said, there is a lot that he left out and readers who are more technical will be disappointed. The list of missing technologies is long: Williamson and ultra-linear amplifiers, MOSFET amplifiers, transmission line speakers, the industry's flirtation with quadrophonic audio, direct-to-disc recording, direct drive turntables, and improved digital recording and playback (e.g., Neil Young's Pono) to name a few. There is a lengthy discussion of the initial concept of stereo circa 1890; however, no mention of Alan Blumlein's 1933 patent and a mere one sentence mention of stereo LPs (despite the very interesting history of failed Rube Goldberg approaches to putting stereo channels on a disc).

Technical details are infrequent and obtuse. For example, "Class A operation in an amplifier uses 100% of the input signal from a power supply, with no connected elements required in the circuit itself." Descriptions of solid state vs vacuum tubes are similarly obtuse, and that of electrostatic speakers is misleading, if not wrong. Schwartz states that "When the grids are driven by an audio signal, this, in turn, causes a movement of the air on each side of the diaphragm." In fact, the grids don't directly cause the air to move. They cause the diaphragm to deflect, which in turn moves the air. A half-page is spent on pitch, unnecessary for anyone buying this book, while many other terms and acronyms are left unexplained.

The entire DIY and kit movement is absent, save for a short discussion about the Dynaco ST-70. There is no mention of Heathkit or the Hafler DH-200, both of which were state-of-the-art in their day. There are a handful of internal product photos to whet the appetite of the more technical readers.

Design: The photographs do a very good job visualizing the evolution of the design and style of Hi-Fi products through the decades. Schwartz places the design changes in the context of the social environment of the day—early adoption of Hi-Fi components by actors and musicians of the day, the bold visual design of the sexy 1960s. There is not a connected narrative on the topic though—the earlier chapters start with a few paragraphs that set the context for the decade. Several of what I consider to be pivotal visual designs are missing, such as the Yamaha TC-800 cassette deck, the Mark Levinson No 33 power amplifier, and Lecson Audio's products. However, Schwartz does a good job showcasing the best in craftsmanship in the photographs.

Companies: Each chapter is organized as a series of short articles with accompanying photographs for 10 to 20 companies. The better of these contain interesting histories of the individuals behind the

companies and the company origin stories. They range from a short paragraph to a page in length. I felt the quality of these are better in the earlier chapters. In the final chapter, the companies are at the extreme esoteric (read expensive) end of the spectrum and are almost exclusively ones that Schwartz sells in his New York audio boutique store. These feel to me more like advertising than a history.

There have been a lot of Hi-Fi companies in the last 70 years and any book must cover a subset. There does seem to be a strong bias to Swiss companies ("The micro-precision nerve center of the audio world" in Schwartz's words) and to companies that Schwartz represents in his store. There is very little coverage of any Japanese companies until the final chapter, and then only the very esoteric. A few missing companies are Rega, Oracle, Cambridge Audio, Denon, Stanton, and Luxman. Surprisingly the (Swiss) firm Nagra, which produced some of the most beautiful and precise portable tape recorders, is omitted.

Overall Impressions

Schwartz ends the book abruptly, finishing on one of the company threads in the 2000's chapter without any coda or commentary. The opportunity to provide some context on the arc of 70 years of Hi-Fi or its future direction is missed. I was left wanting some form of conclusion.

Good books on audio history are few. *Sound Design* by David Attwood does a better job of telling a connected story of the visual design history, although it showcases fewer high-end products. *Tube Amplifiers* doesn't pretend to provide a history—just a collection of excellent photographs of vacuum tube amplifiers. *The Art of Sound – A Visual History for Audiophiles* by Terry Burrows covers a broader range of products (consumer to high end) and more of the popular culture. There are also a few good books on individual companies: Quad, KEF, McIntosh, and Rega are the ones of which I am aware.

All said, I would purchase Schwartz's book again for the photographs and company anecdotes, despite the lack of a connected design or technology history. It provided a nice walk down memory lane, and showcased many fine examples of design and craftsmanship. 

About the Reviewer

Mark Driedger has been experimenting with tube audio since 1980. He completed an MSc in electrical engineering at the University of Waterloo, Canada, and has worked in the telecom industry for more than 30 years in technical and executive roles. His other passions include woodworking, the guitar, and spending time with his wife, children, and grandchildren.

A Continuous Calibration Method to Optimize Conference Call Acoustic Echo Cancellation Performance in Real Time



Is it an understatement to say that echo cancellation has been a kind of “holy grail” ever since the first conference call? Nureva’s patented Microphone Mist technology rethinks that quest in innovative ways. In this article, Erica Yorga shares how the development process awarded surprising results.

By
Erica Yorga
(Nureva)

Conference call quality has been an area of concern since the beginning of remote communication between individuals and/or teams. Conference calls can be plagued with numerous technical issues, from poor microphone pickup and placement to properly hearing participants, and adequate noise control to connection and bandwidth problems. With these issues in mind, establishing and maintaining a high sound quality and a reliable conference call experience has proven to be very challenging. Somewhat compounding the problem, product designers need to develop products that can work in as many spaces as possible, as you never know how your product is going to be used and installed.

A common issue with conference calling is a phenomenon called acoustic return echo. Nureva has developed a unique and innovative approach to effectively deal with acoustic return echo by adapting and calibrating the acoustic echo canceller in a manner that other systems are unable to achieve.

Did You Hear an Echo, an Echo?

Acoustic return echo is the effect of callers hearing their voices in their headset speaker delayed in time after they have spoken, which sounds just like the echo you may hear in a canyon or very large room. Say you are at the Grand Canyon and you yell out the words, “hello out there!” Sometime later you will hear the same words echoed back at you, say maybe half a second to a second later. If the echo is really bad, you might hear the same phrase echoed back at you more than once ... maybe two or three times. Now, if this phenomenon happens to you during a conference call, it does not take much imagination to realize how bothersome this will quickly become, resulting most likely in listening confusion and poor intelligibility of the voice from the far end. The person at the far end of the call does not hear this echo as you do, so if the person starts talking before the return echo has dissipated, you will hear both your echo and the other person’s voice at the same time.

My Room is Like the Grand Canyon

When a sound is produced by conference system speakers, it bounces around the room and is picked up by the system microphones as illustrated in **Figure 1a**. If this effect is not dealt with, an acoustic return echo signal is generated. If there is no echo canceller circuit to eliminate this acoustic return echo signal, the remote participants hear their voices delayed in time in their earpiece—kind of like what happens when you yell into the Grand Canyon.

Each speaker and microphone combination has its own unique set of acoustic echo path delays that are determined by the acoustic properties of the room. The sound emitted from each speaker takes a different path (room bounce) to each individual microphone, creating a unique acoustic echo path delay for each microphone. The more speakers and microphones you have installed, the more acoustic delay combinations your system creates, resulting in numerous acoustic return echoes that must be cancelled. With so many variables and possible combinations, this is not a trivial problem to solve (see **Figure 1b** and **Figure 1c**).

Typically, the larger the room, the longer the acoustic delay will be. The room acoustic delay, also known as echo path delay, is a measure of the time in milliseconds it takes the signal to travel from system

loudspeaker(s) to bounce around the room and be picked up by the system microphone(s). Small rooms have a short delay and larger rooms have larger delays. To add to that, a room can be dead sounding (minimally reverberant) or lively sounding (highly reverberant). The echo canceller needs to be able to handle any scenario, which adds to the processor complexity and system memory requirements.

We Can Adapt

A common approach to deal with return echo issues is adaptive echo cancelling as illustrated in **Figure 2**. Adaptive echo cancelling has well-known constraints limiting its effectiveness to adapt, which can result in poor call quality. At a basic level, adaptive echo cancelling compares the speaker output signal that contains the remote talker conversation referred to as remote sound $x(n)$ to the microphone input signal $d(n)$ that contains the remote talker reflected sound $h(n)$ by the room. The adaptive filter measures $e(n)$ and adjusts the echo canceller parameters $y(n)$ over time with the goal of reducing the level of the acoustic echo return signal (remote sound) before it is sent out by the conference system to the remote participant.

What may not be obvious is that for the adaptive echo canceller to measure and adapt its filter

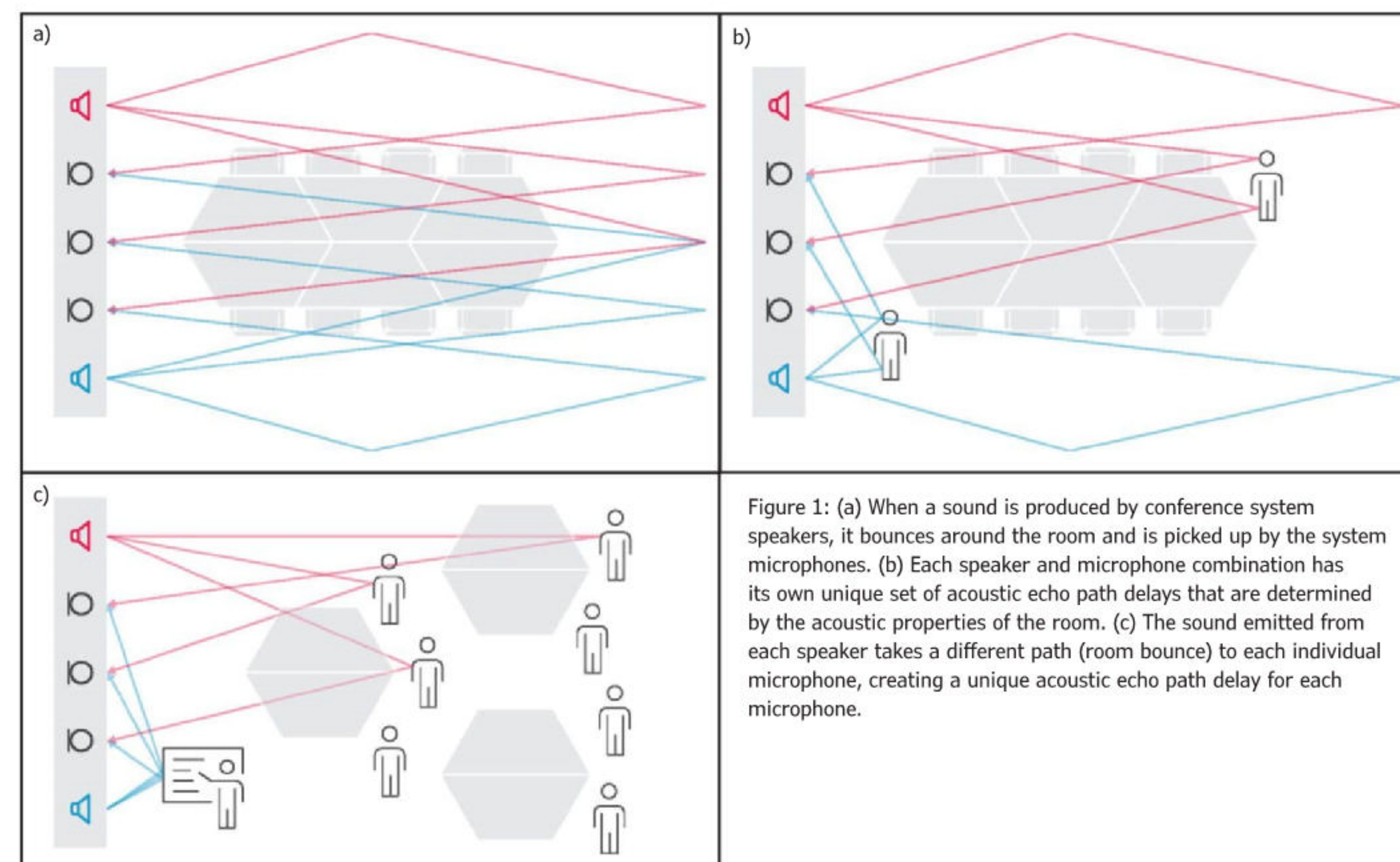


Figure 1: (a) When a sound is produced by conference system speakers, it bounces around the room and is picked up by the system microphones. (b) Each speaker and microphone combination has its own unique set of acoustic echo path delays that are determined by the acoustic properties of the room. (c) The sound emitted from each speaker takes a different path (room bounce) to each individual microphone, creating a unique acoustic echo path delay for each microphone.

parameters, it requires an active far-end talker signal $x(n)$ and an in-room microphone signal $d(n)$ that contains only acoustic echo return of $x(n)$ as an input. This significantly limits the adaptive echo canceller's ability to manage real-time room

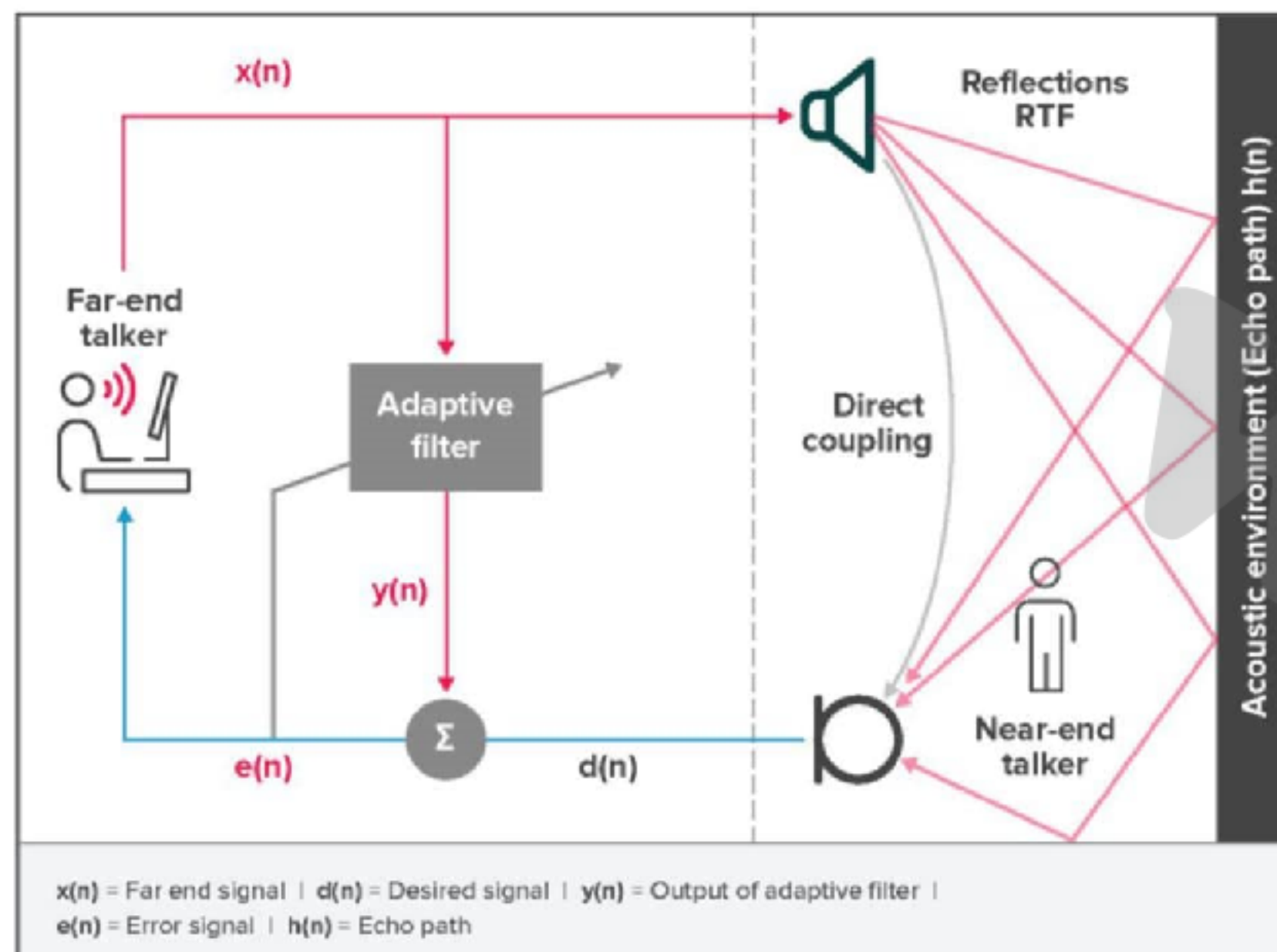


Figure 2: A common approach to deal with return echo issues is adaptive echo cancelling.

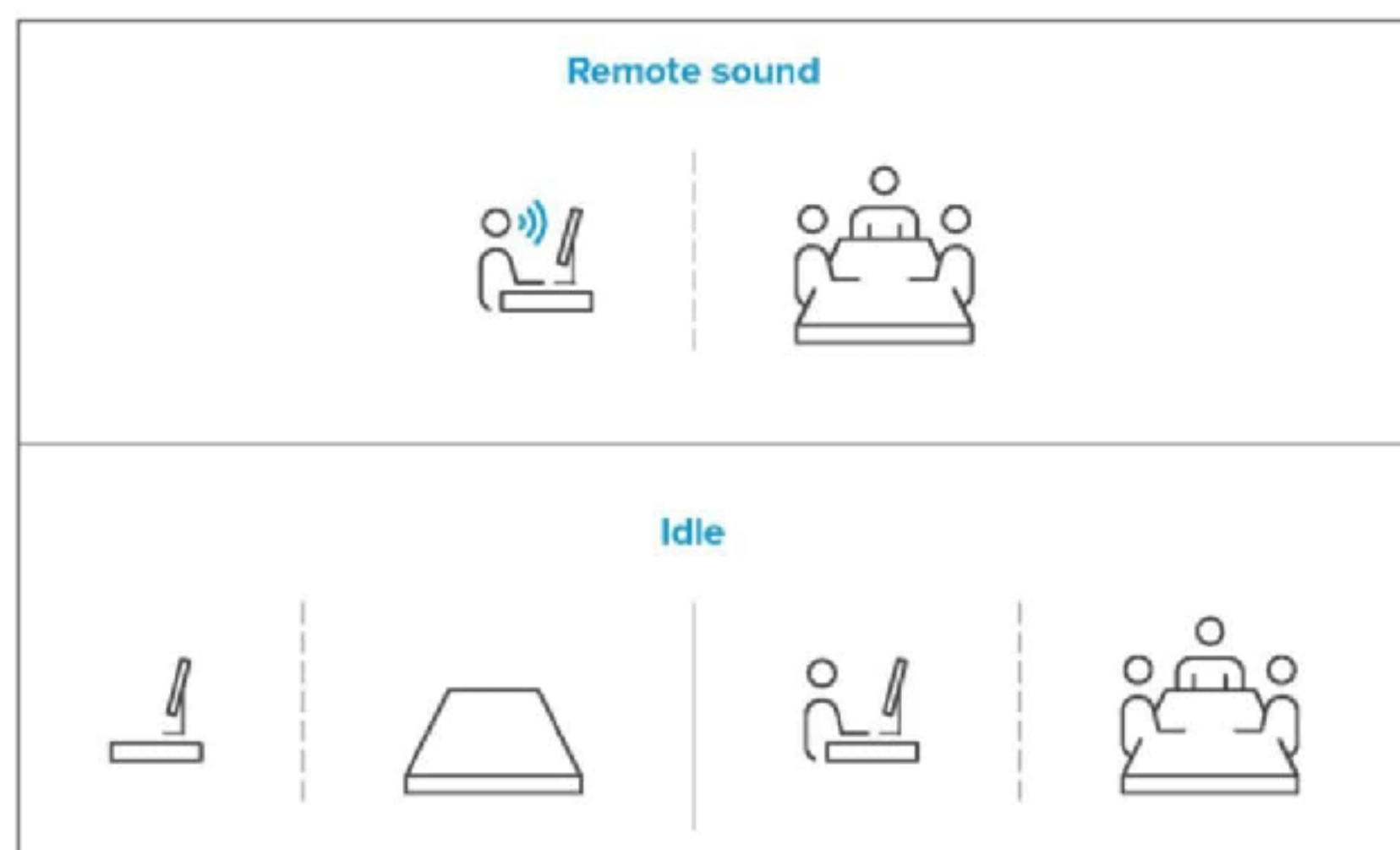


Figure 3: There are four possible states the system can be in at any time during the call.

About the Author

Erica Yorga is the Principal Intellectual Property manager at Nureva, a Canadian technology company. Her career spans a variety of leadership roles researching, developing and commercializing a broad spectrum of technologies in markets such as audio/visual, oil and gas, telecommunications, military defense, and Government regulatory. The challenge of solving difficult problems by constantly learning and enabling teams through the application of unique approaches to technology and process is a passion she brings to every endeavor she undertakes.

changes and provide high-quality conference call performance.

Let's take a slight detour to understand when an adaptive echo canceller can and cannot dynamically adapt its filter parameters. **Figure 3** outlines the four possible states the system can be in at any time during the call. The "remote sound" state occurs when the only person talking is the remote talker. If only near-end talkers are talking, the call state is defined as the "in room" sound state. If the remote talker and the near-end talker happen to talk at the same time, which can often happen, the call state is defined as "remote and in room sound" state, also known as double talk. The final state is referred to as "idle state," because no one is talking or the room is empty.

As it turns out, the adaptive filter (shown in Figure 2) is limited to only calibrating and adapting the echo canceller circuit during the "remote sound" state. Ideally, the adaptive filter is designed to detect non-valid states and pause the calibration routine. This can be difficult to do in practice, which leads to echo canceller parameters that may not be optimal, because the echo canceller is calibrating in a call state that it should not be in, resulting in the echo canceller requiring extra time to recover. All this can, and often does, lead to compromised call quality.

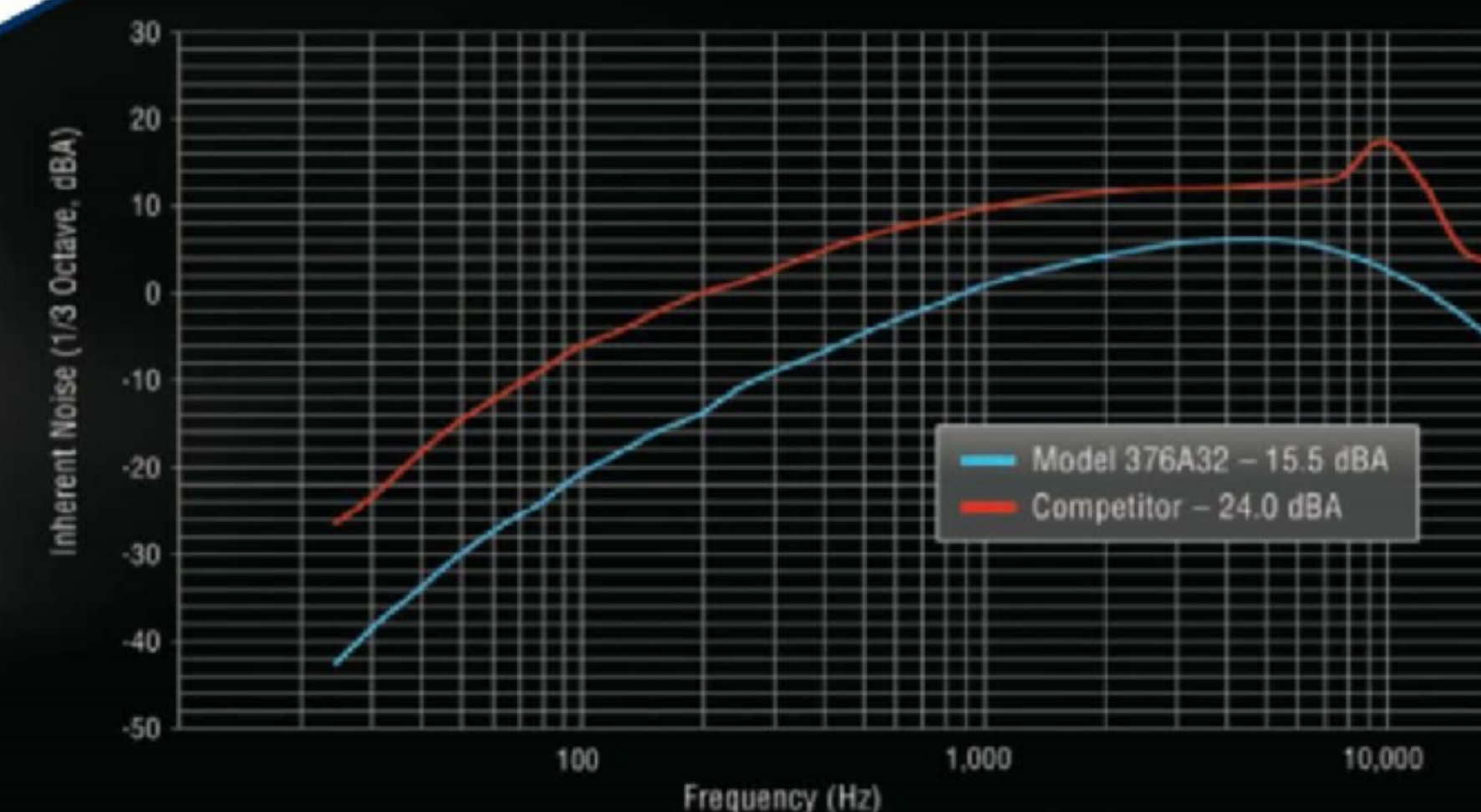
Complex system installations may need to have a technician configure and calibrate the echo canceller system through a manual calibration process that attempts to consider the room acoustic properties. Any changes to the room (e.g., number of people, furniture arrangement, or microphone and/or speaker locations) after the manual calibration is completed could reduce the effectiveness of the manual calibration and may potentially impact the echo canceller performance. Once again, far-end talkers hear their return acoustic echo, which would then require a new manual calibration process.

To undertake a manual calibration, an impulse response (a short duration tone pulse—think of a hand clap or a balloon burst) is transmitted through the speaker system that is picked up by the microphone system. The acoustic return echo delay can be measured and used to calculate echo canceller parameters for the system. The process can be intrusive and time-consuming and is not practical while a meeting room is in use, so the meeting room is taken offline and calibrated.

The Hidden Room Is Exposed

A novel approach to echo canceller calibration and performance is achieved by Nureva's patented Microphone Mist technology. Current approaches and technology seemed insufficient, so we set

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a challenge for ourselves to improve the far-end talker conference call experience, by continuously accounting for situations and environment changes while avoiding the limitations of adaptive echo cancellers requiring only a “remote sound” state and/or, in more complex systems, a manual calibration step. We’re excited to be able to report that we met the challenge!

Using a concept commonly deployed in many

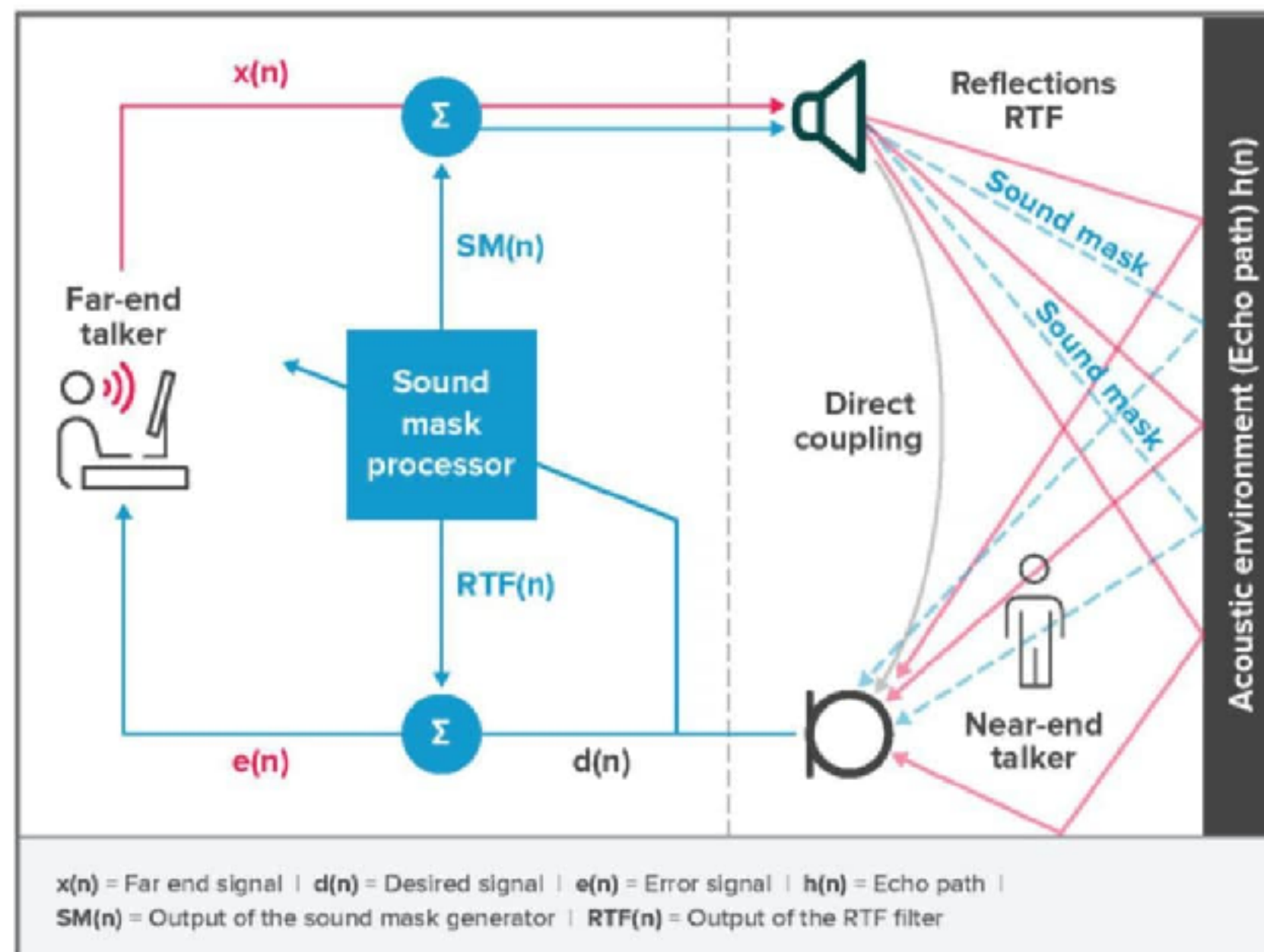


Figure 4: The common adaptive echo canceller stage is replaced with Nureva’s unique sound mask processor.

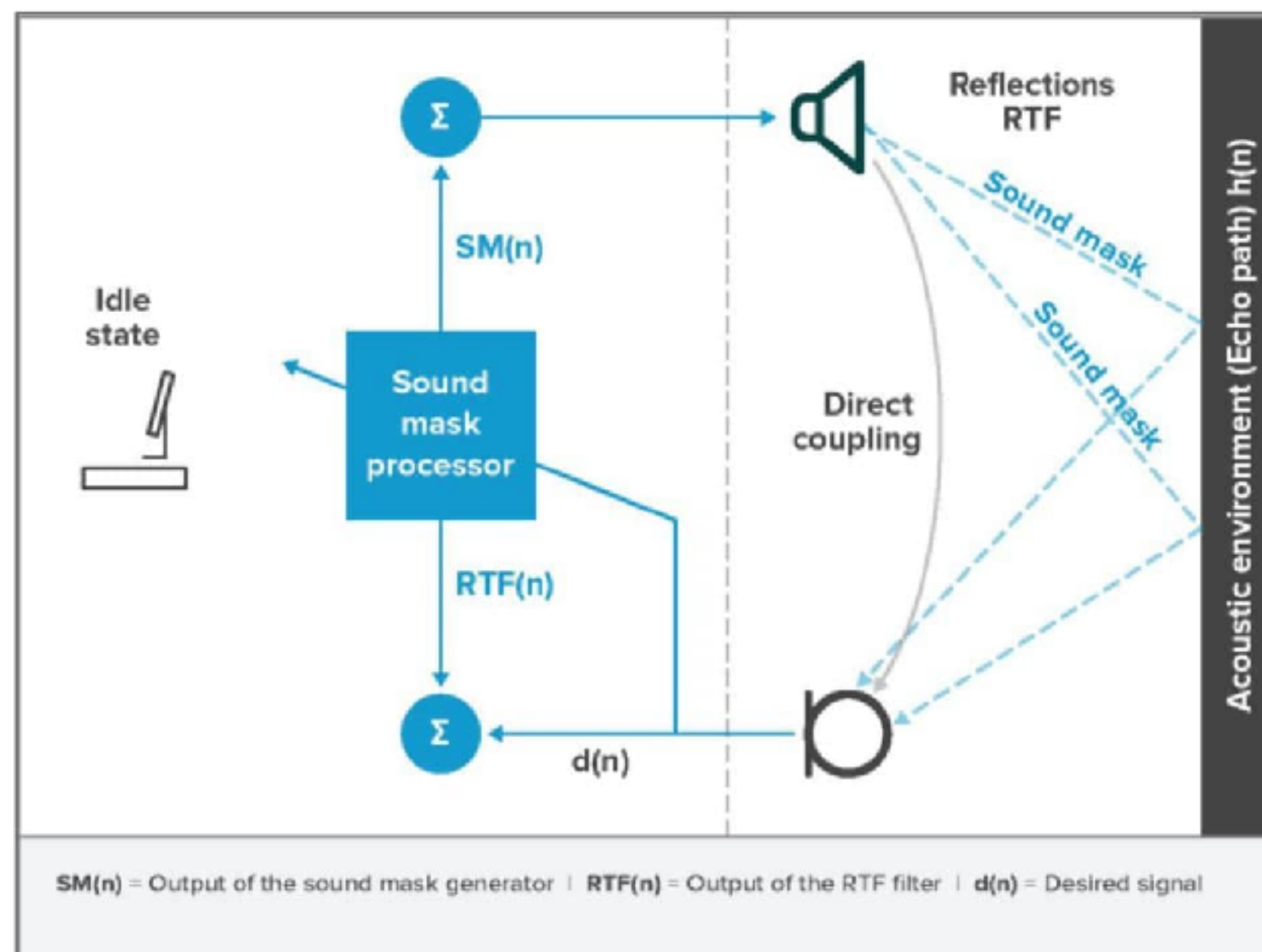


Figure 5: Because the sound mask processor does not require a far-end talker, the conference system is able to adapt and optimize echo canceller parameters when other systems cannot.

commercial environments called sound masking, we can calibrate the echo canceller in situations not normally suited to calibration. The novelty is in using a sound mask-like signal to measure for the acoustic environment in real time for each microphone and speaker combination. Not only is the loud calibration signal not required (which would be impractical when the meeting room is in use), the echo canceller is not limited to adapting during the remote sound state only. The echo canceller also adapts to the idle call state and is ready to perform during all call states. This means that the echo canceller will have consistent performance from the beginning of the call to the end.

A sound mask signal, which is usually some form of shaped acoustic pink noise, is transmitted in commercial spaces to increase the ambient background noise of the environment. Because the sound mask is perceived as background ambient noise, it is easily ignored. By using the sound mask in this novel way, Microphone Mist technology can leverage a common concept to achieve significant increased benefit over typical approaches to echo canceller calibration performance.

Figure 4 illustrates replacing the common adaptive echo canceller stage with Nureva’s unique sound mask processor. The sound mask processor generates a sound mask $SM(n)$ -like signal that is injected into the speaker audio output path along with any far-end talker $x(n)$ signal, which is sent to the speaker system. The sound mask signal $SM(n)$, after it has been reflected in the acoustic environment, is picked up by the microphone system $d(n)$ and re-input into the sound mask processor. An echo path $h(n)$ measurement for each speaker and microphone combination can be completed. This is important, because in a distributed microphone and speaker system, each microphone requires its own echo canceller function. The acoustic return echo path for each microphone and speaker combination is unique and cannot be treated as a summed microphone signal for the purposes of acoustic return echo cancellation. The sound mask processor outputs a correction signal $RTF(n)$ to the output path for each microphone, canceling out the return echo signal present in the microphone channel.

Figure 5 illustrates how the sound mask processor meets the challenge. Because the sound mask processor does not require a far-end talker, the conference system is able to adapt and optimize echo canceller parameters when other systems cannot. For example, during the idle state, which may be an empty or silent room, the sound mask processor can measure the acoustic environment and determine return echo path values to calibrate

the echo canceller. This ensures the system will be ready to perform the instant it is required for a conference call.

Always Ready to Perform

While other systems are sitting idle doing nothing, Microphone Mist technology is preparing for your next call. To top it off, a manual calibration process is no longer required. For example, as people enter the room and/or arrange the furniture and supporting equipment, the sound mask processor can measure this change and adapt the echo canceller parameters in real time even if there is no one talking in the meeting. Remember, adaptive echo cancellers require a speaking far-end talker to calibrate and adapt. The sound mask processor is able to adapt in real time, so the participants at the far end (remote sound) are able to stay engaged and focused on the conversations in the conference room without having to deal with their own or other remote participants’ acoustic return echo sounds during the call.

The implementation is future-proof and scales with an increase in system complexity. The costs associated with having an acoustic or IT person to maintain the system calibration are eliminated.

Part of a Solution

The continuous echo canceller calibration functionality builds on Nureva’s patented Microphone Mist technology, which places thousands of virtual microphones throughout a space to pick up sound from any location to ensure that everyone is clearly heard

regardless of where they are in the room or the direction they are facing. Users also have the option to limit audio pickup to a specific zone within a space, such as the front of the room, making it ideal for corporate presentation scenarios and lecture capture in higher education.

The technology uses sophisticated algorithms to simultaneously process sound from all virtual microphones to provide remote participants with a high-quality listening experience, enabled by continuous auto calibration, simultaneous echo cancellation, position-based automatic gain control and sound masking. Microphone Mist technology adjusts the sound source level to work with Microsoft Teams, Skype for Business, Zoom, Cisco Spark, Cisco WebEx, GoToMeeting, Pexip Infinity Connect, and other common UC&C applications. **ax**

Resources

Nureva, www.nureva.com

“Nureva Granted Three Patents for Its Microphone Mist Technology Platform,” *audioXpress*, <https://audioxpress.com/news/nureva-granted-three-patents-for-its-microphone-mist-technology-platform>

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More on Restaurant Noise

By
Richard Honeycutt
(United States)

Noise in dining spaces was the topic of the July 2017 Sound Control article, which concentrated on loud “background” music and speech. In this article, we will focus on the noise from a restaurant’s HVAC system, which made conversation all but impossible.

Prior to the coronavirus (COVID-19) pandemic, I decided to try a new restaurant in my hometown. While the food was good, the prices reasonable, and the concept unique in my experience (a Southwestern salad buffet), my wife and I almost could not communicate because of the noise. In this restaurant, the noise from the HVAC system made conversation all but impossible. This problem is not unique to restaurants, occurring in auditoriums, churches, and other venues as well.

Measuring Noise

In architectural acoustics, noise is often measured in dBA, which is a frequency-weighted measurement that takes into account the frequency response of the human hearing system. However, noise specifications are often given in Noise Criteria (NC) or Room Criteria (RC) values. The NC specification was introduced by Leo Beranek in the late 1950s. The method essentially compares noise spectra to the equal loudness contours. Although this is similar to what A-weighting does, the A-weighting curve only correlates with the 40-dB equal-loudness contour. **Figure 1** shows the curves that define the specific NC values.

However, NC specifications suffered from two shortcomings:

- The NC ratings did not correspond well to the greater low-frequency noise levels produced by energy-efficient HVAC systems that began being produced in the early 1980’s
- NC ratings did not reflect the greater annoyance resulting from rumbly or hissy spectra (excessive low- and high-frequency noise levels, respectively).

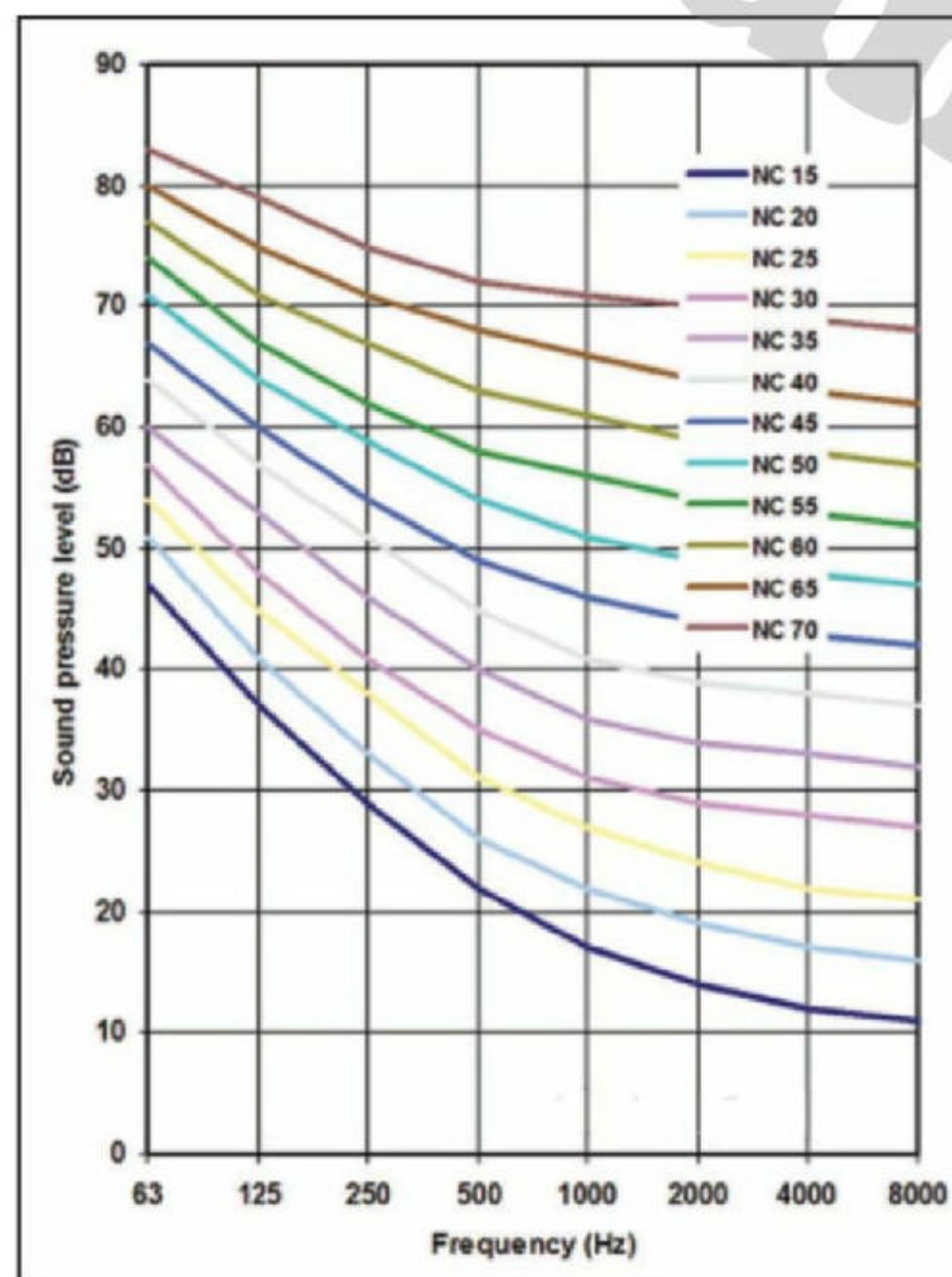


Figure 1: The NC curves are one standard way of specifying noise levels.

The NC method was updated in American National Standards Institute (ANSI) Standard S12.2-2008. The new standard extends NC curves down to 16 Hz and includes checks for spectral imbalance to identify rumbly or hissy spectra that could be highly annoying. **Figure 2** shows the new NC curves.

The RC system was developed in the late 1970s and early 1980s by Warren Blazier for ASHRAE as a response to low-frequency and high-frequency problems with NC and includes the effects of spectral balance. The spectral balance component is based upon a subjective decision, leading some prominent consultants and architects to prefer the NC system. **Figure 3** shows the RC curves.

RC Mark II was introduced in the ANSI/ASA Standard ANSI/ASA S12.2-2008. As seen in **Figure 4**, the RC II curves are similar to RC curves in that they exhibit a -5 dB/octave slope from 32 Hz to 4000 Hz. However, the curves are flat between 16 Hz and 32 Hz. Recent research by Erica Ryherd and Kerstin Waye (see Resources) has shown that simple A-weighted levels correlate with worker perception and performance as well as any other room criteria.

Problematic HVAC Noise

My first exposure to problematic HVAC noise levels occurred in 1985. I was reviewing the acoustics of a church sanctuary built in 1954, and measured the HVAC noise as part of my evaluation. The HVAC noise level was about 55 dBA. Recommended noise levels for churches are about RC 30-35 or NC 30-35, corresponding to an equivalent 40 dBA to 45 dBA. HVAC noise is affected by several things:

1. The noisiness of the air-handling units (AHUs)—fans, heat pump elements, and air-conditioning chillers
2. The design and length of the ducts between the above devices and the air diffusers
3. The design of the air diffusers
4. The velocity of the air passing through the diffusers.

Item 1 is controlled by the selection of the HVAC devices. Item 2 depends upon the design of the system for distributing the heated or chilled air, as well as the duct shape (round, oval, or rectangular) and size. High frequencies are absorbed as the air passes through the ducting, and corners add additional high-frequency attenuation.

Large ducts and plenum chambers can allow low-frequency noise to vibrate their walls, which

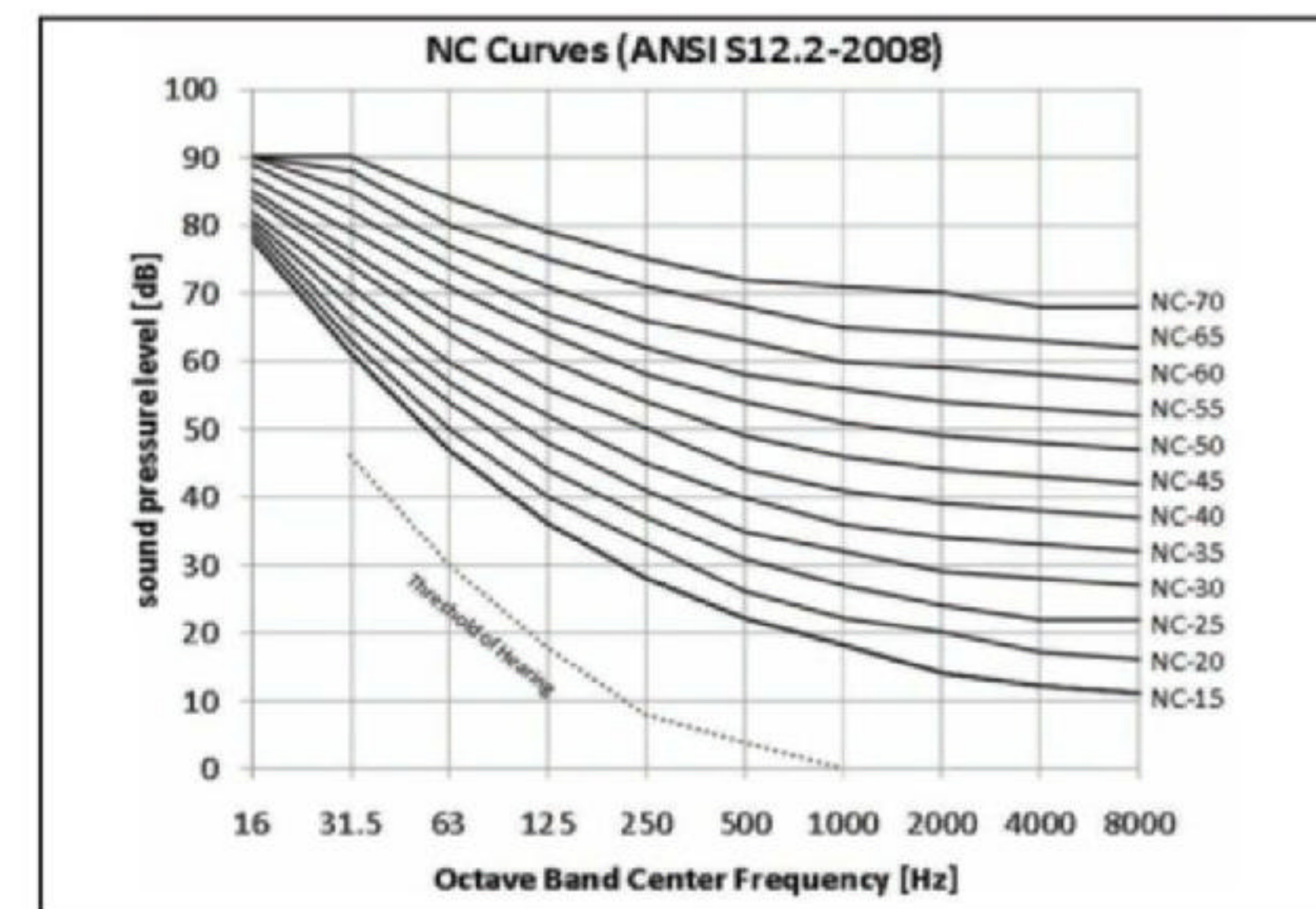


Figure 2: The new NC curves better match perception.

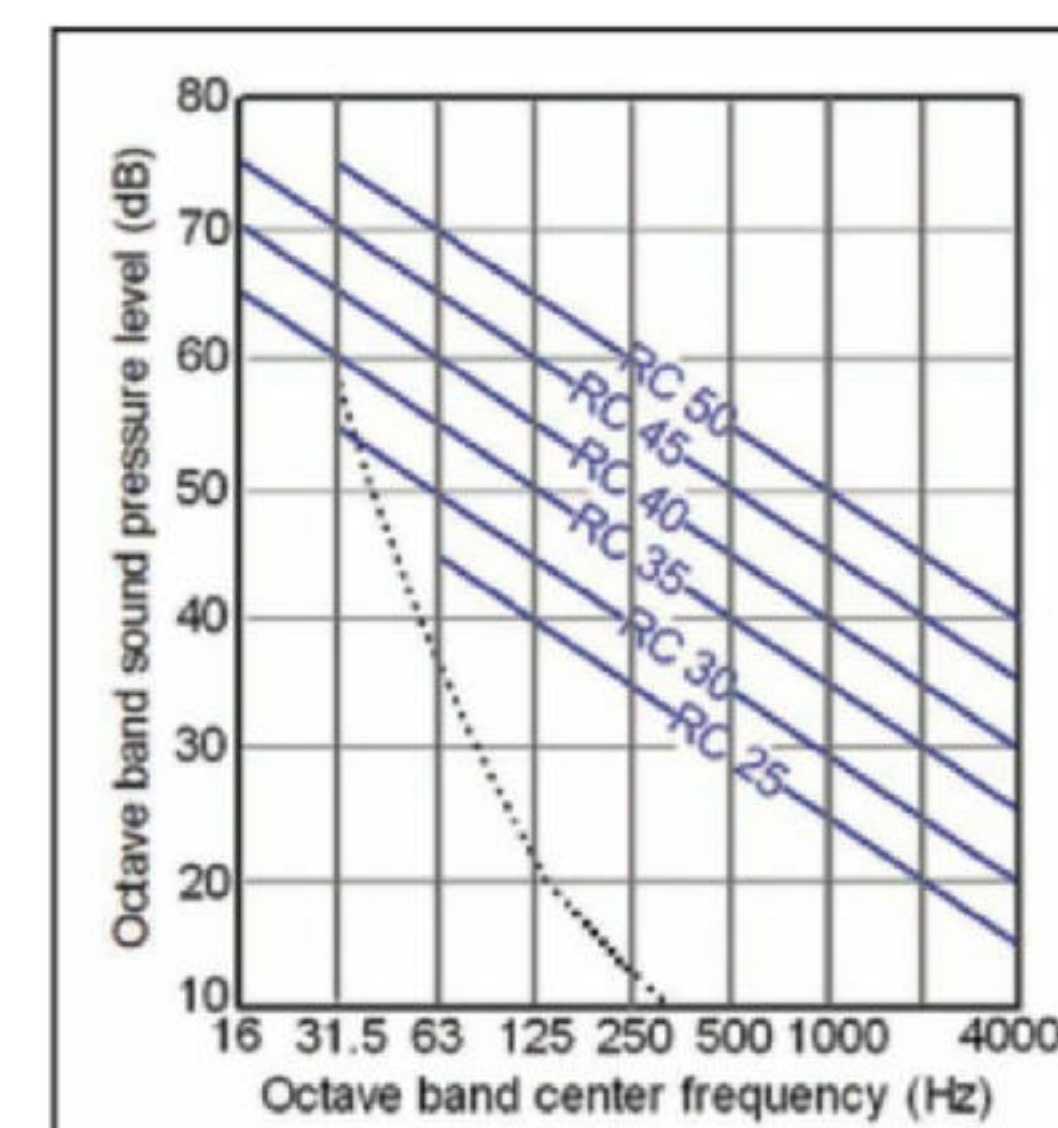


Figure 3: RC curves were developed to better account for the frequency balance of noise. The dotted line represents the threshold of hearing.

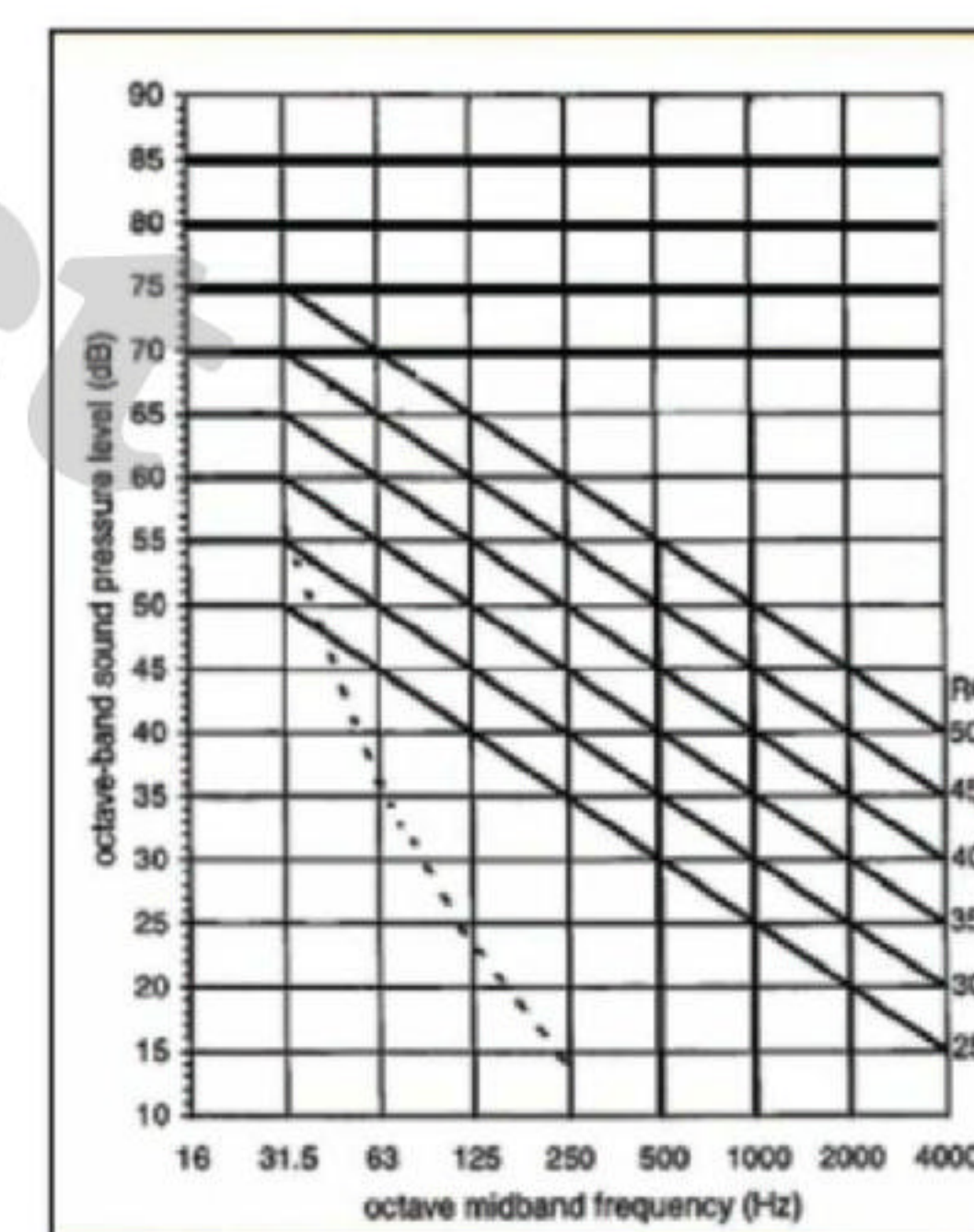


Figure 4: The RC Mark II curves are flat from 16 Hz to 61.5 Hz. The dotted line represents the threshold of hearing.

then re-radiate noise into occupied spaces. Several types of duct liners are available to increase noise attenuation in the ducts. Some ducts are flexible, and these can provide greater noise attenuation, although deflecting the direction of the airflow increases noise generation.

There is a wide variety of designs for “registers,” “vents,” and diffusers through which conditioned air flows from a duct into the room (see **Photo 1**). In general, the more restrictions the diffuser introduces into the air path, the more noise it creates. Dampers used to regulate air flow increase noisiness.

In designing an HVAC system, the engineer must determine the number of complete air exchanges per hour that must be provided in order to properly control the temperature. This leads directly to a specification of the volume flow (cubic feet/minute or CFM) that the HVAC system must deliver. If you divide the needed CFM by the total area of the diffusers, you will find the necessary air velocity in feet/minute for each diffuser.

Clearly, smaller diffusers will require a higher air velocity, which means more diffuser noise. (Compare this to whistling: When you blow air through pursed lips, you make noise. If the internal volume of your oral cavity is within a certain range, the resulting Helmholtz resonator gives the noise a pitch. But even without making a pitch, you will notice more noise if you blow harder, increasing the air velocity.) In older HVAC designs, the desire to minimize wall thickness led to the use of smaller ducts; and hence, smaller

diffusers requiring high air volumes. In the 1954-vintage church sanctuary, it was high air velocity that caused the HVAC noise to be so high.

It’s worth mentioning that the human tendency to ignore constant quasi-random sounds leads most people not to notice background noise—at least, not directly. In the sanctuary under discussion, nobody ever complained about the noise; they said the pastor was not talking loudly enough, or that he was not enunciating clearly, and “they couldn’t hear him.” In fact, they could hear him, but could not understand him because of the detrimental effects of high background noise upon speech intelligibility.

Another church experience illustrates this effect. A prominent acoustical consultant had created the sanctuary acoustical design for a major renovation in 2015, but when he asked about reviewing the HVAC design, he was told this would not be necessary. After the renovation was complete and the new sanctuary was commissioned, he received a call from the pastor informing him that congregation members could not hear him preaching.

The consultant visited the church. He, the pastor, and another member of the church staff went to the sanctuary and turned on the sound system. The pastor began reading into the pulpit microphone. It being a cool day in late fall, the HVAC system was on. The staff member stated that she could not understand the pastor’s words from most seats in the sanctuary. Then the consultant quietly walked over to the wall and turned the HVAC system off. Suddenly, to

the surprise of everyone but the consultant, the pastor’s words were clearly audible and easily understood!

At the time that I visited the 1954-vintage church sanctuary in 1985, nobody ever noticed the high noise levels, except hearing-aid users. Particularly during that time (but even, to a lesser extent, today), hearing-aid users are particularly sensitive to noisy environments. But as with the congregation in the second church, they do not perceive that there is a noise problem; they just think they can’t hear. Interestingly, the noise of the pipe-organ blower in the 1954-vintage church was louder than the HVAC noise, but nobody ever commented about that.

Back to the Restaurant

So where does all this tie in to my restaurant experience? First of all, I saw no evidence that the staff noticed the noise. Normal speech has a sound level of about 60 dB at one meter from the talker; raised-voice speech, 65 dB; loud speech, 72 dB; very loud speech, 78 dB; a shout, 84 dB; and the loudest possible shout (for most people), 90 dB. Humans can understand speech in the presence of noise levels a couple of decibels above the speech level.

But if the noise is more than a decibel louder than the speech, we unconsciously raise our voices to compensate. If we are concentrating on our work, we may not notice the greater vocal effort, although a full day of speaking at or above raised-voice levels can lead to vocal-abuse symptoms. When we are conversing over a meal, having to raise our voices is more likely to be unpleasant to us. Thus, restaurant patrons are more likely to recognize the effects of high noise than are the employees.

HVAC noise is less distracting than sound containing intelligence (e.g., speech or music), and thus, less critical to patron response to the restaurant. All this leads me to the conclusion that the HVAC noise level in this restaurant was about 70 dBA to 75 dBA.



Photo 1: Diffusers (also called vents or registers) affect HVAC noise level.



Photo 2: Air supply diffusers mounted directly to a duct are noisy.



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Photo 3: Flexible ducts between the main duct and the diffuser can reduce noise.



What caused the HVAC noise to be so high? **Photo 2** shows the air-supply system for the restaurant. Notice that the diffusers are mounted directly to the duct. Not only is the air deflected by a significant angle when it moves from the duct through the diffuser, but there is no flexible duct between the main duct and the diffuser. As mentioned earlier, a flexible duct (see **Photo 3**)

would have attenuated the noise. I consulted on a similar problem in a council chamber some years ago, in which the heat supply diffusers for the entire room were mounted directly to the main duct above the stage, creating a noise level that pretty effectively masked speech. In both the restaurant and the council chamber, relatively small diffusers were used, resulting in the need for high air velocity.

Photo 4 shows the other source of excess HVAC noise in the restaurant: the return air duct. Most people assume that most HVAC noise comes from where the conditioned air enters the room—the diffusers—and forget the effect of the air return vent. In fact, the noise power at the return vent is nearly the same as at the supply vent. If the AHU's return duct communicates directly with the occupied space, it can contribute significantly to the noise level.

The Solution

The solution to this is to design the system so that there is a significant length of duct (preferably lined) between the return grille and the active



Photo 4: Return vents feeding directly to the AHU are noisy.

HVAC system components. Separate duct silencers are available that can reduce the length of duct needed for adequate sound attenuation. In this restaurant, the return grille was mounted directly to the AHU, so there was no attenuation between the AHU return and the room.

In an online article, Julia Belluz writes, "Both *Zagat* and *Consumer Reports* surveys have found that excessive noise is the top complaint diners have, ahead of service, crowds, or even food issues. Tom Sietsema, the restaurant critic for the *Washington Post*, also told me noise is 'by far' his chief complaint about the restaurants he reviews."

Some restaurant patrons have taken to measuring sound levels in restaurants, using a variety of apps. Belluz comments, "Decibel-reading apps have also proliferated. Some, like SoundPrint, crowdsource users' decibel readings to rate venues so that people who are

hard of hearing or sensitive to noise can find quiet spots. (In DC, for example, I found some popular restaurants with sound levels below 70 decibels, according to the crowdsourcing, and others that were 'very loud,' over 80 decibels.) The NIOSH Sound Level Meter also allows you to measure sound in restaurants or your workplace."

As I pointed out in my November 2015 Sound Control article, "Smart Apps for Sound Measurement," there is a wide range of accuracy among the various apps. Yet the causes of excessive noise are well-known: loud music, too many people talking in an acoustically hard space, and excessive HVAC noise. All these are subject to correction, given the inclination on the part of restaurateurs to correct them. As patrons increasingly demand more acoustically friendly eating places, the owners will eventually respond. 

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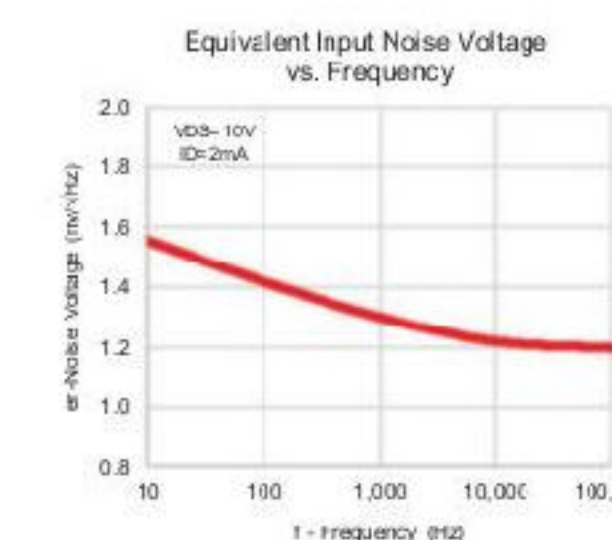
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Hidden Low-Frequency Noise Contributors Applying FDAs to Audio Digitizer Channels

Along with considerable convenience and performance advantages using the newer Fully Differential Amplifiers (FDAs) to interface from measurement mics to 24-bit audio ADCs comes some added risk of impaired low frequency spot noise with those solutions. Where that comes from and paths to mitigation will be reviewed here.

By
Michael Steffes
(United States)

Thus far, the classical input noise models have delivered an accurate output differential spot noise prediction including the 1/f effects for the differential input noise and bias current terms. Not considered yet is what impact (if any) the common mode (CM) loop amplifier will have on the output differential noise. FDAs have migrated to providing an internally developed mid-supply bias as a default input for a floated input control pin (see Section 8.2, Reference 1). This is intended to save components when the FDA can operate with an approximate mid-supply common mode output voltage for single-supply operation.

To keep the total power budget low, these divider resistors are often very high valued (300 kΩ each in the THS4551). That control pin then feeds into one side of a servo loop amplifier that also operates at

low bias current to save power. This external V_{OCM} input path is usually relatively low bandwidth (see Figure 43, Reference 1) while once past that first stage, the rest of the CM servo amplifier operates at speeds comparable to the differential loop.

The immediate difficulty to surmount is that it does not appear any of the extant simulation models correctly produce the measured (and sometimes reported) floated input CM noise over frequency. Also, CM noise reporting as shown in **Table 1** is very spotty and possibly suspect in several of the FDAs suited to these applications.

The floated control input noise for the THS4551 shows an approximate 1/f corner at 1 MHz increasing its flatband value from 15 nV/√Hz to 320 nV/√Hz at its minimum 100 Hz plot (see Figure 45, Reference 1). This likely arises from the input bias current noise 1/f effect for the buffer transistor gained by the nominal 150 kΩ source impedance. It only requires 2.1 pA current noise at 100 Hz to produce that 320 nV/√Hz value at 100 Hz through that 150 kΩ source. Extending these 100 Hz numbers down to 1 Hz to cover the measurement mic span will increase these by 100 times, if they follow the 1/f model.

Figure 1 shows a balanced feedback divider ratio simulation with the CM noise sensed through two 100 Ω resistors to a common point at the output. This same sensing approach would feed out through a 50 Ω cable to a 50 Ω terminated noise preamp for CM noise measurement purposes. Since the 2016 THS4551 TINA model does not produce the correct output CM noise, a noise macro was added

Wideband FDAs for Audio ADC Driving— 100 Hz Floated Control V_{OCM} Spot Noise				
Part Number	Supply	SSBW	V_{OCM} Noise	Notes
	Current	$A_v=1$	100 Hz, Floated Control Input	
THS4531A	0.25 mA	36 MHz	No spec	
THS4561	0.78 mA	60 MHz	3,000 nV/√Hz	
MCP6D11	1.4 mA	82 MHz	1,000 nV/√Hz	
ADA4945	4 mA	145 MHz	no plot	35 nV at 100 kHz
THS4551	1.37 mA	150 MHz	320 nV/√Hz	
ADA4940	1.25 mA	260 MHz	no plot	92 nV at 100 kHz

Table 1: This is the output CM noise specification for FDAs suitable to audio in ascending bandwidth.

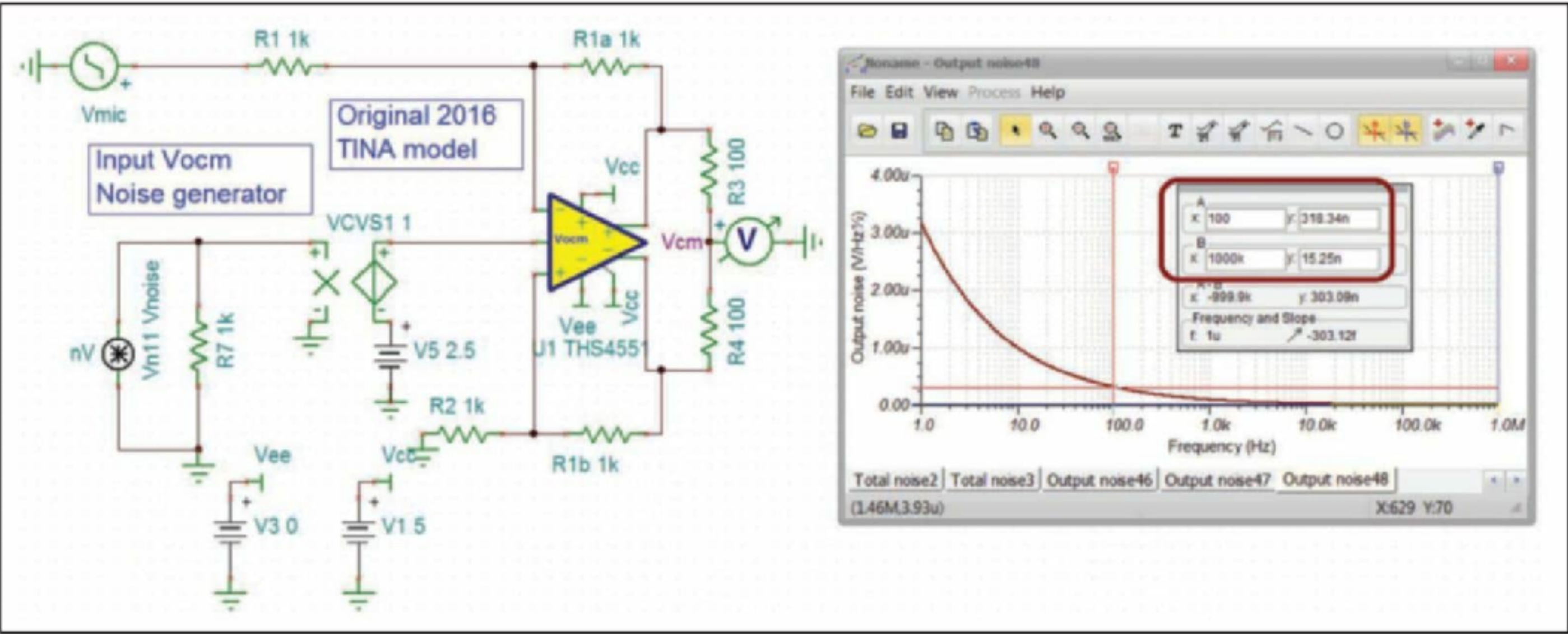


Figure 1: This output CM noise simulation is shown with an emulation source modeling measured floated V_{OCM} noise.

to emulate the floated input measured output CM noise (see Figure 45, Reference 1).

The 100 Hz and 1 MHz markers approximately match the measured plot. What is concerning is that $>2 \mu\text{V}/\sqrt{\text{Hz}}$ at the 2 Hz low end of the measurement mic span.

FDAs can apparently produce a very high CM output noise at lower frequencies—but why care? The ADCs have extremely high common-mode rejection ratio (CMRR) and that CM noise stays CM if the feedback ratios are perfectly balanced. However,

if the feedback ratios are even slightly unbalanced, that high CM noise below 10 Hz will generate a meaningful differential component that adds to the FDA's intrinsic differential output noise. **Figure 2** shows a CM to differential mode (DM) conversion simulation driving an AC signal into the V_{OCM} control input with worst case feedback ratio imbalance using 1% tolerance R values.

This -34 dB gives a 0.02 gain using $\pm 1\%$ resistor tolerances—or essentially two times the tolerance of 0.01. Applying that gain to the $2.2 \mu\text{V}/\sqrt{\text{Hz}}$ CM noise at

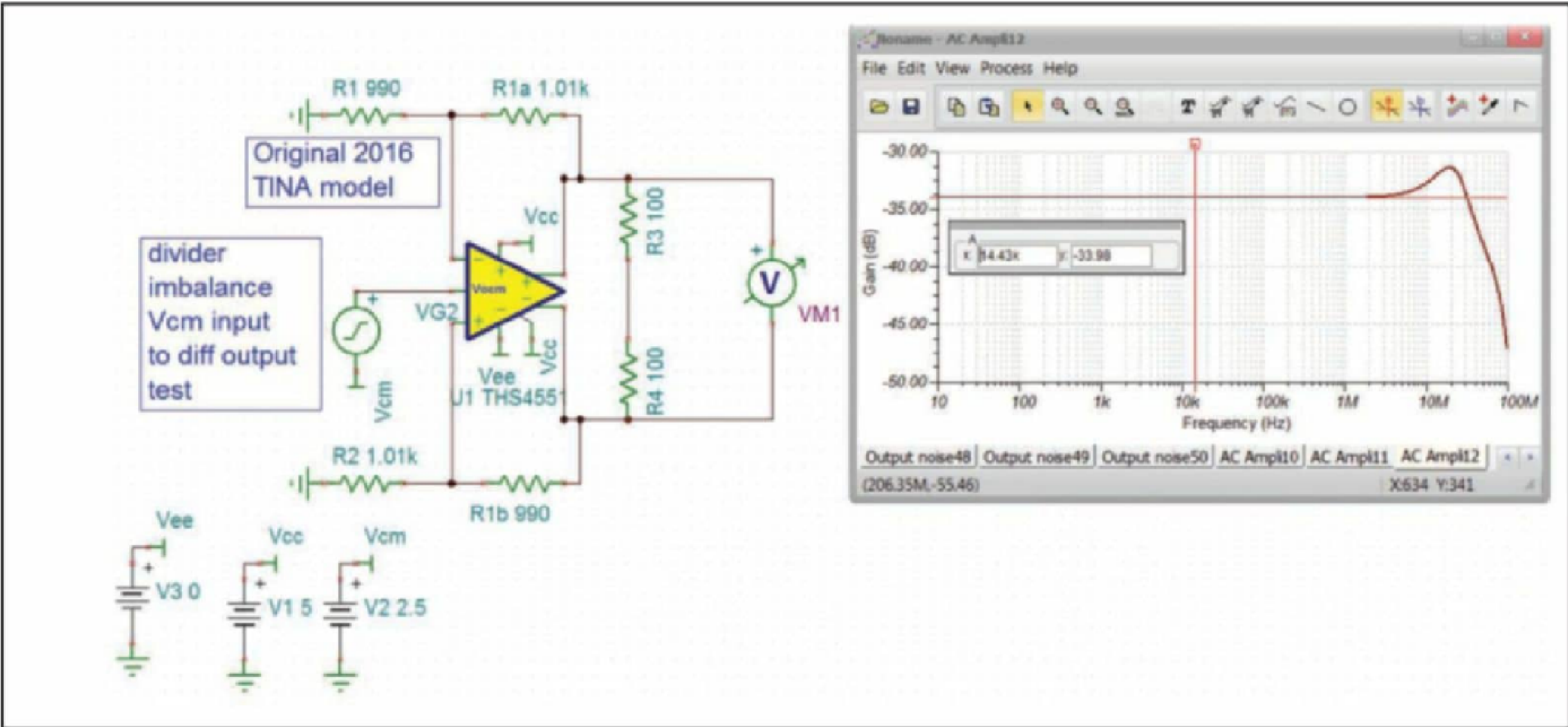


Figure 2: The conversion gain from CM noise to DM noise is depicted with ratio imbalance.

2 Hz shown in Figure 1 adds a 44 nV/vHz differential noise term to the FDA's intrinsic differential output noise at 2 Hz (an RMS spot noise addition). **Figure 3** shows both a matched resistor divider differential output noise and with 1% mismatched resistors using the CM noise model to capture this effect. This is clearly showing the perhaps unexpected degradation in differential output noise through this new mechanism.

Mitigating This Low-Frequency Noise Degradation

The work thus far with the THS4551 is likely showing the lowest CM noise device in Table 1. Much

higher noise (THS4561), or unknown, low-frequency CM noise appears across the range of FDAs. Of course, the floated input CM noise is the highest due the very high supply splitting resistors used to set a default mid-supply output CM operating point for single-supply operation. Since some of these devices might have very high (but unknown) low-frequency CM noise, a range of ascending effort mitigation approaches are suggested:

- If the default midscale V_{OCM} value is acceptable, at minimum place a capacitor to ground on the V_{OCM} control pin to filter those internal resistor noise terms.

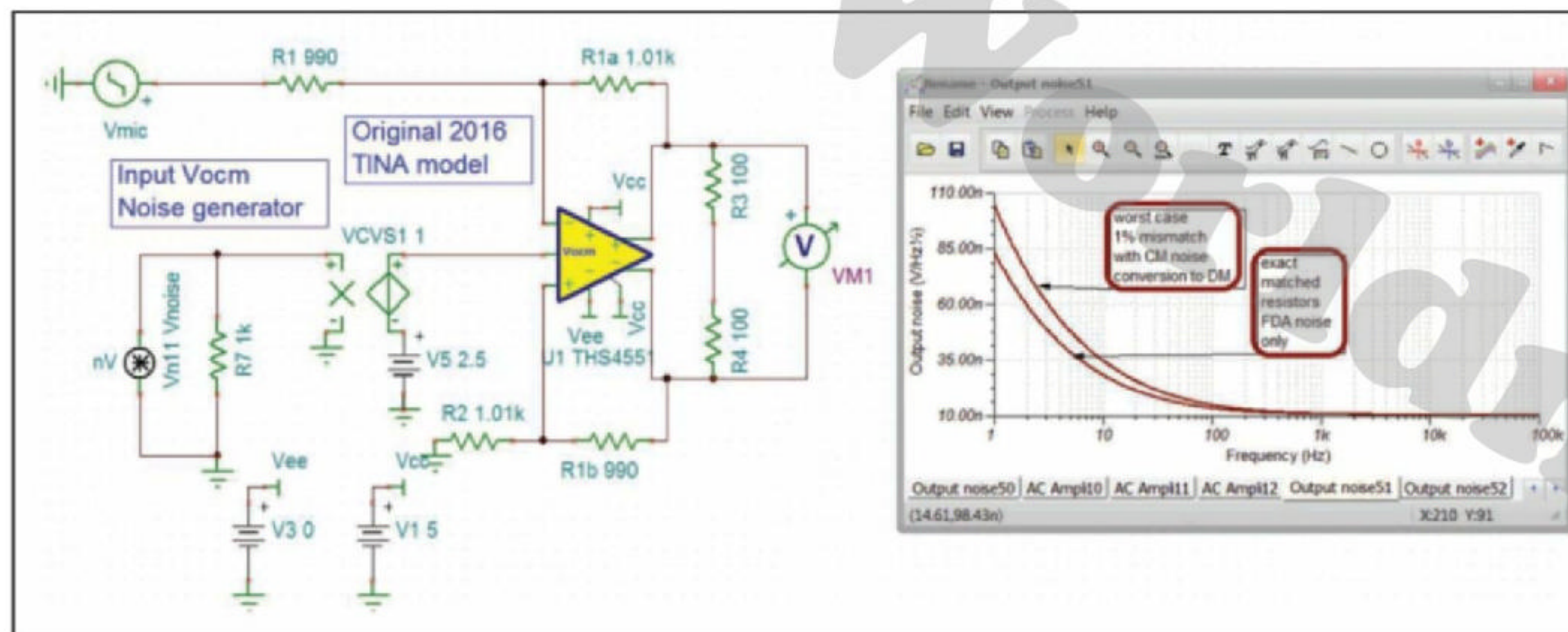


Figure 3: This graph shows the effect of CM noise converted to differential noise by resistor ratio imbalance.

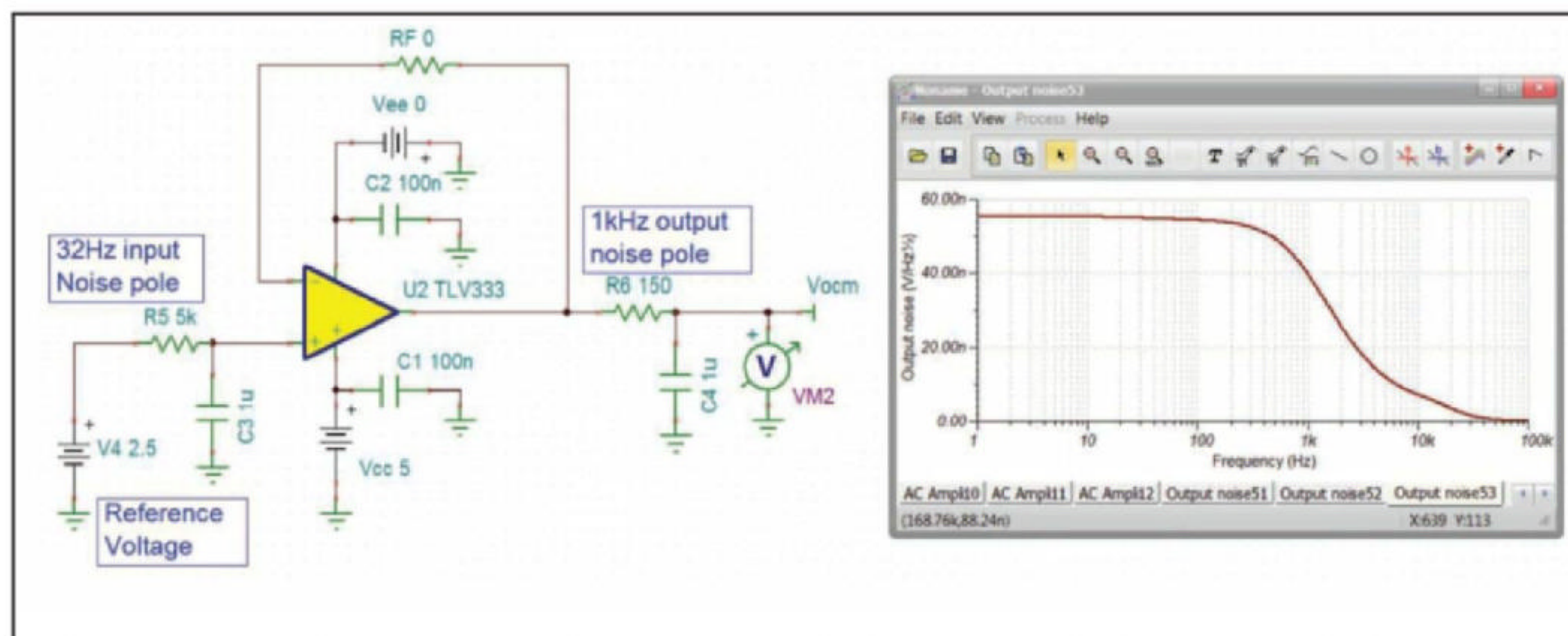


Figure 4: This is the output noise spectrum for a chopper amplifier buffer to V_{OCM} design.

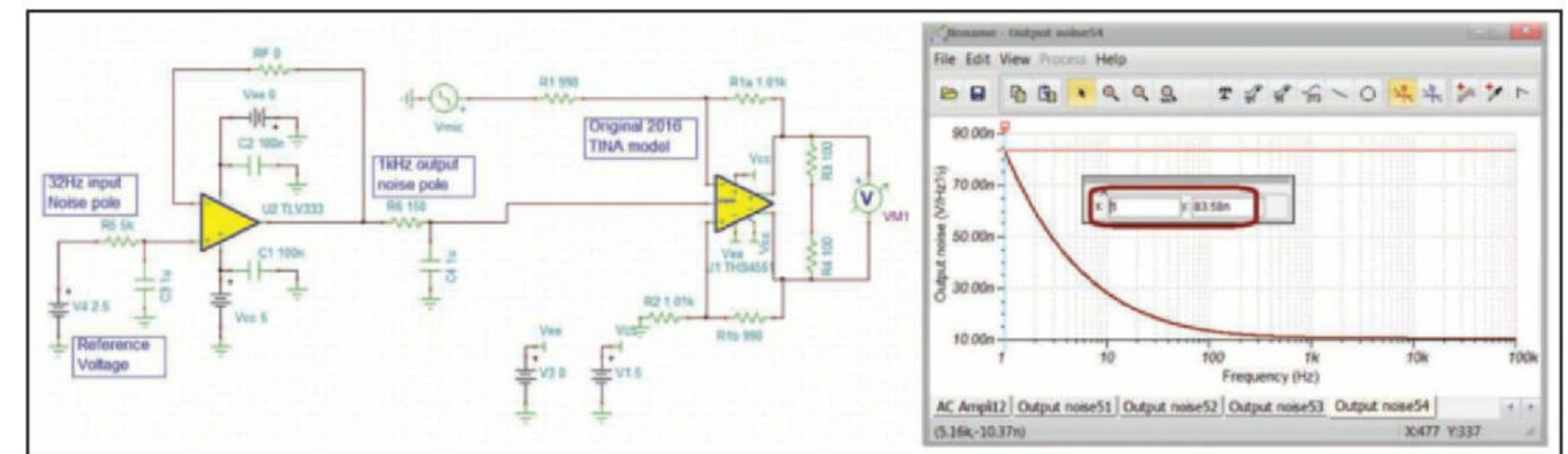


Figure 5: Chopper amplifier buffer circuit is shown driving the V_{OCM} control with unbalanced FDA resistors.

- If you are going to use the ADC reference output, since they never show their noise spectrum, you should at minimum add an RC filter to the FDA input V_{OCM} control pin and provide that cap to ground on the V_{OCM} pin.
- Consider using tighter tolerance resistor values. It seems $\pm 0.5\%$ R's in E96 steps are a small price increment over $\pm 1\%$.
- If the cost of a buffer amplifier is acceptable, or a controlled V_{OCM} voltage is desired, use a low cost op-amp with a filtered output.

If you are coming from a measurement mic source, attempt to balance the impedance looking back into that element on the other side of the FDA circuit. Some measurement mics have two outputs (see Reference 2) and show a balanced 20 Ω source impedance on each side. Those phantom powered mics will show a DC level on each side (with the audio signal one side) that will need to be blocked off with input capacitors. Feeding that in differentially will keep the feedback dividers balanced but adjust the input R for that 20 Ω to hit the desired filter shape if implementing an active filter.

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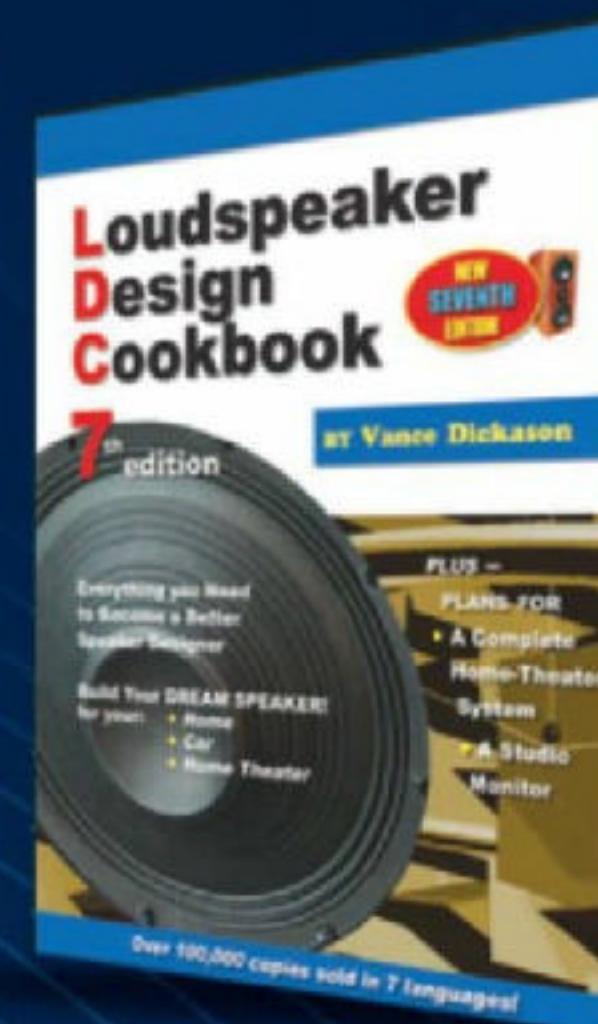
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Some of the FDAs listed in Table 1 have shown measured popcorn noise in the differential path. Those same mechanisms may be in the common mode path as well—or in the external buffer amplifiers, if poorly chosen. The need here is for a popcorn free buffer amp with low (or no) 1/f corners—but not necessarily the lowest flat noise as the CM to DM conversion gain is relatively low.

One solution appears in **Figure 4**, using the low-cost TLV333 (see Reference 3) chopper amplifier. Chopper amplifiers do not necessarily have low noise, but they do have no 1/f or popcorn noise effects. This circuit filters the unknown reference source noise then adds an RC pole at the output. This noise spectrum feeding into the V_{OCM} control input for the FDA should remove CM noise as a possible degradation to output differential noise using any of the FDAs listed in Table 1.

Placing this reference buffer circuit into the gain of one FDA design with $\pm 1\%$ unbalanced resistors shows the differential output noise is unaffected by this CM noise source. **Figure 5** shows the noise spectrum of Figure 4 fed into the V_{OCM} control with unbalanced FDA resistors producing the same 1 Hz differential spot noise as the balanced resistor

curve shown in Figure 3. Since there is no switching effect on these very high input impedance V_{OCM} control inputs, a single buffer amplifier can easily drive multiple FDA V_{OCM} control inputs if needed.

There is currently a gap in reporting and modeling the low frequency CM control loop noise. Not knowing how high this could get at the 2 Hz low end of the measurement mic frequency argues for one or several of the suggested external improvements.

Next up, estimating the 0.1 Hz to 10 Hz peak-to-peak noise with a typical noise preamp design for that, directly measuring that span, as well as popcorn noise. 

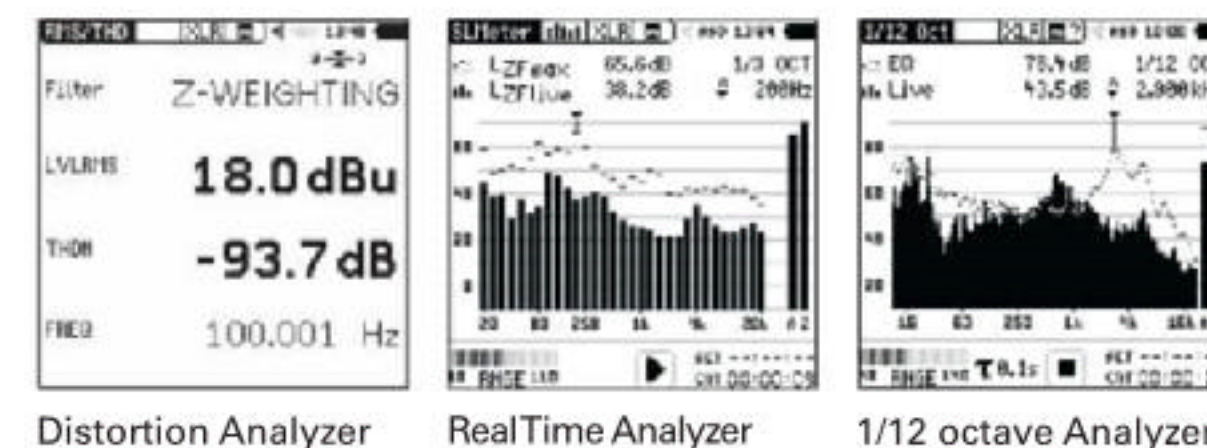
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Moving Audio Landscape

By
**J. Martins and
Mike Klasco**

In this Market Update, we discuss the automotive OEM landscape and relevant audio-related trends and opportunities, focusing on the technologies and solutions announced recently and some of the radical audio innovations reaching the 2021 car models.

As CES shows have increasingly become an automotive-focused industry event, with many of the key technology companies preferring the Consumer Technology Association (CTA) umbrella to the specific channel-driven massive auto shows, we notice that more and more of the topics discussed in the audio industry are now commonplace. Voice interfaces, immersive audio, sophisticated digital processing algorithms for noise cancelling, and zoning increasingly converge with the latest consumer industry standards and efforts (e.g., Wi-Fi, Bluetooth and USB) to deliver more and better in-car connectivity options.

As Strategy Analytics noted in its “In-Car Wired and Wireless Connectivity” report, some of the big developments in the space include the long-anticipated wireless implementations of Apple CarPlay and Android Auto finally going mainstream with the introduction of Ford’s Sync 4 system utilizing Wi-Fi, increased use of the versatile and powerful USB-C standard, and more cars using “phone-as-key” technology utilizing Bluetooth Low Energy (BLE) and near-field communication (NFC).

“The influence consumer electronics and mobile devices is having on the automotive industry is undeniable,” said Edward Sanchez, Senior Analyst, Automotive for Strategy Analytics. “Although automotive validation and testing for some of these technologies is significantly more rigorous than

for consumer electronics, customer demand is unrelenting. We’re seeing innovation happening in the in-vehicle connectivity space like never before.”

“OEMs and Tier 1s are significantly stepping up their game when it comes to connectivity,” adds Strategy Analytics. “The connectivity features and capabilities now available in the car were the stuff of dreams just a few years ago.”

Thanks to all those efforts and increasing systems integration of connected technologies in the car infotainment area, audio systems are also receiving considerable attention.

A great example comes from Harman International, one of the market leaders in the automotive technologies space, which at CES 2020 unveiled its unified driving experience with connected car, audio and service technologies.

Now a wholly-owned subsidiary (but independent business unit) of Samsung Electronics, Harman is pushing this level of integration with other car systems (e.g., ADAS and telematics) in order to promote lower power, reducing the required power draw from premium audio and entertainment solutions—a clear priority for electric vehicle (EV) owners. Next, at par with efficiency, is the ability to use multiple voice assistants, access to global media services, and upgradable software. Among the innovations is also the use of natural sound landscapes (Harman



calls it NatureScapes) inside the vehicle, automatic adjustments of music levels when conversations are taking place, and the use of HALOsonic Road Noise Cancellation (RNC) for improved in-cabin comfort. All this, is directly integrated and dependent of Harman-designed audio systems.

Sound Systems

Even before CES, we attended the Tokyo Motor Show (October 24 to November 4, 2019), and the headlines at this massive event were the use of phased arrays of small speakers as the new direction of auto-sound. The promise is that each individual seat can have separate program material, more effective active noise canceling, and for the vehicle driver, enhanced situational awareness.

Designing and manufacturing sound systems for the automotive industry is extremely rigorous and only the most formidable speaker companies fight for this business. Harman does more than \$4 billion in automotive audio (80% of its total revenues), while Bose, the other big player, does comparable numbers as Harman. But in auto-sound OEM, there are another half-dozen contenders, including ESTec in Korea (vendor to Hyundai, KIA, and Ford), Eastech (KIA and Hyundai), Sonavox (supplying Ford and Chrysler), SIPE/ASK now part of JVC/Kenwood (Geely, Volvo, and Fiat), Fatial, Alpine (Honda and Acura), Dynaudio, a division of Goertek (VW group, China), and Bowers & Wilkins (BMW, Maserati, and Volvo).

But it is one thing to sell car speakers, and another to sell the electronics, which have considerably evolved since the days of the “car radio” to infotainment centers and now to fully connected systems able to connect with the user’s devices and services.

Let’s step back to Motorola, which in 1930 had designed an inexpensive car radio that could be installed in most vehicles. The name “Motorola” was created by linking “motor” (for motorcar) with “ola” (from Victrola), a popular ending for many companies at the time. Initial sales were made to police departments. Delco Electronics Corp., the

automotive electronics design and manufacturing subsidiary of General Motors (GM) began producing the first dashboard-installed car radios in 1936.

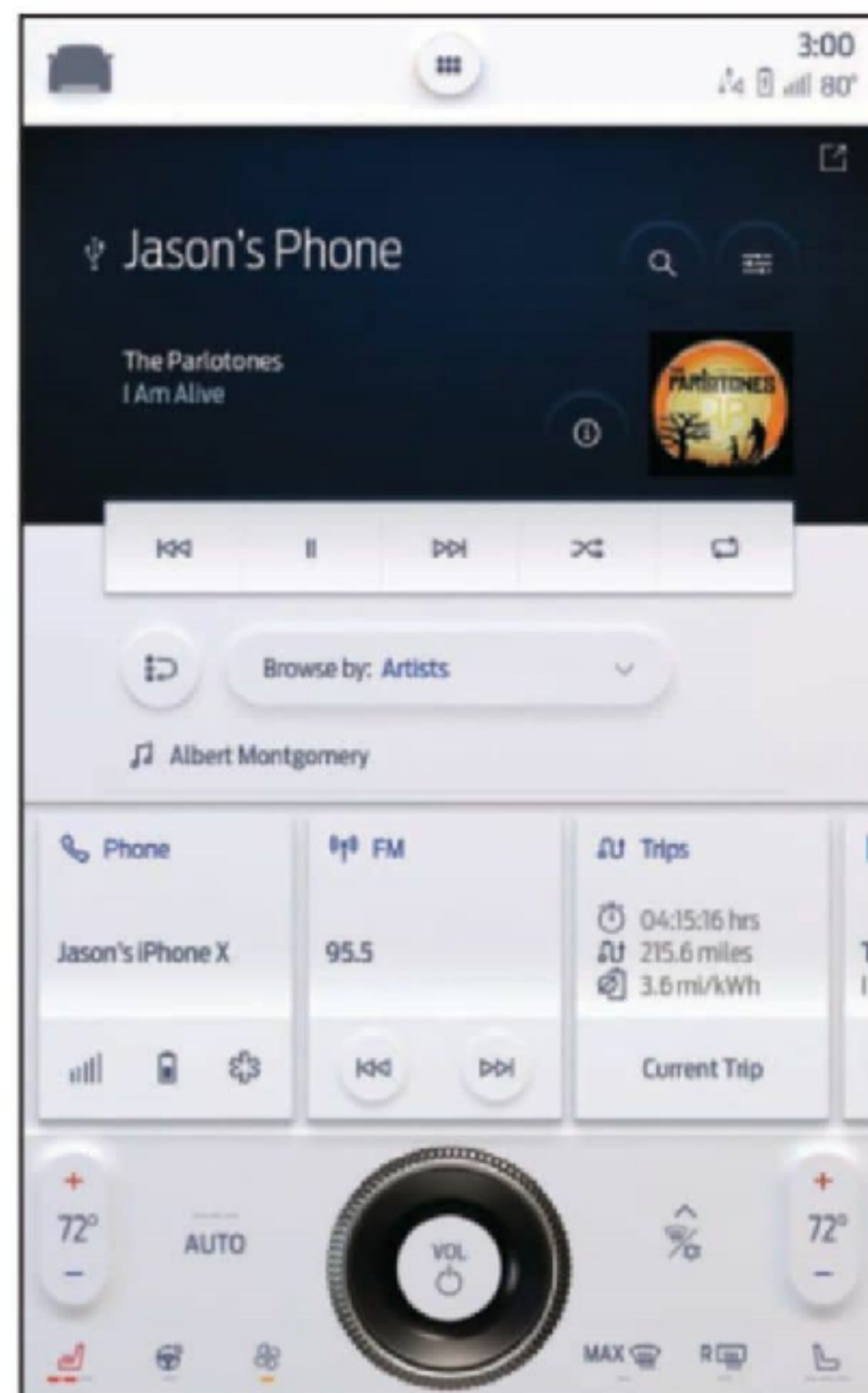
Back then, being able to just hear the radio above the vehicle noise was considered epic. AM radio vulnerability to interference was a given and eventually FM car radios were introduced by German brands Blaupunkt and Becker in the early 1950s. Philco developed the all-transistor car radio in 1955, which was offered as a factory option in Chrysler cars under the Mopar brand. In early 1956, Delco produced a transistorized/tube hybrid signal-seeking car radio, soon followed by an all-transistor design. By the 1960s, Philco was owned by Ford and was the in-house car radio vendor.

The 8-track cartridge became a Ford factory-installed option in 1965 with a unit supplied by Motorola; and toward the end of the 1960s, Philips introduced a car radio with a compact cassette—the



Good old auto-sound and aftermarket car radios are a tough sell in today’s market and traditional brands such as Pioneer are completely focusing on smartphone and connectivity integration.

Ford's next generation of SYNC 4 communications and entertainment systems help make navigation, music, and connecting the user's smartphone even faster and easier, adding cloud connectivity, voice recognition, and cord-free support for Apple CarPlay, Android Auto, and SYNC AppLink compatibility.



hiss was so loud that it masked the vehicle noise—an inadvertent early application of sound masking in automotive audio!

In the 1970s, auto-sound OEM dashboard systems were mere place holders for aftermarket gear and every audio shop had install bays where they pulled out the factory equipment and installed a real audio system. There was a comprehensive aftermarket replacement kit supply chain for adaptors for specific cars matched with specific upgrade head units. Finally, to make life simpler for installers, the DIN/ISO standard for dashboard audio systems was adopted



A classic car radio from the 1950s, when audio finally became part of the experience.

in 1984, right down to the I/O connectors.

A decade later, a misstep by Bose changed everything. Bose first auto-sound aftermarket system based on its 4" full-range speaker in the early 1980s was a dud. Back then, aftermarket audio offered some serious high-performance gear and the Bose product did not impress. Turning a lemon into lemonade, Bose's marketing found its way into GM and offered to create the world's first factory-installed premium sound systems for some upscale trim models. Unlike conventional or aftermarket automotive systems, Bose systems were designed and tuned for a specific vehicle—and this, together with the integrated infotainment system, changed the industry forever.

Pioneer, Kenwood, Proton, and the rest of the "12V" auto-sound aftermarket industry soon sped down the slippery slope. The option of the touchscreen infotainment central unit for climate control, security alarm, navigation, voice command and hands-free functions locked out and vanquished the aftermarket audio head unit category.

Vehicle manufacturers now hold the cards for co-marketing of the sound system and the rule today is design and supply by audio brands with installation by the vehicle assembly factory. Your interior accoutrements from the sound system to the seat materials and even the drivetrain and wheels are preordained by the trim level. The tradition of ordering a car with your options at the dealership was not viable as the Japanese were taking over and their supply chain lead time was too long for build-to-order, so locking the customer into trim levels and picking a car off the lot became the rule.

OEM/ODM and Design

Auto-sound OEM system design is in the process of making another step-change and again we have to give credit to Bose for leading the charge. Traditionally, the speaker selection and layout design has been configured by the design team for the entire passenger compartment as a whole cohesive space, but the trend for the near future is "personal audio arrays" designed for each individual seat. Bose has coined this approach as "SeatCentric."

The "ultra-nearfield" concept uses clusters of small full-range speakers configured and tuned as phased-arrays to direct sound to individual seats. If only one person is driving the vehicle, then the arrays are optimized for immersive audio for the driver. When kids are in the back, they can watch and listen to cartoons, while the adults in the front seats get the news or talk on the hands-free speakerphone. Bass can be somewhat contained by the use of haptic (bass shakers) in each seat, driven by the relevant program material.

In the 2021-onward models, we will see innovative mounting locations for new form-factor compact speakers in microarrays, including center dashboard, visor areas, A and B pillars (front windshield supports and a door pillar behind the driver position), and also in the front and back of the headrests. These next-generation autosound speakers will likely be in the 2" range (at least in one axis) both to fit and to function in a full-range array.

Aside from enabling "personal immersive audio" these micro-array configurations enable higher performance from active noise canceling (ANC) systems for effective drive train road and wind noise reduction up to 500 Hz or even higher (using the headrest speakers for ANC).

The micro-arrays and close coupling to the listeners enable practical ANC with far more viable results than using existing auto-sound design practices. Aside from smaller speakers, expect to see higher thermal power handling in these drivers than what you would expect. ANC requires continuous operation resulting in the need to re-scale for voice coil overheating and power compression. So larger voice coil diameters, ferrofluids (even in full-range drivers) and other design techniques will keep transducer engineers busy adopting techniques learned in pro sound as well as past solutions from auto-sound aftermarket designs.

Bose's vision for the future of auto-sound can already be seen in this year's Nissan Kicks and the Cadillac ST-6. The flagship showcase is the Cadillac CT6, which combines digital signal processing (DSP) with 34 speakers—4" diameter or less—strategically positioned throughout the cabin.

Kicks is Nissan's take on a Kia Soul-style "cool" subcompact crossover. The concept is to combine a decent (but cheap) car with an innovative sound system. First launched in the Nissan Kicks SR, the Bose Personal Plus is part of the \$1,000 Premium trim interior package. The system just dips its toe into personalized audio predominately for the driver's seat with two speakers integrated into the headrest. Headrest speakers have application to music/entertainment, ANC, and fidelity enhancement/bass enhancement, in combination with more conventionally positioned speakers.

Bose has used headrest speakers in the Infiniti convertible and Mazda's MX-5 Miata for years, but for the Kicks, the 2.5" Bose UltraNearfield speakers are fed by their PersonalSpace Virtual Audio processing. While not the full dose for passengers, there is widening of the sound stage for all the seats in the house.

Some of Bose's OEM autosound systems feature their PowerProfile woofer modules comprised of an array of 4" drivers facing into a manifold array. As there is always more room in one direction or another,

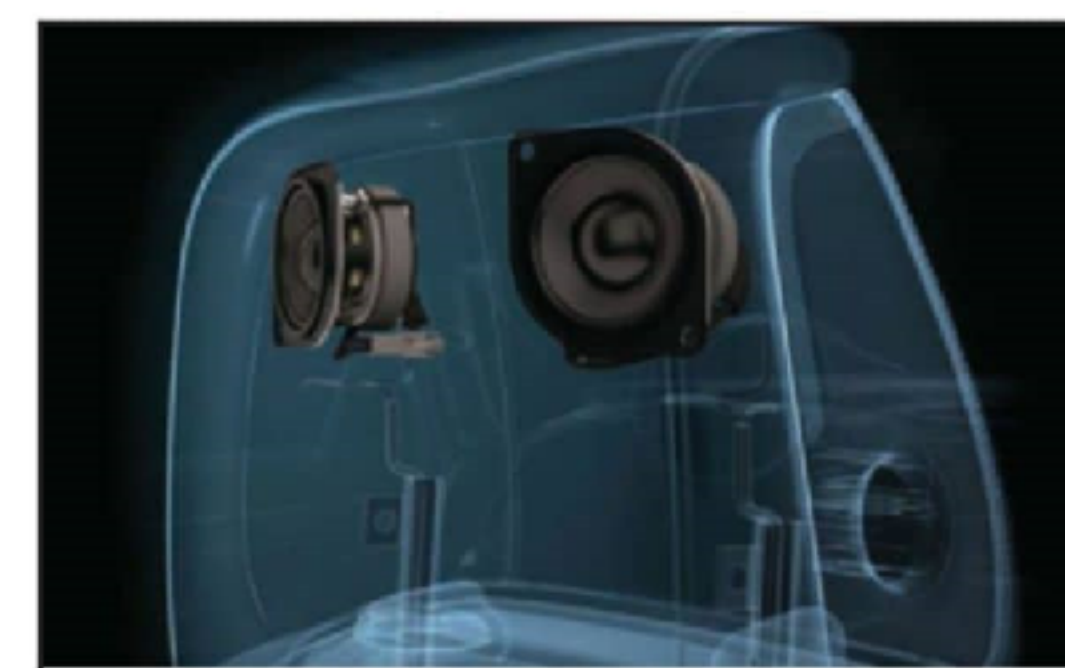


having this flexibility in the footprint can come in handy in fitting beneath the seating or on the floor.

The next generation of speaker micro-arrays for auto-sound will impact transducer performance designs. While smaller speakers positioned near-field to each listener will enable personalized audio for each seat, they can also provide localized ANC for reduction of road/tire and wind noise to 500 Hz or higher. Yet attempting this for extended driving hours will result in power compression to build up in typical mini-speakers.

One promising solution is the ThinDriver from Trulli due to its combination of high thermal power handling while maintaining small footprint size and shallow depth. The design repositions and expands the voice coil to the juncture of the flat diaphragm periphery and the surround so a 3" speaker boasts over a 2.5" voice coil.

Another interesting technical contribution to the seat localization concept was demonstrated by Noveto Systems, the company that took CES 2018 by storm when it demonstrated its focused audio beaming system for conferencing and entertainment zoning applications. The company returned in 2020 to release the first part of its chipset powering Noveto Smart Audio, a focused sound solution to precisely generate sound to the users' ears. Augmenting Noveto's acoustic beamforming technology to create a personal soundfield, the technology allows manufacturers to locate the position of a user's head in space and beam acoustic energy to create tiny sound bubbles next to the ears. When users move their heads, the sound bubbles will follow the movements. The chipset is scheduled to enter mass production by 2021 and is



Bose SeatCentric approach lets passengers enjoy their own music while the driver discretely handles a phone call through ultra-nearfield headrest speakers. It also allows for a call to be transferred from the driver to the passenger through the speakers. Rear Seat Volume Attenuation allows rear-seat passengers to enjoy a movie or music via a brought-in device, without distraction from the front seats.

able to handle numerous audio channels to support surround and 3D sound configurations.

The DSP Effect

By bringing speakers closer to the listeners, there is a significant weight savings, and anyone involved in the automotive industry knows that all vehicle designs are subject to strict weight limits. Compared to current state-of-the-art automotive systems, micro-array and headrest systems offer a weight savings of more than 50%, while still delivering an immersive audiophile-quality listening experience. This is achieved using small drivers in lightweight enclosures, lightweight and streamlined wiring harnesses (often with

aluminum conductors instead of copper) with multi-pin termination, multichannel low-power Class D amplifiers, and DSP-powered beamforming to deliver acoustic pressure to precisely defined sweet spots. Adding smart haptics leveraging DSP and tactile perception, significant further gains can be realized by lowering subwoofer amp and speaker weight.

A new take on signal processing was introduced by Dysonics, a behind-the-scenes spatial location processing specialist in ultra-nearfield auto-sound, that worked closely with Faurecia, a Tier 1 interior and seat vendor to Nissan, Renault, and many other automobile companies and the parent company of Clarion and Parrot.

Dysonics' DSP-based algorithms render multichannel content (surround sound or object-mixed) with precision to a single listener or to multiple listeners simultaneously. This is a result of cross-talk control afforded by transaural beamforming. Head modeling detail and sound imaging projected to multiple listeners exceeds legacy high-end systems. Each listener may adjust EQ, volume, bass/haptic response, and in the near future, aperture and focal distance. Rear imaging and fidelity enhancement can be augmented with headrest speaker support. Using sensors to report real-time situational updates, control and personalization are further enhanced.

A unique aspect of micro arrays in sound field control is the ability of a system to shift rendering modalities. For example, when all occupants of a vehicle are listening to the same material, multiple small arrays can synchronize to deliver extremely uniform coverage to the entire interior. The same arrays can be instantly reset to provide binaural coverage to a single listener, highly intelligible mono voice rendering simultaneously to another, or spatially accurate 360° situational awareness cues to the driver, rendered for maximum clarity and accuracy.

Additionally, micro-array systems require significantly less power (only a few watts per driver) to deliver sufficient SPL to the listener's ear enabling the usual 12 V rails adequate without expensive DC-DC boost converters for the amplifiers.

Another leading company in all that's audio signal processing and audio algorithms for automotive applications is DSP Concepts, which until recently was a relatively small company based in Santa Clara, CA, with subsidiaries in Boston, MA; Stuttgart, Germany; and Shanghai, China. And now it can claim the biggest hardware partner ecosystem, widest industry support, and largest customer base. DSP Concepts offers Audio Weaver as a hardware-independent audio development platform, currently used in more than 40 million devices within the automotive and consumer electronics industries.

With the introduction of Audio Weaver Designer (AWD), a graphical block diagram editor, DSP Concepts has expanded industry reach by basically attracting audio engineers who understand audio processing but don't master writing coding, simply by combining software modules for audio processing. AWD currently offers more than 400 different modules, all ready-to-use. This is critical in specific applications where it is hard to predict or simulate the challenges, and multiple real-time tests are the only solution. And now DSP Concepts is also leading the way with its TalkTo software for voice control in environments with very high noise levels (e.g., the car).

In the automotive sector, Audio Weaver software is used by companies as varied as BMW, Tesla, and Porsche, which has helped raise funding to support its expansion from carefully selected investors. Not surprisingly, DSP Concepts has now raised more than



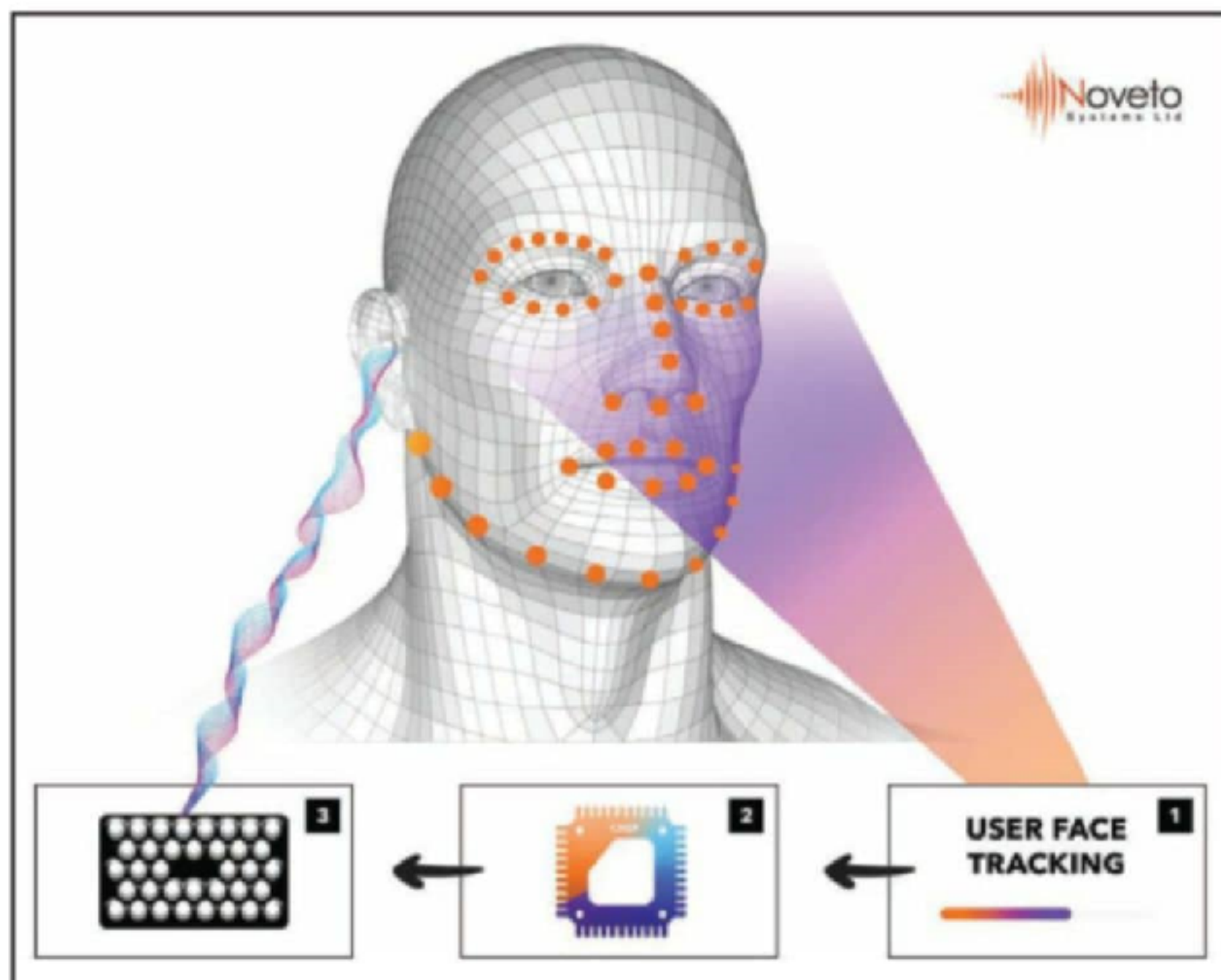
\$25 million in total funding, and Porsche Ventures acquired a direct stake in the company.

DSP Concepts' software is being used by major car manufacturers for complete specification of hardware, software, and acoustic components in turnkey designs, including final cabin tuning, but Audio Weaver was also key in the recreation of the typical Porsche engine sound, generated for the brand's latest Taycan electric model. This clearly shows the diversity and expansion of application fields in automotive audio, making it one of the most excitement segments in the near future.

Today's car dashboards offer a fully integrated array of all the car's technologies and connected services, all powered by sophisticated software and digital signal processing.



The Bose headrest speaker system is integrated in the Nissan Kicks model.



Noveto Systems proposes a different approach that focuses sound to create a personal zone by tracking a user's face.

Analog Devices

Internal and External Engine Sound for Electric and Hybrid Electric Vehicles

Analog Devices introduced an embedded system specifically for generating engine sound for electric vehicles (EV) and hybrid electric vehicles (HEV). The ADSP-BF706 digital signal processor and Electric Vehicle Warning Sound System (EVWSS) firmware enables automobile manufacturers to comply with safety regulations mandating external engine sound for EVs and HEVs traveling at low speeds.

The ADI solution includes a complete hardware and firmware reference design. It can be scaled for high performance with the ADSP-BF706 or entry-level applications with the ADAU1450 Digital Audio Processor. The ADSP-BF706 can also be used to create in-cabin engine sound to contribute to an improved driving experience for vehicle owners. The ADSP-BF706 uses memory mapped quad SPI memory providing faster and simpler access to the stored audio files used to create the engine sound. It can access up to 24 WAV files simultaneously, compared to conventional products that can typically access only five.

ADI's Sigma Studio Integrated Development Environment (IDE) enables OEMs to graphically tune the audio sound when using the ADAU1450. This enables sound engineers with minimal programming experience to optimize audio performance, while reducing development time. In addition, a low-cost CAN software stack runs on the ADSP-BF706 to help developers quickly build automotive-grade prototypes.

The ADSP-BF706 features a Blackfin+ core with up to 400 MHz performance, dual 16-bit or single 32-bit MAC support per cycle, 16-bit complex MAC and many other instruction set enhancements. Packaging for the ADSP-BF706 is 88-Lead LFCSP, while the audio processor ADAU1450 is a 72-Lead LFCSP chip.

Analog Devices also offers additional digital signal processors (DSPs) qualified for automotive applications, all featuring a unique hardware architecture optimized for efficient audio processing and supporting the emerging demands of new and improved audio algorithms. The ADAU1466 and ADAU1467 SigmaDSP processors offer market-leading performance in fixed point DSP processors, while the ADAU1463 and ADAU1467 utilize an 88-pin LFCSP package and feature eight pins that can be configured as serial data inputs or outputs. These additional pins provide greater flexibility in amplifier or head unit designs to support applications such as hands-free, active noise cancellation (ANC), and acoustic noise reduction, among others. The ADAU146x series uses ADI's SigmaStudio graphical programming tool that allows signal flows and applications to be developed and tuned quickly and with a minimum learning curve.

www.analog.com



Silentium

QuietBubble Licensed by AKM

While electric vehicles may be too quiet, requiring the generation of additional awareness sounds, the need for quietness in the car cabin is also a main concern in the automotive industry. Silentium, a company located in Israel with a global R&D and sales base, is working with many industry partners to deliver solutions for automobile interiors based on its unique active noise control technology.

Recently Asahi Kasei Microdevices (AKM) concluded a license agreement with Silentium to incorporate its proprietary QuietBubble (QB) software technology for Engine Order Cancellation (EOC) directly into AKM's audio voice processors, and to distribute such products worldwide. In addition, AKM's audio voice processors incorporating Silentium's Active Noise Control (ANC) technology software will reduce not only engine



noise but also any noise entering the car cabin.

While EOC is one of the active noise control technologies that generates a reverse phase signal specific to the noise generated from the engine and cancels unwanted noise, ANC technology offers the same against any unwanted noise.

www.silentium.com

Audio Precision

A2B Multichannel Audio Test Solutions

Audio Precision, in collaboration with Mentor, a Siemens business, supports multichannel audio performance testing of A2B (Automotive Audio Bus), the digital audio bus technology developed by Analog Devices to deliver high-quality audio over an unshielded, twisted-pair wire.

Distributing audio and control data together with clock and power over a single wire, A2B significantly reduces the weight of existing cable harnesses (by upward of 75% in key applications), resulting in improved vehicle fuel efficiency.

Audio Precision's APx500 audio analyzers support connectivity with Mentor's A2B Analyzer to deliver closed-loop, multichannel A2B audio testing, providing A2B network

configuration and simulation, as well as flexible routing and high-fidelity audio.

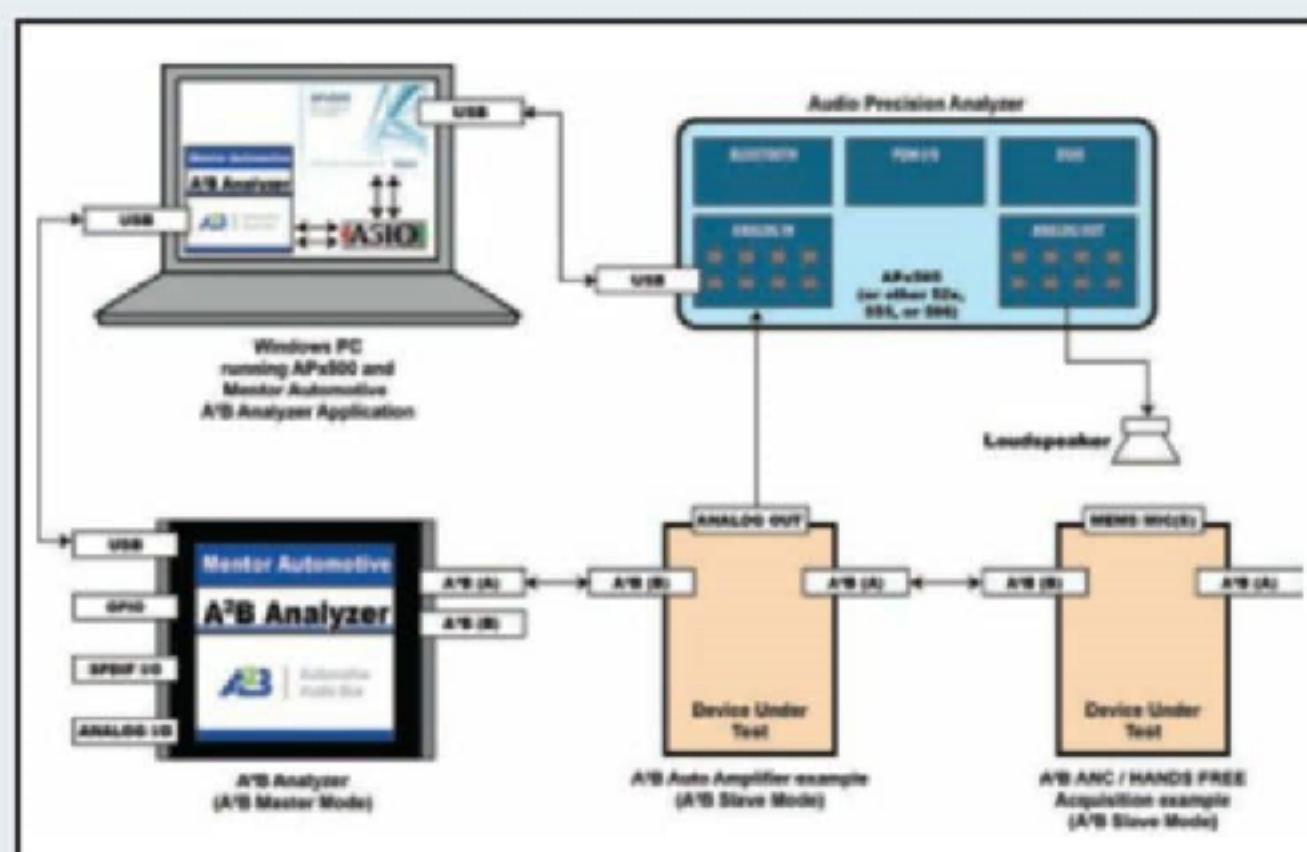
From simulation and bench testing in early-stage development to validation and production testing, having a closed-loop system to measure audio and dynamic signal performance over an A2B network is an integral part of incorporating the bus technology into next-generation automotive systems.

At the same time, APx audio analyzers offer a wide range of interface options, including analog, Bluetooth, digital serial (TDM) and PDM, working as a complete system, supporting hands-free calling, active noise cancelling (ANC), and entertainment systems in vehicles.

www.ap.com



The Mentor Automotive A2B Analyzer System, combines with Audio Precision's APx500 audio analyzers to provide extensive test and measurement options for automotive applications.



An example use case of closed-loop A2B audio testing through the combined capabilities of Audio Precision's APx500 Audio Analyzer System and Mentor's A2B Analyzer System.

Dirac Research

Audio Processing for Car Audio Systems

Bringing truly premium sound to the automotive industry has always been a challenge. The less-than-ideal placement of loudspeakers and complicated car cabin acoustics have both been significant roadblocks for perfect audio experiences in all seats in the cabin.

Swedish audio processing specialist Dirac developed sophisticated sound quality optimization, sound field control, and room correction solutions that have been in use in high-fidelity and consumer electronics for more than a decade, but actually were developed for automotive applications in the company's origins and have been adopted by leading automotive brands such as Volvo, Bentley, Rolls Royce, and BMW.

Center among those solutions is the Dirac Live digital room correction technology that helps optimizing the impulse responses and magnitude frequency responses of the car sound systems. The signal processing solution helps define stereo imaging, avoid boomy and uneven bass response, and deal with frequency response problems in all seating positions. Critical to these corrections is impulse response optimization. By acting in the time domain in parallel with the frequency response of a sound system, Dirac Live improves all the listening area, not just at a single point.

Also, the dedicated toolset offered by Dirac for this purpose, allows tuning the car sound much faster. The software technology is implemented in the amplifier or the head unit.

Dirac Panorama Sound is another sound processing innovation from Dirac, designed to expand the soundstage from the loudspeakers placed inside the cabin, including from only an array of small loudspeaker elements in the dashboard of the car, with no speakers in the doors. Dirac Panorama Sound is a measurement-based tuning and audio enhancement solution that broadens the soundstage well beyond the physical position, creating a perceived audio experience inside the car that surpasses design limitations. Harman International implemented early designs based on the technology.

Dirac Unison

More recently, expanding on Dirac Live, the company developed Dirac Unison, using the company's most advanced signal processing methods to implement those sound corrections and optimizations, taking the loudspeakers and the listening space as a single unit, rather than as individual elements that interfere with one another.

Based on ideas from the fields of active noise control, sound field synthesis, and room correction, Dirac Unison uses precise measurements to create an active acoustic treatment of the listening space. Dirac Unison also features a completely new kind of bass management that automatically adapts to the room and speakers. By co-optimizing up to a certain frequency, sound systems can deliver a bass response that is more unified and tighter than previously thought possible in a car.

The first commercial application of Dirac Unison was in the premium audio system developed by Bowers & Wilkins for the Volvo XC90.

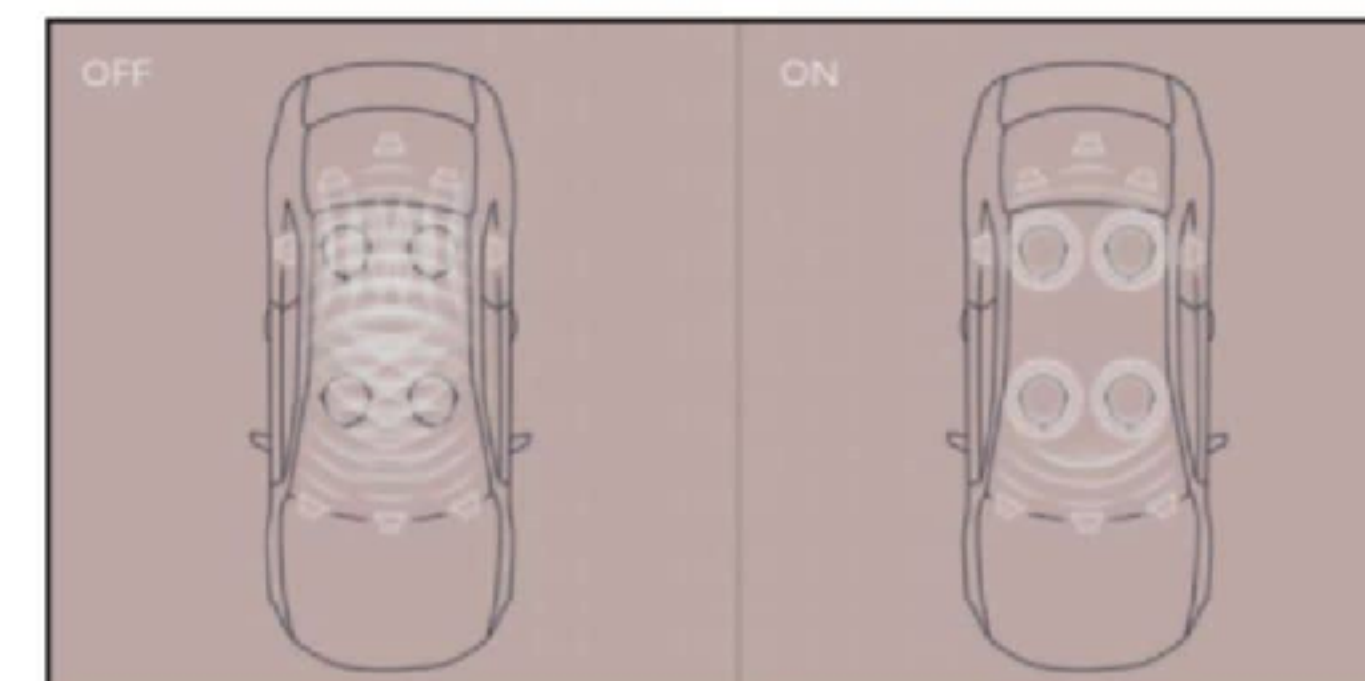
Other technologies developed for automotive audio include Dirac Virtual Center, to correct the stereo sweet spot from near-side bias and reposition the entire soundstage to the front, instead of the closest loudspeaker. For each car model, the optimal filter

is calculated carefully, allowing a high level of consistency across car models, and the technology can be implemented by FIR filters or a number of all-pass IIR Biquads, which ensures flexibility for different DSP platforms. The Dirac Virtual Center technology can be combined with other Dirac processing technologies.

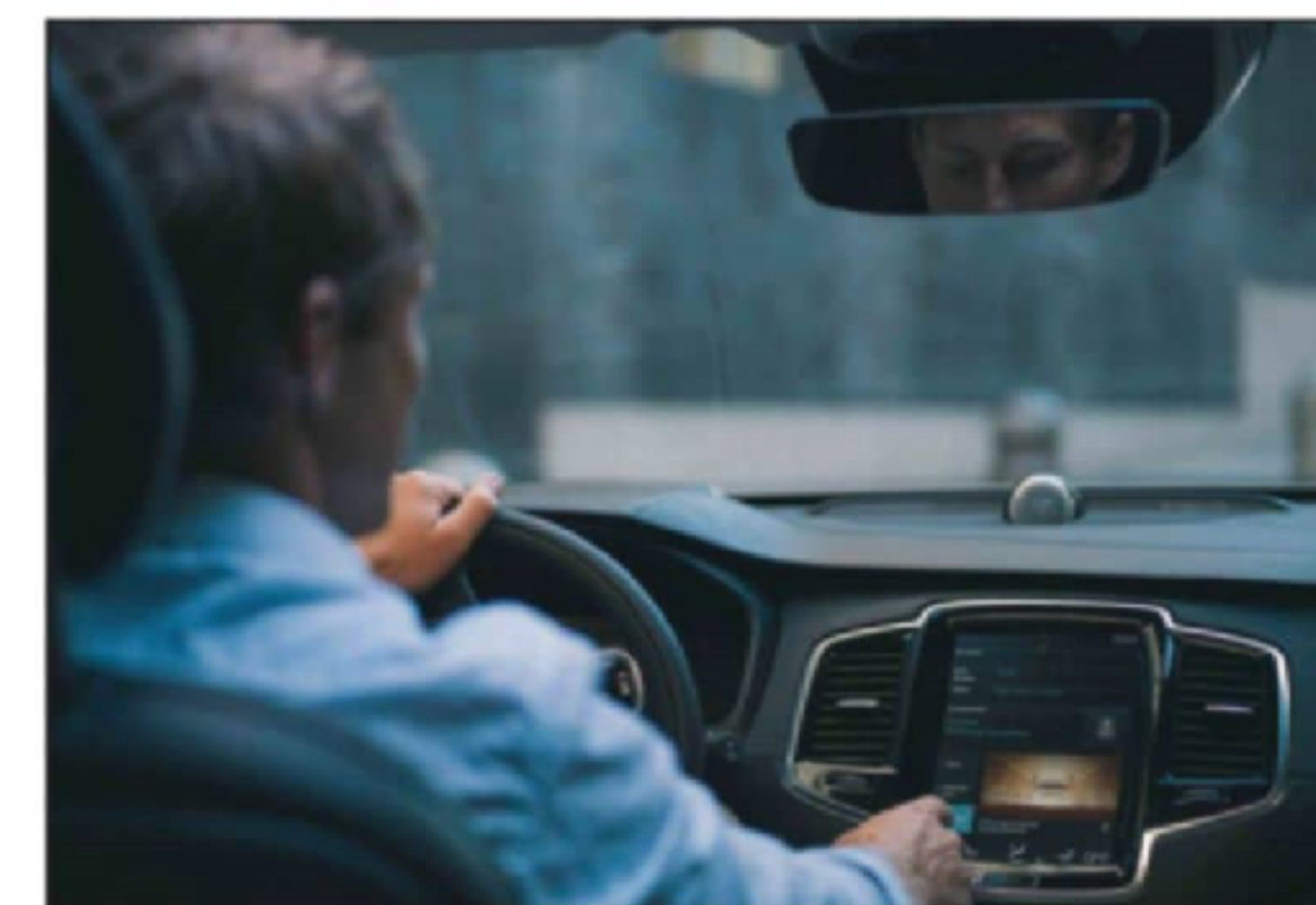
www.dirac.com



Dirac automotive processing solutions include its integrated optimization solution Dirac Unison, which also features a completely new kind of bass management that automatically adapts to the cabin and speakers.



Dirac's integrated audio processing solution allows creating optimized audio experiences for all passengers inside the car.



Dirac Unison is integrated into Dirac's proprietary automotive tuning tool, enabling tuning automotive sound systems with unprecedented speed and quality-consistency across all individual cars within the model.

Resonado

Flat Core Speaker Ideal for Automotive

A more organic path to low-profile automotive speakers was recently introduced by Resonado, a South Bend, IN company, and a spin-off of the University of Notre Dame, which introduced the technology at CES 2020.

Resonado's Flat Core Speaker (FCS) architecture enables unique installation placement. FCS is a new take on woofer topology using two parallel flat voice coils - both mounted perpendicularly to its flat "racetrack" diaphragm enabling thinner, lighter, and more efficient speakers.

The full-range FCS speaker is identical to the woofer except that a single planar voice coil is used. What is ingenious is that the planar voice coil drives the full length of the long and narrow diaphragm, achieving wide uniform coverage without the breakup challenges of a conventional round voice coil used with a rectangular diaphragm. The cross-section of the soft parts is an I-beam formed by the diaphragm, the vertical planar voice coil, and the spider, which is located behind the voice coil.

Auto-sound applications for these unique full-range FCS drivers are extensive, including in-door speakers with clearance for full

window roll-down—and most intriguing, the center stage in-dash, the A and B pillars and head rests. When the racetrack speaker is oriented vertically, it will yield wide line source horizontal coverage while reducing midrange ceiling and car window reflections. With an aspect ratio typically of 4 to 1, far more piston area results in lower cost and less weight than conventional round speakers. The FCS diaphragm material selection is dependent on the application. Where higher sensitivity and extended response past 20 kHz are critical, TeXtreme's TPCD composites are an upgrade option.

Initial Resonado auto-sound design wins include the speakers for a select few of Airstream's ultra-luxury coaches built on Mercedes Benz Sprinter Chassis.

www.resonado.com



Continental and Sennheiser

Speakerless Immersive Sound

At CES 2020, Continental and Sennheiser introduced a new audio system for vehicle interiors, intended to generate more realistic immersive experiences. The project combines Sennheiser's patented AMBEO 3D audio technology with Continental's Ac2ated Sound system. The concept departs from conventional speaker technology and basically adopts an actuator approach to excite select surfaces in the vehicle interior to produce sound. In comparison to conventional audio systems, Continental says, Ac2ated Sound enables a reduction of weight and space of up to 90%. Combined with Sennheiser's AMBEO Mobility spatialization algorithm, the concept is said to generate 3D sound reproduction that envelops passengers, even from existing stereo material, while at the same time saving space and weight, as required for electric vehicles.

Specially developed actuators excite specific surfaces in the vehicle interior so that they emit sound in different frequency ranges.



www.continental.com | www.sennheiser.com

Resources

Analog Devices | www.analog.com/A2B
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 Apple CarPlay | <https://www.apple.com/ios/carplay>
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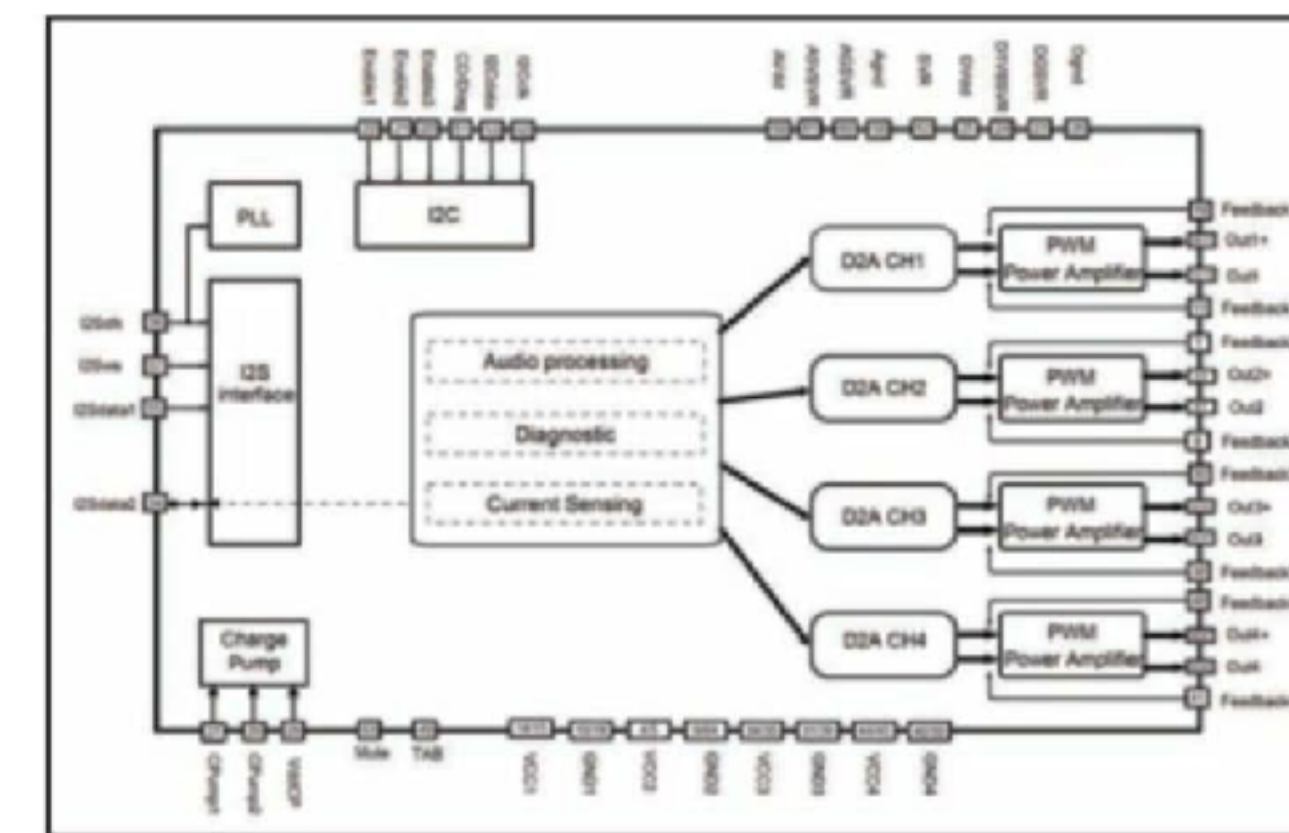
STMicroelectronics

New-Generation Automotive Class-D Audio Amplifier IC

STMicroelectronics (ST) recently introduced its new FDA901 Class-D audio amplifier IC, a new design that incorporates contributions from Japanese manufacturer of car audio equipment, Alps Alpine. The new ST chip aims to contribute to the creation of multi-function, high-fidelity car audio systems that fuse the high efficiency of Class-D amplifiers with the high-quality sound of ST's Class-AB amplifiers.

As ST explains, the new FDA901 IS combines low residual noise with low distortion ratio, a flat frequency response achieved through feedback technology, and low electromagnetic interference (EMI) levels. The new solution is equipped with a range of self-diagnostics and other advanced functions to measure real-time speaker current and impedance, factors prized in the next-generation car audio market for diagnostics and to compensate for loudspeaker imperfections.

Specifically, the FDA901 IC is a new BCD technology Quad Bridge Class D amplifier, using a high-performance DAC together with powerful MOSFET outputs in Class D, to get an outstanding efficiency compared to the standard Class AB. The integrated DAC with support for sampling frequencies from 44.1 kHz to 192 kHz, ensures a 115 dB signal-to-noise ratio (SNR) with 110 dB of dynamic



range. The feedback loop includes the output L-C low-pass filter, allowing superior frequency response linearity and lower distortion independently from the inductor and capacitor quality.

The FDA901 is fully configurable through I²C bus and it includes not only a complete load diagnostic matrix but also a unique real time load current acquisition on I²C and I²S data lines, enabling sophisticated speaker monitoring functions. Thanks to the solutions implemented to contain EMI emissions, the device is intended to be used in the standard single DIN car-radio box together with the tuner. Finally, the FDA901 is able to work with power supply as low as 6 V, thus supporting the most recent low voltage (start-stop) car-makers specification.

Starting from the adoption in 39 models of Alpine's new BigX series of after-market car-navigation systems to be released in the first half of 2020, the FDA901 IS will also be adopted by several major car makers.

www.st.com

HEAD acoustics

coreA2B High-Performance Solution for Automotive Audio Performance Testing

HEAD acoustics is one of the world's leading companies in software and hardware for the measurement and analysis of sound quality in communications and automotive applications. The German company works closely with many leading car manufacturers in creating made-to-measure software for that industry, and offers an extensive catalog of interface and front-end solutions in critical areas of car design and acoustic analysis.

The company offers solutions for multiple automotive systems, including for CAN-fd (flexible data-rate), and developed early solutions to support Analog Devices' Automotive Audio Bus (A2B) technology.

Specifically, HEAD acoustics offers coreA2B, a high-performance solution for testing audio performance on the A2B bus technology implemented on a measurement extension board for the

company's multi-channel hardware platform labCORE. An ideal solution to support hands-free calling, voice recognition, personal audio and active noise cancellation (ANC) development inside modern vehicles.

Analog Devices' Automotive Audio Bus (A2B) networking technology transports digital audio over a single unshielded twisted pair of cables, providing critical support to the vehicle infotainment systems with higher fidelity and less cabling.

HEAD acoustics' new coreA2B extension board for labCORE connects to any A2B bus in four user-selectable modes: Master mode, Slave mode, Bus monitor mode, and Proxy mode.

Proxy mode is the primary mode of operation of coreA2B and connects anywhere on the bus, enabling developers to manipulate all of the digital data via



coreA2B is a new labCORE extension board for testing audio performance on the A2B bus technology. ©HEAD acoustics GmbH

HEAD acoustics communication quality analysis software ACQUA. Only Proxy mode enables to record, process and send user-defined audio and configuration data from and to any channel and any node while the bus itself remains fully operational.

In conjunction with labCORE and ACQUA, the Proxy mode allows coreA2B to receive, mix, and insert arbitrary signals to the bus without interference with the original, unaltered signal. coreA2B does not occupy the master or any slave node position, thus developers can also use this mode on fully equipped buses.

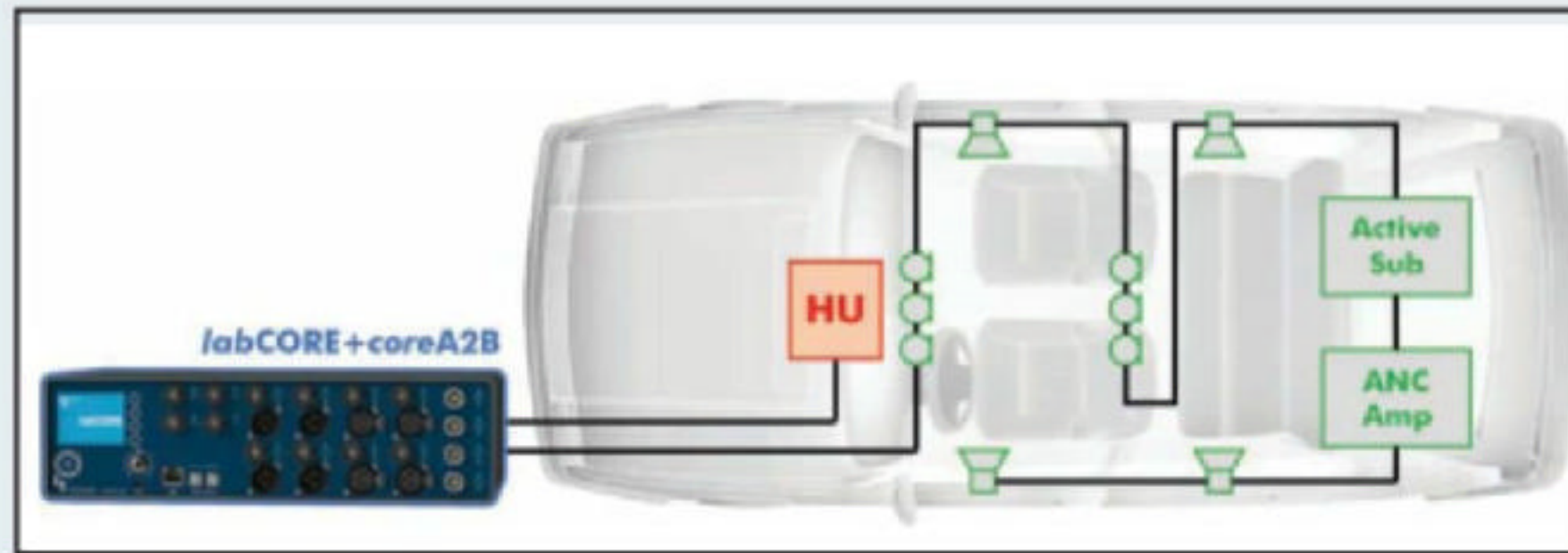
Another key characteristic of coreA2B is the Bus monitor mode, for analyzing data traffic on the bus without interference (e.g., when trouble-shooting an existing bus). The new extension board acts as a “neutral entity,” sniffing out audio and configuration data at any point on the bus.

Besides the above-mentioned modes, coreA2B can operate as a Master. In this mode, the new extension board replaces the original master (e.g., the head unit) and acts as the new master for the bus. Developers can connect up to 10 slave devices to coreA2B. The Slave mode, on

the other side, enables the labCORE extension to act as a slave node on an existing bus. Developers are able to insert a node at any arbitrary position. In this mode, coreA2B can send and receive audio data as well as receive configuration data.

Developers can also use coreA2B as an evaluation board for the A2B bus. In this case, it operates in Master mode and allows using all capabilities of analyzing, filtering, and manipulating signals that ACQUA offers.

www.head-acoustics.com



The Proxy mode of coreA2B enables developers to actively receive and insert user-specified signals from the fully-operational bus. ©HEAD acoustics GmbH

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Our Future Car: Audio Bubbles, Sterile Cockpits, New Business Models

By
Elia Shenberger
(CEVA)

We probably listen to more music and podcasts in our cars than anywhere else. Which means this is where we're pushing the boundaries of audio experience—less noise, personalized sound zones, and control. And we're more aware of the safety implications, not distracting or tiring the driver.

The automotive industry is rapidly closing the gap in adopting the new technologies we already know in consumer devices. We want to have the latest features that are being advertised by the car manufacturers, but we also expect safety and reliability over much longer timeframes than we expect from our phones. OEMs and Tier 1s are aggressively rethinking our experience in the car, with more focus on personalized audio experiences around the cabin, higher-quality audio performance, more flexible access to streaming options and, always important, increased safety to offset driver distraction and fatigue.

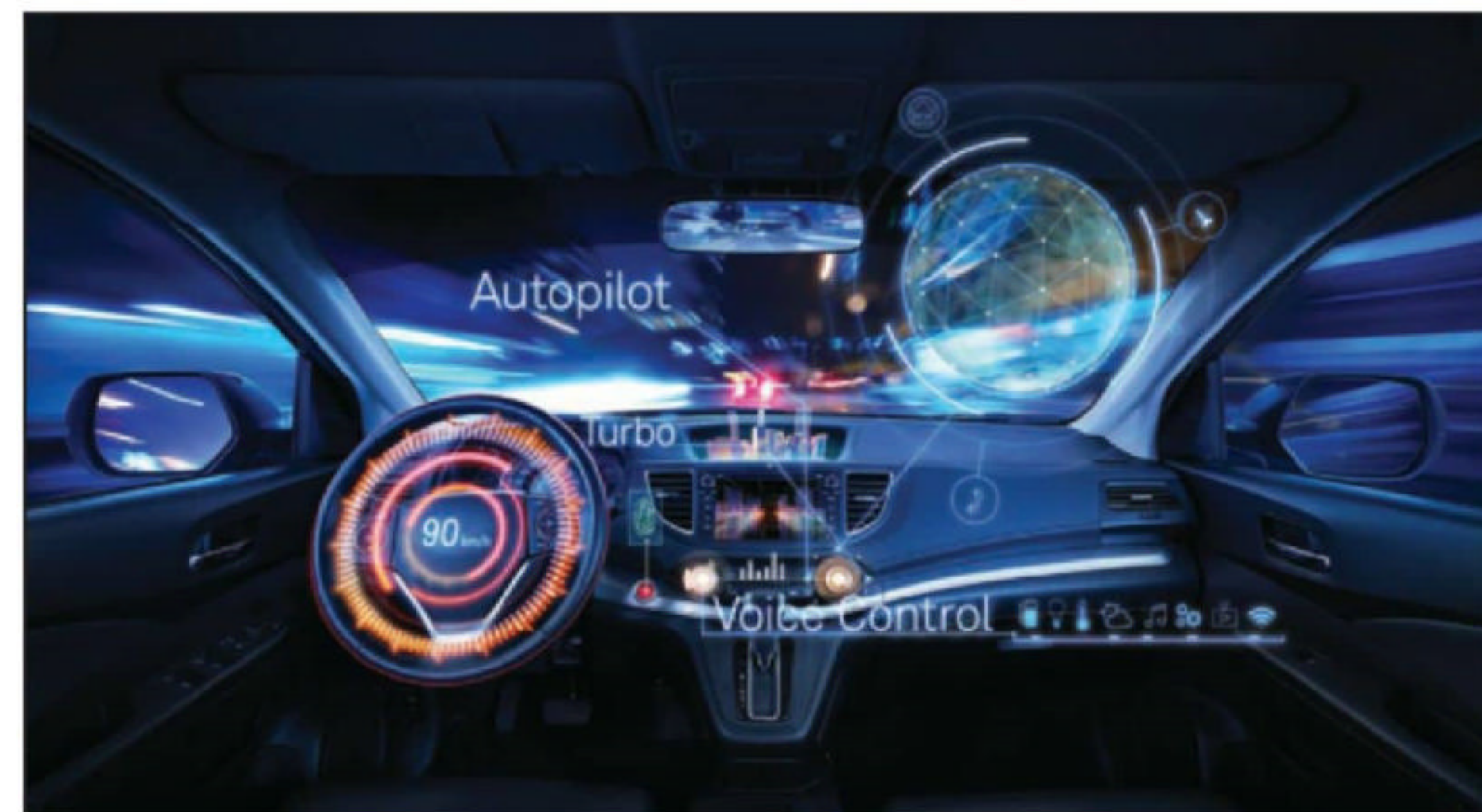
Enhanced Audio Experience

One area we keep seeing Automotive OEMs investing in is improving in-car audio experiences, especially in acoustic isolation and noise control technologies. For cars with internal combustion engines, active noise control (ANC) is important to cancel whatever engine noise gets through to the cabin. As an example, the Genesis GV80 now offers road active noise control (RANC), using data from acceleration sensors around the car to predict

noise and generate in real-time reverse-phase audio signals for noise cancellation in the cabin, improving In-Car-Communication (ICC) capabilities. Some OEMs design in-house solutions and some use aftermarket solutions by companies like Silentium.

Increasingly, we demand fully immersive sound, with speakers mounted around the car interior, providing a high-quality 3D surround sound experience in premium cars certainly but increasingly in value-priced cars as well, providing premium quality audio even when supported by entry-level hardware. Fraunhofer Symphoria provides one such solution.

Separated audio zones in the car are already possible. Dirac provides systems for high-end automobiles allowing active separation between front and rear seats and a constant sweet spot in each, regardless of your position. This concept is now extending to personal zones. Silentium offers its Personal Sound Bubble to create independent sound zones for the driver and a rear passenger. Hyundai-Kia offers a proprietary Separated Sound Zone technology to provide a personal experience per passenger. Harman Individual Sound Zones



provides a similar experience. All of these solutions work without the need for headphones through clever audio cancellation between locations.

Safety and Situational Awareness

These audio technologies will greatly improve our in-car experiences and communication, but how will they affect the driver when intermingled with alerts and notifications? It isn't difficult to overload the best of us with too many inputs. Even if we're not overloaded, a stream of inputs requiring attention—including entertainment—can become tiring. An overloaded or tired driver is not a safe driver.

This concern becomes even more acute as we automate more of the driving task. It's no surprise that the more we automate for a driver, the less rapidly they can respond to potential dangers.

Some refer to a sterile cockpit rule to minimize distractions. This is already common in aircrafts. To a first order that will require turning down other distractions when, say, a need to brake or swerve may be detected. To a second order, it might require more anticipation. As an obvious example, if sensors detect fast-moving and heavy traffic, everything else should be turned down or off.

Monetization

There are already signs that the business models for cars may be evolving (e.g., a car-as-a-service model, which is making a debut now). Meantime, we see revenue opportunities for the audio premium services increasingly likely to explore subscription models. If Sirius-XM and Spotify can do it, why not others? In-car streaming audio, in particular, is a market ready for rapid growth, moving us away

from the limitations of built-in captive hardware.

Supporting Technology

Many of the technologies and experiences mentioned here depend heavily on fast, efficient, and affordable signal processing solutions, for contextual awareness in the cockpit, for noise and echo cancellation, beamforming and audio zoom, personal sound bubbles, voice activity detection, voice biometrics, and accurate voice command recognition. A number of these will be needed not just in the cockpit but also in other areas of the cabin—there's little value in separated sound zones if only the driver can control what's playing in your zone. Which means that signal processing and sensor fusion is likely to play an increasing role in our future cars.

About the Author

Elia Shenberger is a Business Development Director in the CEVA Sound BU, and has more than 15 years of experience in the Professional and Consumer Audio industries. Before joining CEVA, he was the Director of Strategic Alliances at Waves Audio, building and managing strategic partnerships and relationships with Fortune 500 semiconductor companies, as well as with OEM mobile and PC customers. Shenberger is also a professional audio engineer, who graduated from SAE Sydney.

Resources

CEVA, Inc., <https://www.ceva-dsp.com>

CEVA, Inc., <https://www.ceva-dsp.com/app/audio-voice-and-speech>

"CEVA Announces SensPro High Performance Sensor Hub DSP Architecture for Smart Devices," *audioXpress*, <https://audioxpress.com/news/ceva-announces-senspro-high-performance-sensor-hub-dsp-architecture-for-smart-devices>

Redefining Surround Sound in Cars

By
Birgit Bartel-Kurz
(Senior Engineer at Fraunhofer IIS, Audio and Media Technologies Division)

Today's car buyers are looking for something special when they search for a new vehicle. The interior and optional equipment are becoming increasingly important, including sound systems. And these systems are now able to reproduce sound in a more immersive way than ever.

Modern algorithms make it possible to offer high-grade surround sound in vehicles, making driver and passengers feel as if they were on stage with the band they are listening to, or sitting in the midst of a string quartet. But enabling immersive audio playback in the car is a challenge. Neither the speakers nor the passengers are in the right position for a perfect listening experience, and while the passengers sit amidst the speakers asymmetrically, the many smooth window surfaces evoke unwanted reflections. Conventional audio systems hardly overcome the limitation and are not able to reproduce an impressive, genuine spatial sound experience.

The latest automotive audio signal processing technologies now enable music enthusiasts to feel the sound in all its emotional intensity even while on the road. Solutions such as Sonamic Panorama from Fraunhofer IIS make it possible to use a stereo input to produce an immersive in-car sound experience. Sonamic Panorama was developed for medium, compact, and small class vehicles and makes optimal use of existing hardware components. The underlying algorithm separates the individual stereo sound sources and evenly distributes them in a U-shaped soundstage that surrounds the passengers. In addition, the playback is kept free of artifacts and is robust to poor reception quality.

Stereo Sound Goes Immersive

While conventional surround algorithms focus

chiefly on the distribution and partly also on additional generation of ambient components, Sonamic Panorama implements a rendering concept that primarily analyzes the direct sound components and actively works with them. The underlying algorithm extracts the individual sound sources from a stereo recording and fans them out evenly around every single occupant of the vehicle. Sound sources that are panned to the sides in the stereo image are placed at the sides or the rear sides, while central sources remain acoustically unchanged in front of the listener. This lets listeners better appreciate and discover details of the sound mix that would otherwise obscure each other in the original stereo playback.

A stereo recording contains a position in the stereo image for each sound element (e.g., singing and instruments). This corresponds either to the real position of the instruments during recording or else is subsequently assigned during the mixing of the material in the studio. The lead vocals, say, are generally placed in the middle. In this example, this means: both channels of the stereo mix contain the same signal components for the singing, so that it seems to come from the middle.

In a stereo playback, the two stereo channels are each assigned to an output channel. If the loudspeakers are arranged in an ideal constellation, the listener experiences the sound in a spatial yet rigid manner with a deflection of approximately 60°,

which recreates the intended stereo image. However, a stereo recording contains even more. Therefore, the Sonamic Panorama algorithm analyzes the image in greater detail and splits the two channels into several directional segments. The segment in the middle contains the components that are located on both sound tracks with the same presence (e.g., the narrator or the lead vocalist). Meanwhile, the outer segments contain the components that are located only on the left sound track or only on the right one. The segments in between contain components that have a predominantly left or right orientation.

By taking into account the existing loudspeakers and their positions, these segments can be fanned out around the listener, whereby sound sources situated on the edges of the stereo image are placed at the sides or from further back (see **Figure 1**).

In addition to the direct sound components, a stereo recording also contains diffuse, so-called ambient room sound components. Sonamic Panorama can recognize these and separate them from the original recording. For the desired panorama effect, this ambient room information is also spatially expanded, before ultimately being merged back together with the distributed direct sound segments.

One Algorithm, Four Steps, and Intensity Levels

The core of Sonamic Panorama is a modern, frequency-based algorithm consisting primarily of four steps that are executed in real time (see **Figure 2**). Step 1 is the analysis of the (stereo) input signals and their separation into direct and ambient sound components. To obtain the directional information present in the recording, an analysis of the direct components in the frequency range is carried out. These are grouped into several directional segments (e.g., left, near-left, center, near-right and right). In Step 2, the segments are then spread out according to their intensity level and the position of the loudspeakers. Step 3 concurrently

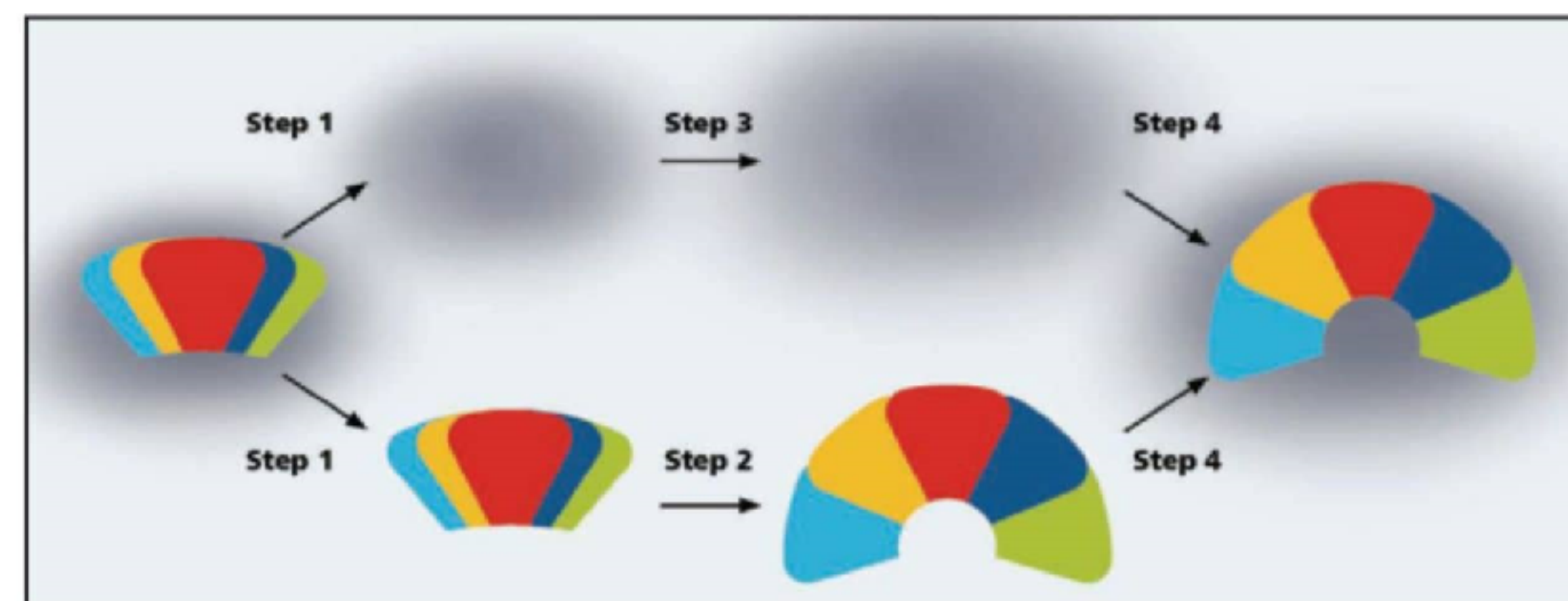


Figure 2: The Sonamic Panorama algorithm has four steps.



Figure 1: The stereo information can be split into several directional segments.

adjusts the ambient components spatially according to the direction of the reverb-producing sources. Finally, in Step 4, the spread-out direct and diffuse components are merged back together in the final panorama audio signal.

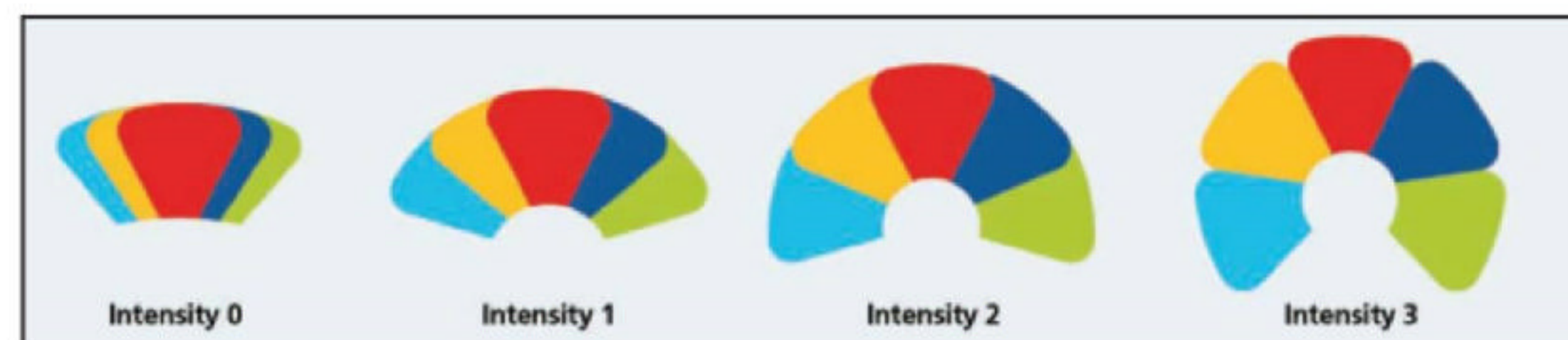
Depending on the sound system's operating concept, the Sonamic Panorama algorithm can be a fixed part of a certain sound preset or it can be connected up by the user as an optional extra.

The final audio signal can be made available in up to four intensity levels: From 0 (stereo) to 3 (intense surround experience) as shown in **Figure 3**. These levels are distinguished by the degree of spreading and the weighting of direct and ambient components. The definition of the individual intensity levels can be coordinated, with their characteristics particularly dependent on the position and quality of the rear loudspeakers.

No Additional Sound Components

The processing of signals takes place in real time without introducing signal delays and switching artifacts even when activating the processing. No extra sound components—whether diffuse or direct—

Figure 3: There are several possible implementations of various intensity levels.



are added: Sonamic Panorama intervenes exclusively in the positioning of the sound events and fans them out. As a consequence, a “dry” recording without dominant room sound components remains dry even after the processing. As for pieces with little in the way of stereo components, such as when the focus is on a narrator, the effect is correspondingly minimal. As such, the narrator remains dominant even after the processing.

Sonamic Panorama is particularly suitable for use in small, compact, and mid-range vehicles and the

panorama effect can be achieved with every basic tuning that generates a conventional, frontal stereo sound image. Equally, combinations with various sound presets and seat-dependent optimizations are possible. The algorithm does not require any additional or special loudspeakers, but is scalable nonetheless. It can be operated from as few as four loudspeaker positions. How far the individual sound segments fan out around the passengers depends on which loudspeakers are fitted behind the passengers and where. If a system has only a small number of

About the Author

Birgit Bartel-Kurz studied Mathematics at Darmstadt University of Applied Science in Germany and graduated in 1987. Prior to joining Fraunhofer IIS in 1998 as a software engineer, she held positions at Bosch Telecom and Mannheim University of Cooperative Education.

After initially working in the field of digital broadcast applications and traffic information systems at Fraunhofer IIS, Bartel-Kurz has dedicated her expertise to the institute's automotive audio technology range as a project manager since 2012.

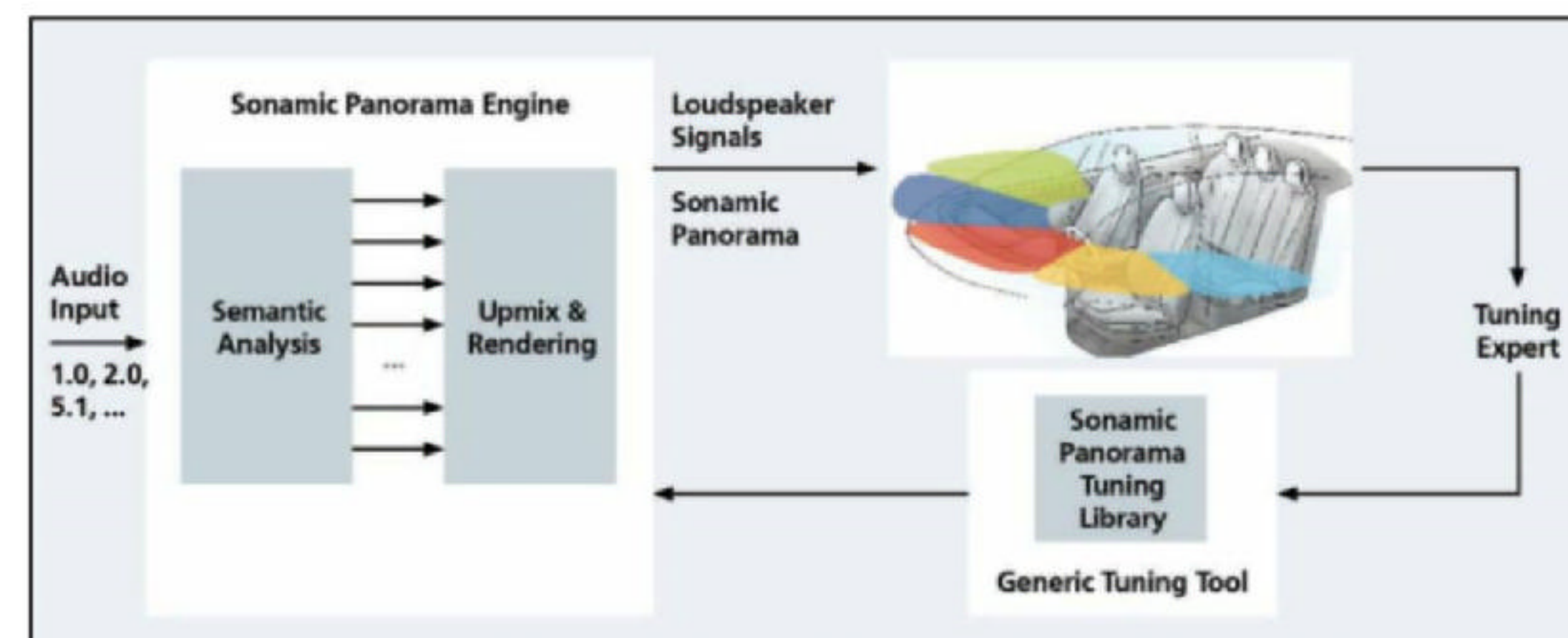


Figure 4: Sonamic Panorama is a modern signal processing algorithm that uses spatial analysis and works in the frequency domain.

loudspeakers, the focus lies automatically on increasing the quality of the sound experience in the front seats. In addition, Fraunhofer sound engineers optimize the sound for every vehicle type through the precise coordination of relevant parameters with the separate Sonamic Panorama Tuning Library (see **Figure 4**).

Through the use of modern, intelligent signal processing based on spatial analysis and the ability to distribute signals in a targeted manner, Sonamic Panorama can achieve a remarkable improvement in sound intensity at no significant extra expense. After all, the small number of loudspeakers required has a positive effect on costs and vehicle weight.

Since the start of 2020, Sonamic Panorama is available in various models from European vehicle manufacturers and enhances the respective in-car entertainment systems with high-grade surround sound. In the future, high-quality audio systems for vehicles are set to become even more important than they are today, especially with regard to autonomous driving: After all, in an autonomous car, the driver no longer has to concentrate on the road. Instead, passengers can make the most of their free time: listen to their favorite music, watch an exciting movie or enjoy a computer game, which will drive the demand for in-car immersive audio even further.

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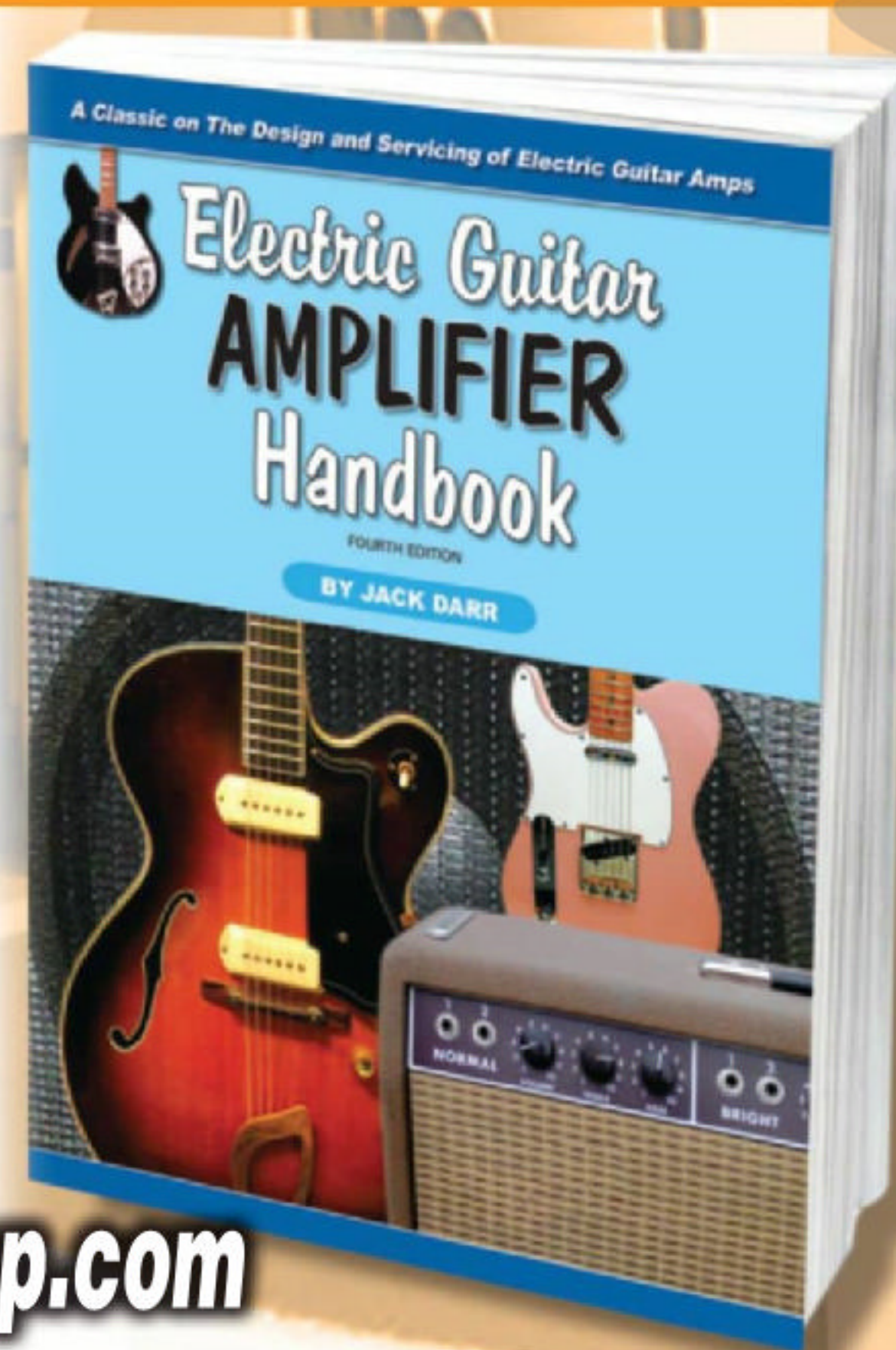
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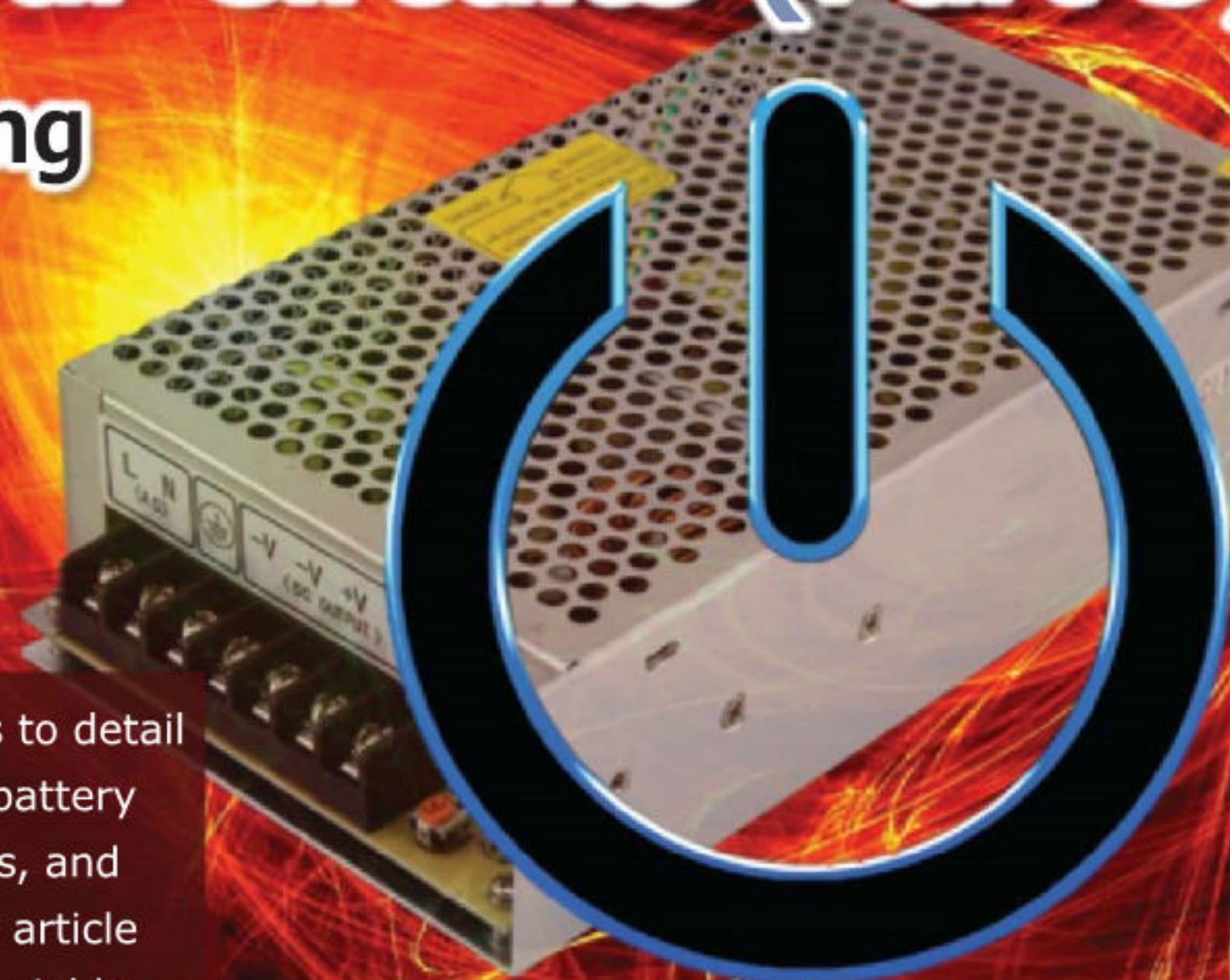
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www.acopacific.com sales@acopacific.com
Tel: +1-650-595-8588

Powering Your Circuits (Part 5)

Mains Noise Filtering

By
Ed Simon
(United States)

In this article series, Ed Simon continues to detail ways to power circuits. After discussing battery power, transitioning to AC power supplies, and some forms of DC power regulation, this article now dives further into diodes, which is a viable method for power supplies.



The low-cost of diodes makes this an attractive method for most critical power supplies. One advantage is that the transformer is no longer directly connected to the rest of the circuitry. Any noise coupled through the transformer can only pass through stray capacitance except when the diodes are conducting.

The test circuit is shown in **Figure 1**. A small ferrite core was used to make a current transformer and this was fed to the oscilloscope's 50 Ω input.

So let us look at the diode's behavior. **Figure 2** shows a close up look at the current when a rectifier diode turns on and then off. This is a single shot without averaging. It shows there is a bit more AC line noise when the diode is conducting.

As you can see, the current rise follows a bit of the sine wave from the AC power source. This tapers off as the small filter capacitor charges.

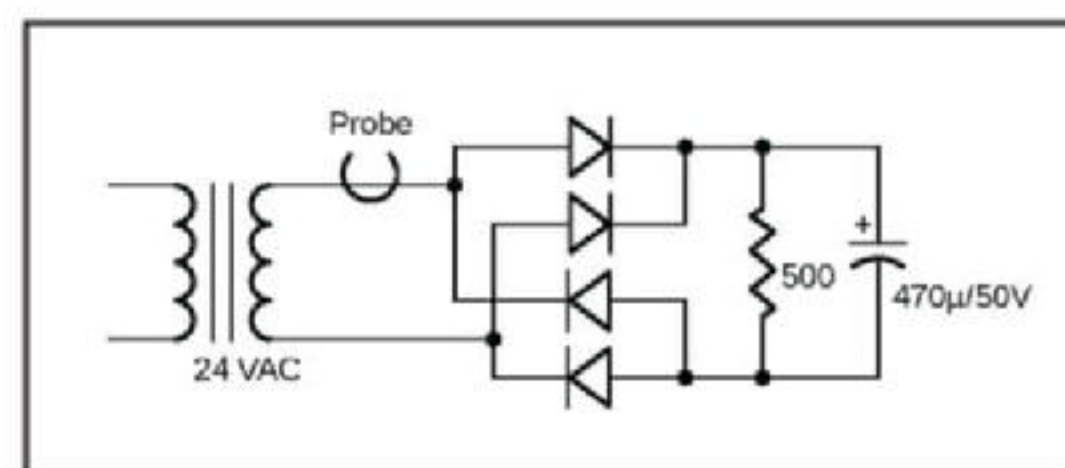


Figure 1: This is the test circuit showing the location of the current probe.

There is little current flow once that happens. Once the voltage on the capacitor is greater than the incoming voltage, then the diode switches off. This is a much larger negative spike. As the diode is opening the circuit, there is almost no direct load on the inductive spike. The diode switch-off spike is very abrupt and, without an easy path to discharge this energy, it may end up radiating or effectively becoming common mode noise.

A designer worried about the switch-off spike causing ringing in the transformer and other circuit inductance could insert a small resistor in series with each diode to provide damping. (Not a very good solution.)

Many designers place a small capacitor across each diode to provide a very short return path for the unwanted energy. This has the advantage of providing a very small bit of extra effective capacitance at high frequencies across the filter capacitors when a full-diode bridge is used. However, it has the disadvantage of coupling more AC line noise past the non-conducting diodes into the following circuit stages.

Of course, to be effective, the leads on the bypass capacitor should be as short as possible. At these switching times, a quarter inch is about

the maximum. Different types of diodes create less noise. There are even diodes designed to minimize such noise. Also note that as the diode begins to conduct current to charge the filter capacitors, there is a sudden high-current draw.

All transformers have some "leakage" inductance due to imperfect coupling and also some inter-coil and inter-winding capacitance. The combination of turn-on and, more importantly, turn-off surge can cause a resonance with the transformer's leakage inductance. This would be treated as differential mode noise.

Typically a capacitor (1 μ F to 10 μ F) and a low value resistor in series across the secondary will reduce this. It should be noted that lossy transformers ring less than well-regulated tightly coupled types such as toroidal transformers. Experimentation is perhaps the best method to determine optimum component values as the inductance values are often not even specified.

Turning AC into DC introduces noise into both the system and back out into the AC power line. Many other sources, such as strong local AM radio stations or data sent over power lines via RF modulation, can pollute the AC power. Most of the noise, however, comes from inside your own home. Computers, appliance motors, electric stove controls, lighting dimmers, other audio equipment, and even cell phone chargers all add noise to the AC line.

External Noise

It is worth taking a look at the external noise that rides in on the AC power line and how it may get past our complete power supply design.

Noise on an AC power line can be classified as either common mode (CM) or differential mode (DM) noise. In common mode, the noise is the same on both AC power wires. It can be from sources such as external radio stations or similar sources where the power lines act as an antenna to pick up the noise. Differential mode is where the noise source is added to the voltage of the AC power line. Most of the transformer and diode noise we have seen would go back onto the AC power line as differential noise.

The CM noise as shown can be addressed by a shield or physical separation between primary and secondary windings. Another method is the use of CM inductors (see **Photo 1**). These are essentially high-frequency transformers where the AC line is passed through the coils. Any CM noise is thus passed from one wire back to the other cancelling the CM signals.

For DM noise the common method is to short it out by placing a capacitor across the AC power line. This value may go as high as 10 μ F. This capacitor

must be rated for across the AC line use. In small values, there are special capacitors (X class for across the line Y class for line to ground) made to be used at line voltages. For larger values, a "Motor Run" capacitor rated at 250 VAC or more may be used for 120 V lines.

One common technique to reduce line noise is the use of a prepackaged filter. These come with detailed specifications including the added loss they

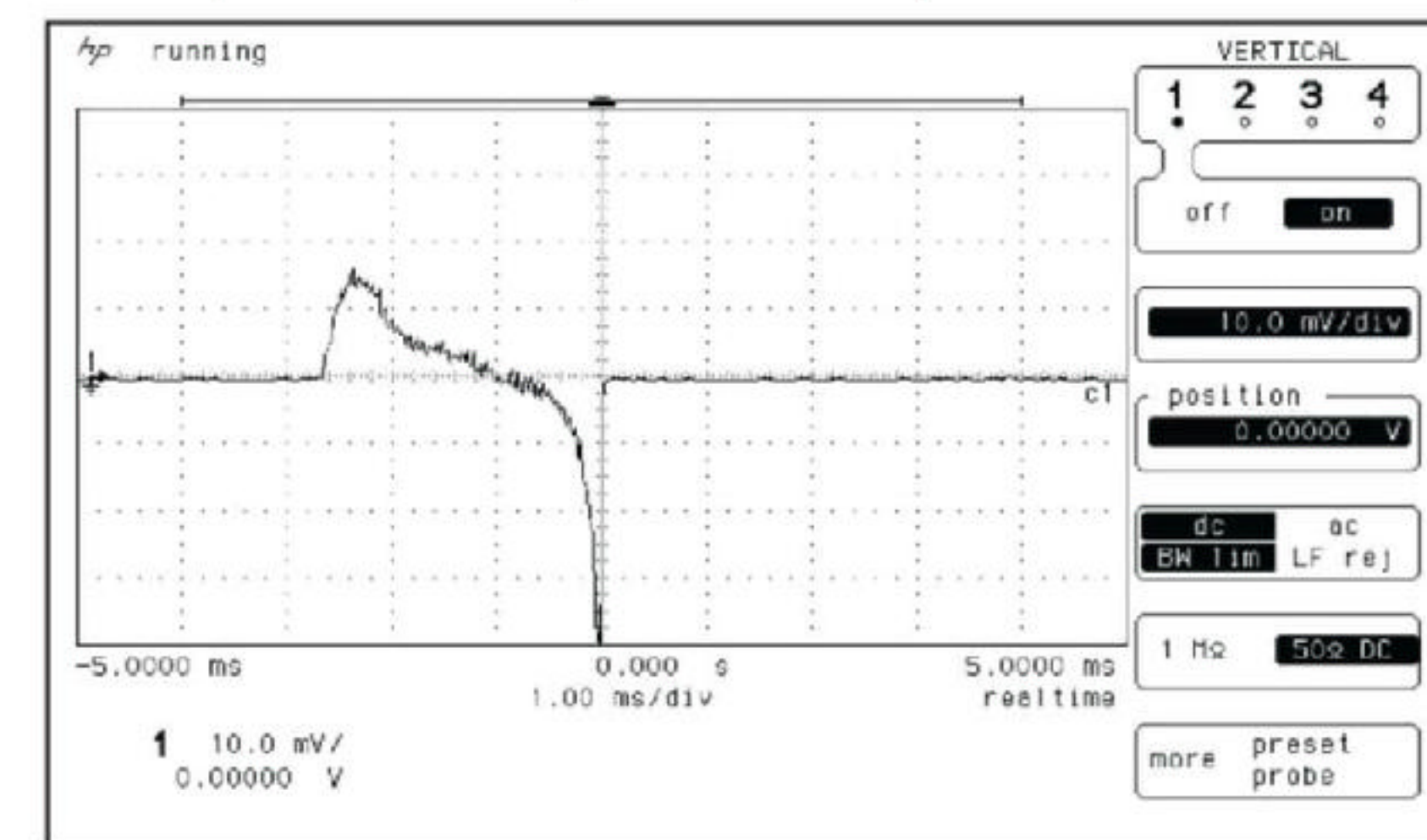


Figure 2: This graph shows a close up look at the current when a rectifier diode turns on and then off.

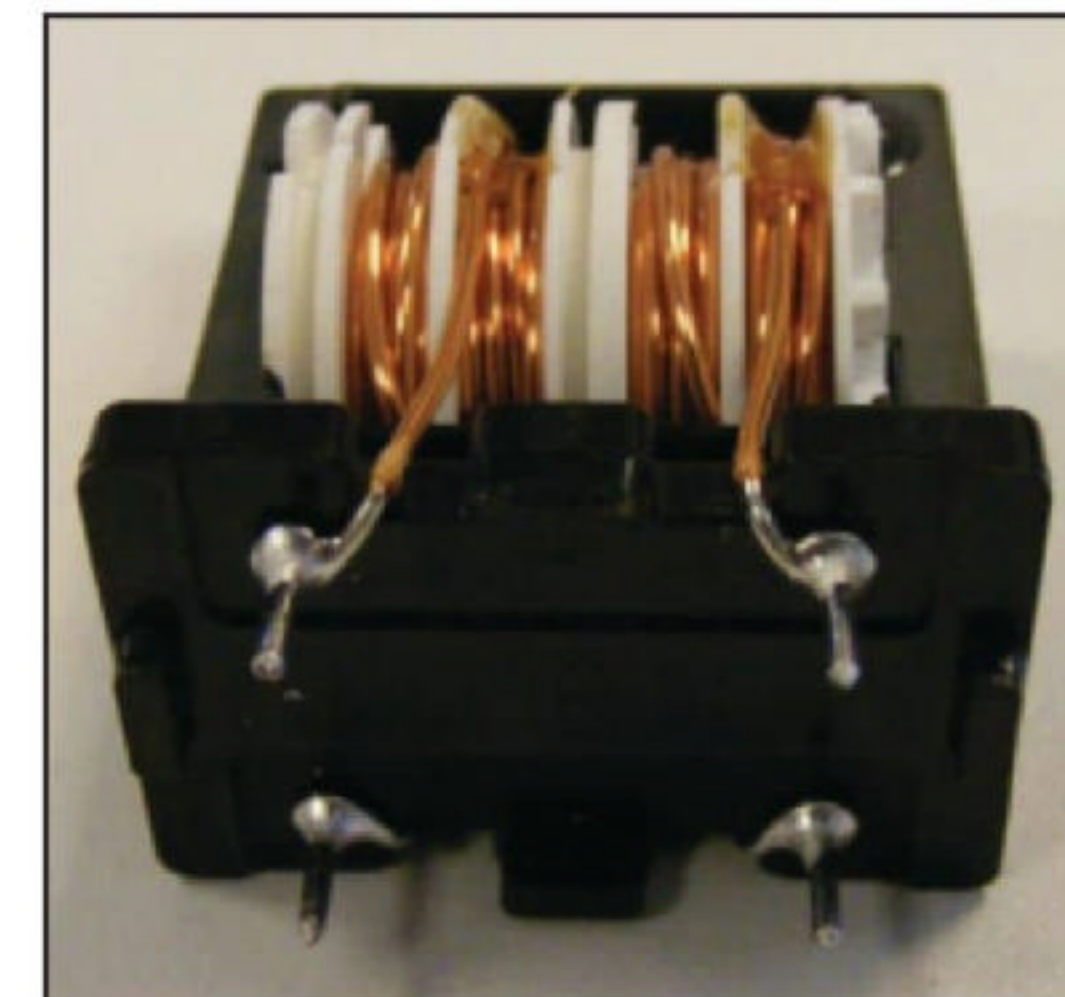


Photo 1: This is a common mode inductor, which could also be used to cancel out noise.

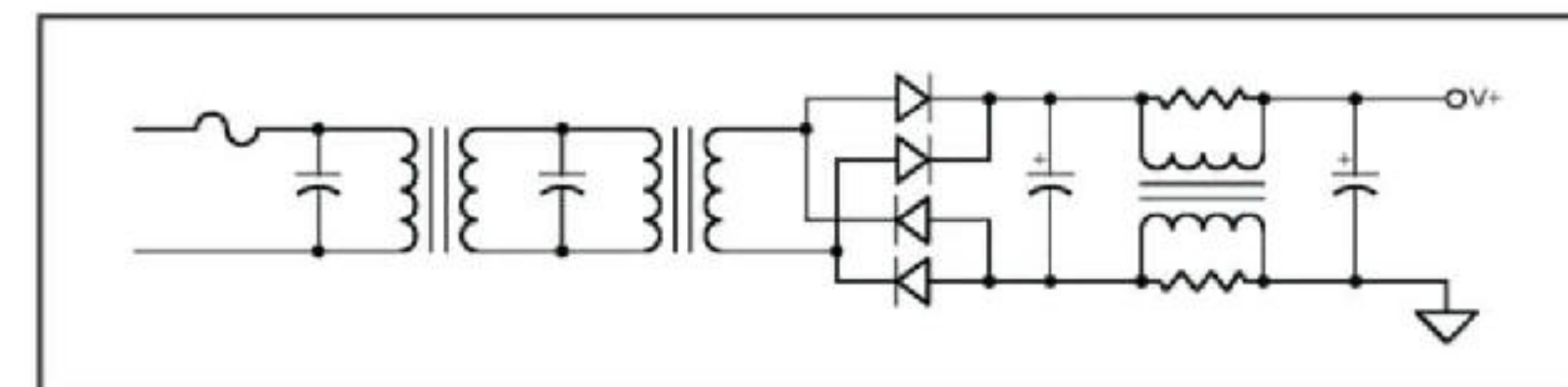


Figure 3: Another way to get noise out is to use two power transformers.

introduce into the power line. A careful user will note that most linear power supplies do not draw current uniformly but as a quick spike. This results in the losses through a filter being three or more times than rated. The second problem is that the filters are designed for a maximum current draw. The filters often do not work as well with a much lighter load. So for example, a Class AB power amplifier, the filtering would not be as effective during soft passages in the music, exactly where filtering would be most needed.

Figure 4: This the schematic for a noise maker relay box.

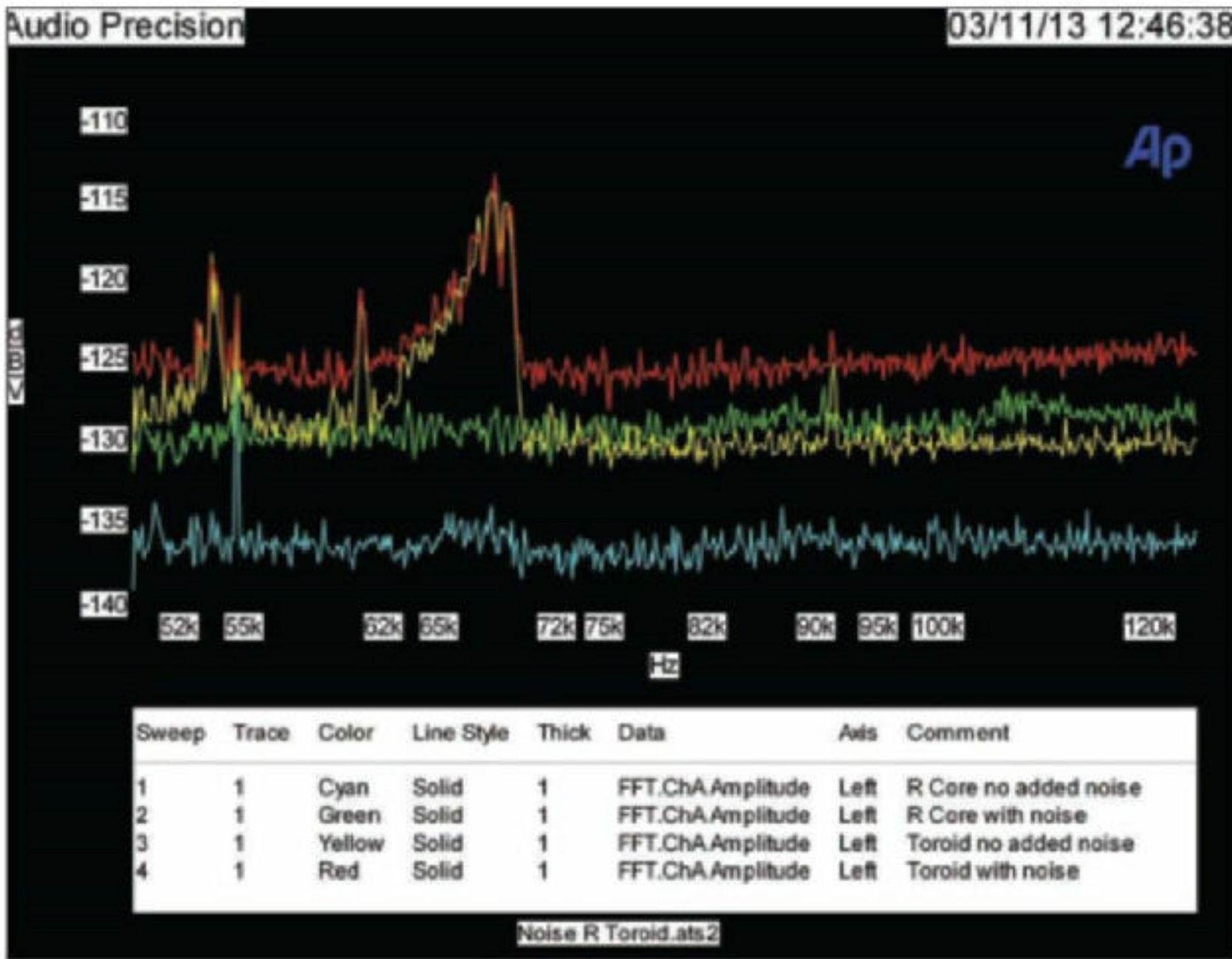
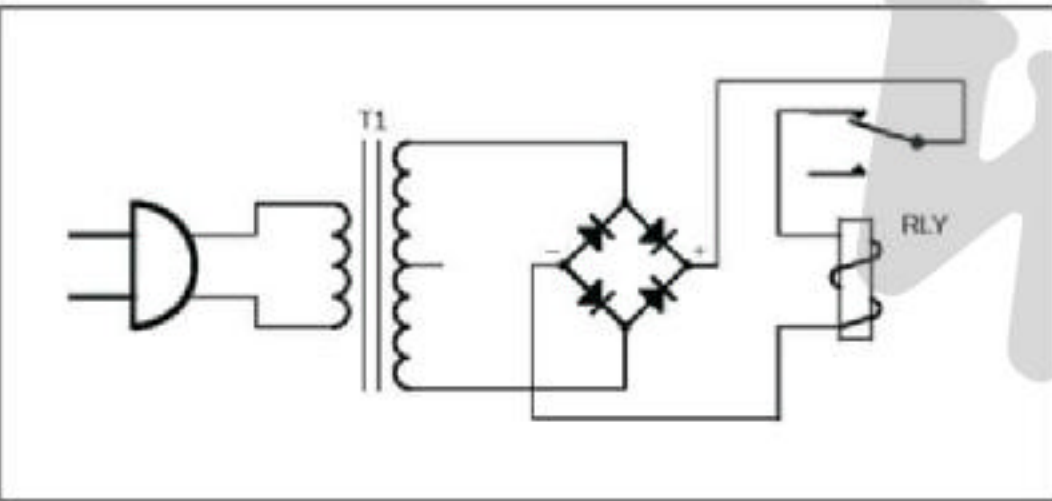


Figure 5: The toroid transformer and the R-core are shown both with and without the added noise source, while looking at the noise just above the audio band.

About the Author

Ed Simon has a BSEE from Carnegie-Mellon University. He was employed by Cameradio, a regional electronics parts distributor from 1973-1975. In 1975, he started Edward Simon & Co. He has published more than 40 articles and taught courses for the NSCA, Carnegie-Mellon University and the University of Pittsburgh among others. His company specializes in sound and communication systems, specializing in systems for public facilities, houses of worship, and allied engineering. The business has completed more than 400 church installations, and 150 other installations including shopping malls, teleconferencing facilities, performing arts centers, television studios, and radio stations.

Inductors in the Filter

Another issue, depending on the filter design, is the possible slow saturation of the inductors used in the filter. This is due to a non-symmetrical AC waveform; this is often called having DC on your AC line.

Now some folks will use a bifilar wound choke on the transformer's secondary even after the rectifier diodes. The idea is that the DC is balanced here and will cancel preventing saturation of the core. Another technique used with inductors is to place a damping resistor across all inductors to prevent self-ringing from the internal capacitance. This resistor is usually sized to carry about 3% to 10% of the normal current flow.

The other trick to get noise out is to use two power transformers (see **Figure 3**). The first may be a 1:1 AC transformer or even have a step down ratio that, when used with the second transformer, results in the desired final voltage. For tube use, this can even be two identical transformers with the secondaries tied to each other and then providing 120 V or even 240 V out.

A useful technique with two transformers is to place a capacitor/inductor filter between them. This not only adds additional filtering but the primary side transformer will prevent the "DC on the AC" from saturating the filter inductors.

AC Power Line Noise

To test for how noise on the AC power line affects us, I made a noise maker relay box (see **Figure 4**). It is quite simple—an E-I transformer steps down the line voltage to 18 V. This then passes through a full-wave bridge rectifier to the normally closed contact of the relay. As soon as power is applied, the relay opens, in turn removing power from the relay coil, which closes the relay, allowing the cycle to begin again. This produced some noise in series and also radiated noise. So we would have both common and differential mode noise.

The circuit shown in **Figure 4**, of course, caused the relay to do nothing more than buzz between closed and open. The energy stored in the relay coil while on had nowhere to go when off, a classic source of noise.

Figure 5 shows the noise just above the audio band for the toroidal and R-core transformers, both with and without the additional noise from the relay box. What we are looking at is the result of a fast Fourier transform (FFT) taken from the output windings of the transformer with a small value blocking capacitor in series with the test equipment input to prevent any 60 Hz input overload.

The R-core was quiet with just the normal

AC line but rose to -129 dB with the noise source powered up. However the R-Core had an inter-winding shield. When this was connected to the chassis and safety ground this became 6 dB quieter. The shield seemed to be useful in reducing common mode noise. This improvement did not actually rely on the power cord safety ground with the high impedance to ground at the test frequencies. None of the other transformers in the group had shields.

The toroid appeared to have a resonance that raised the noise level to -115 dB re 1 V even without any added noise. With the noise it showed a rise with frequency and was the worst power line noise rejection tested.

Due to the difficulty of winding toroids, they almost always use bifilar windings. This means that both primary wires are wound with each other at the same time. A layer of insulation is then wound between the primary and secondary layers. Sometimes a foil shield is also included here. The same winding technique is done with the secondary windings, although in a step down transformer they are of a heavier gauge wire. In some cases the wires are even twisted to each other during the winding process. This insures that both windings are the same length and see as close as possible to each other in the same magnetic field.

Common Transformers

Figure 6 looked at the performance of more common transformers. The PC mounted O-core LP-24-1000 transformer had a noise level of -129 dB the same as the smaller R-core and with a 120 V input winding. This shows the advantages that are produced by the dual bobbin and side-by-side coil windings. (Also note this core had among

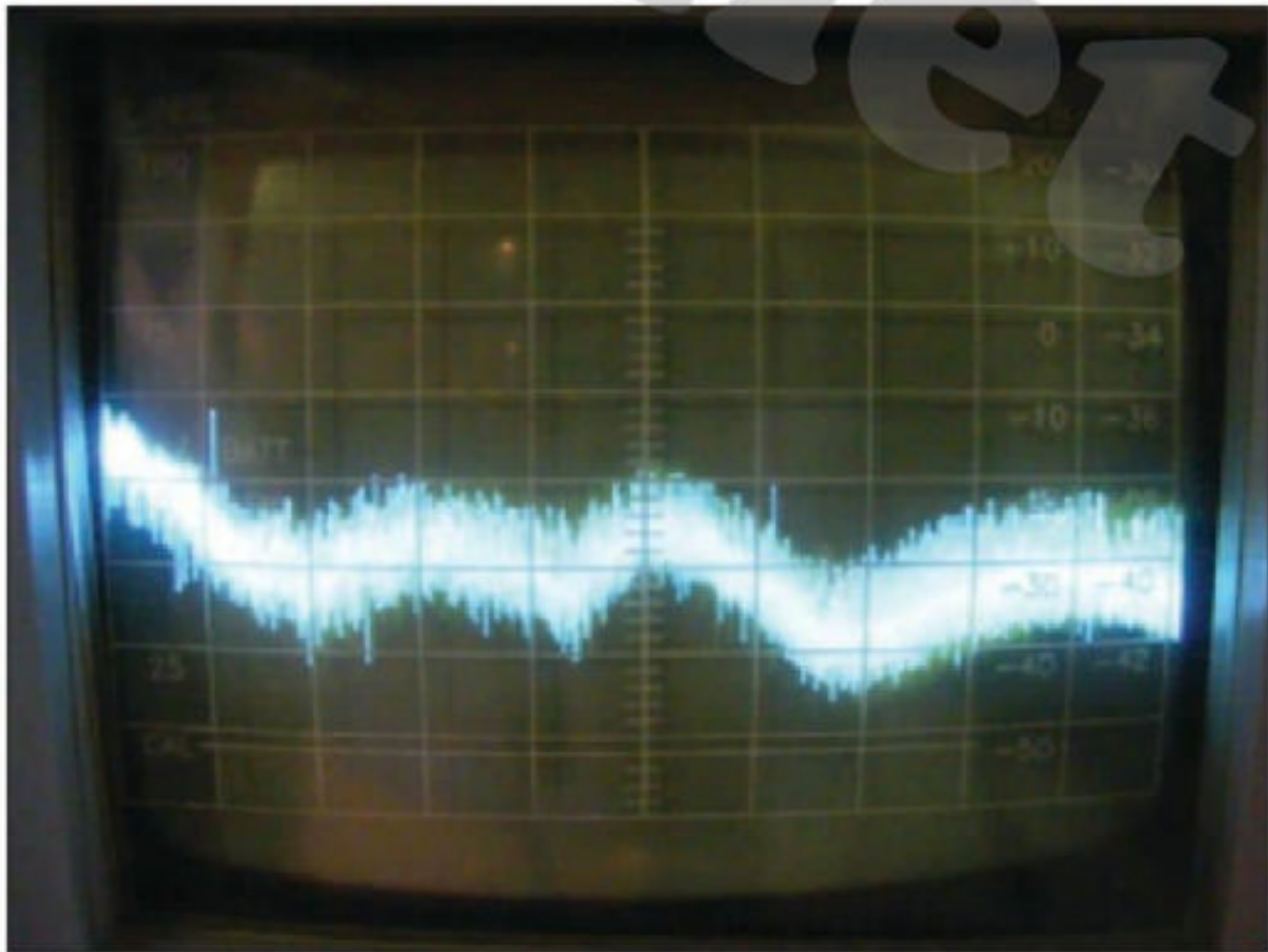


Figure 7: Here is an image of the spectrum analyzer's screen.

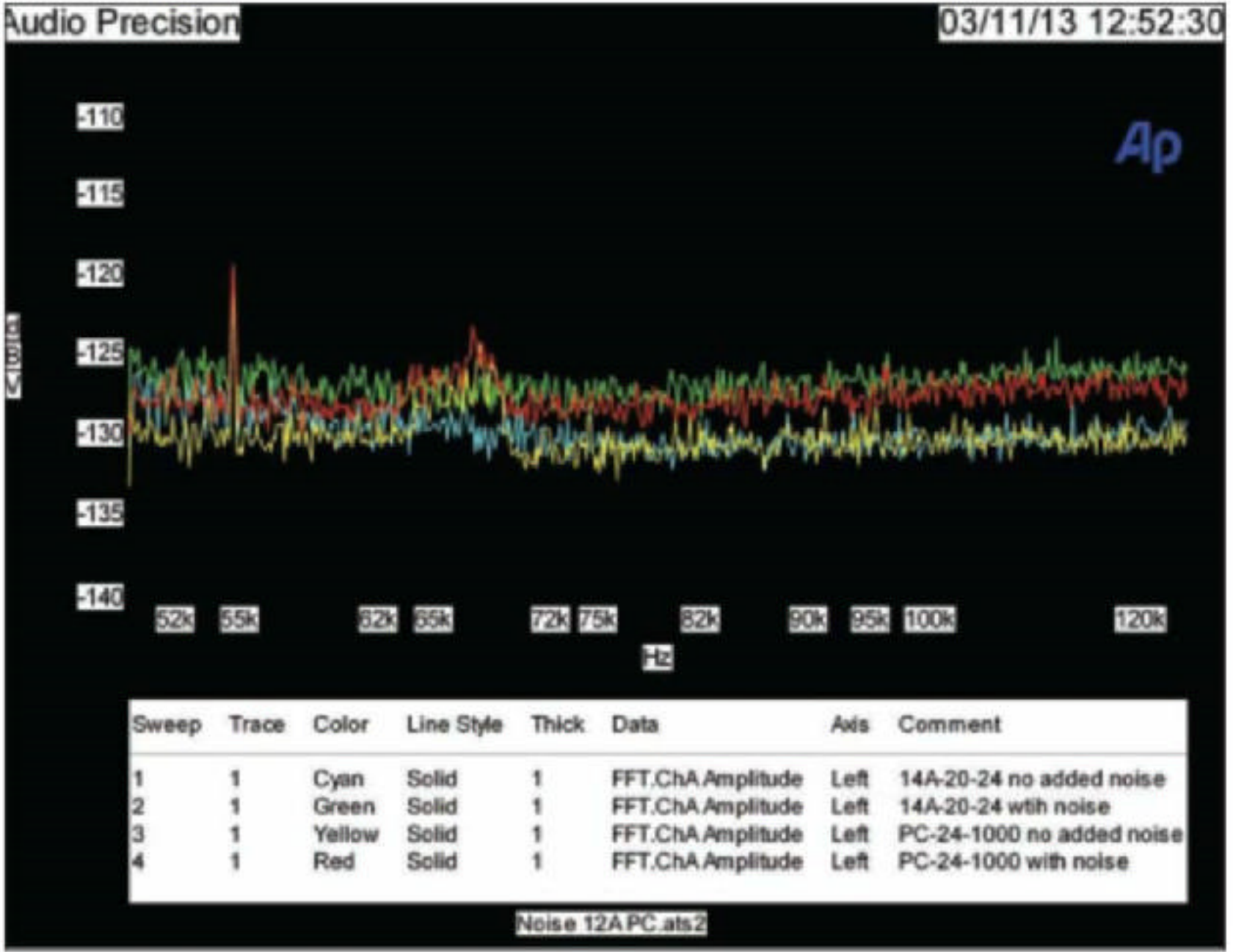


Figure 6: This graph shows the performance of more common transformers.

the highest flux leakage!) The E-I core transformer shown on the same graph in **Figure 6** was just a few decibels noisier. It uses a single side-by-side bobbin.

The RF Spectrum

The next noise issue to look at was the RF spectrum that makes it past a power transformer into the rest of the circuit. Using my old fashioned Spectre 1075 1 GHz swept spectrum analyzer, taking up from what the FFT analyzer showed, 100 kHz to 5 MHz using a short coaxial cable shield to ground and a 0.1 μ F capacitor for a high pass at the

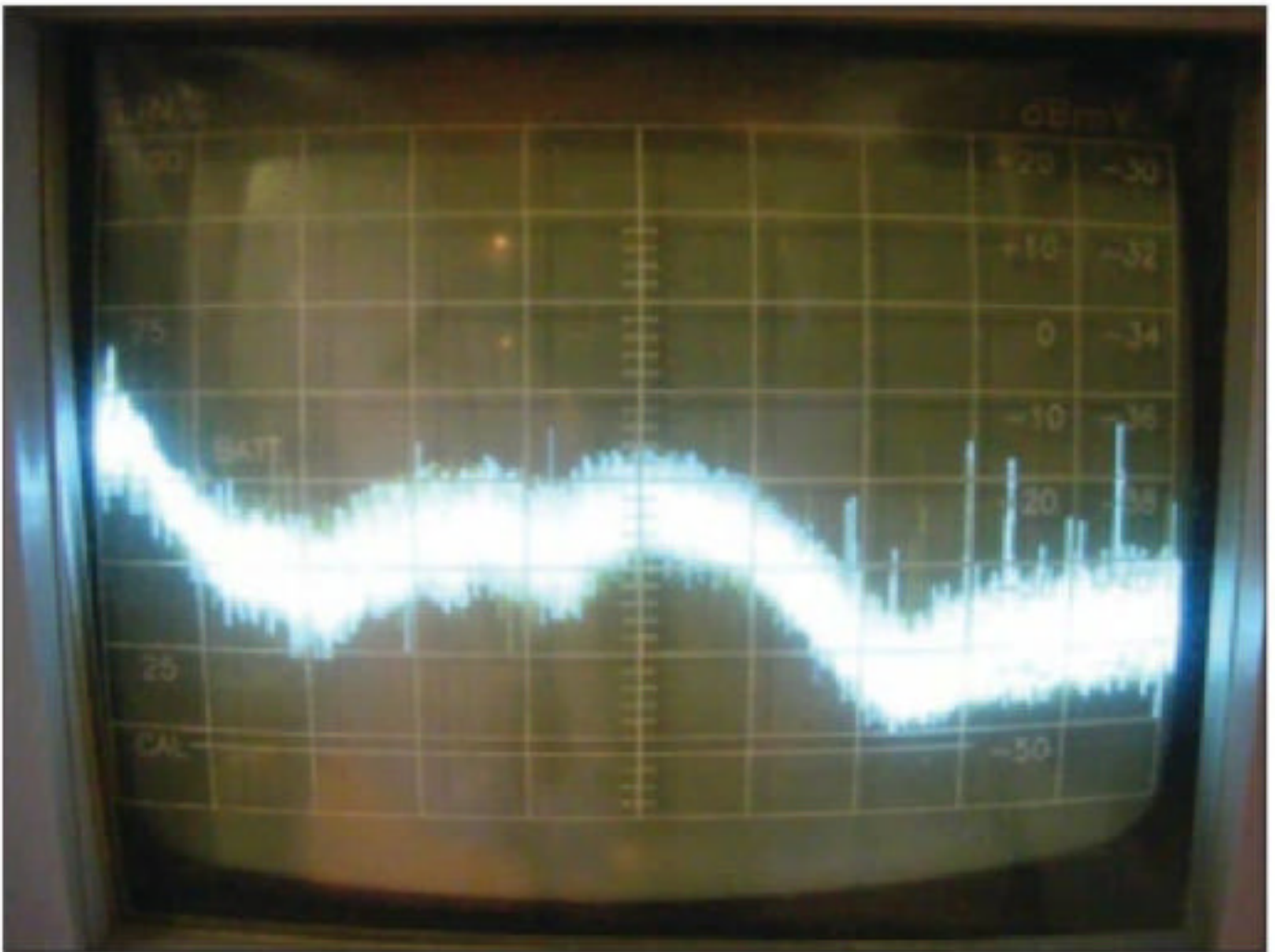


Figure 8: Adding my noise box to the the AC line increased the noise.

75 Ω input, showed a number of interesting issues.

I used my 300 W toroid transformer with dual 12 V windings as a test transformer. The first peak was the RF noise passing through the transformer just from the normal 0.4 Ω source resistance AC line. **Figure 7** shows an image of the spectrum analyzer's screen. As shown in Figure 7, the noise was as high as 11 dB below the reference of 1 mV.

I then used my noise box. Adding this to the AC line showed that this increased the noise. As you can see in the **Figure 8**, the noise was as high as 4 dB below the reference level of 1 mV.

The next step was to make a simple power supply with a center-tapped half-wave bridge rectifier and using a single 4700 μ F capacitor for the filter. This result is shown in **Figure 9**. As you can see, the noise level dropped quite a bit and the diode switch-off

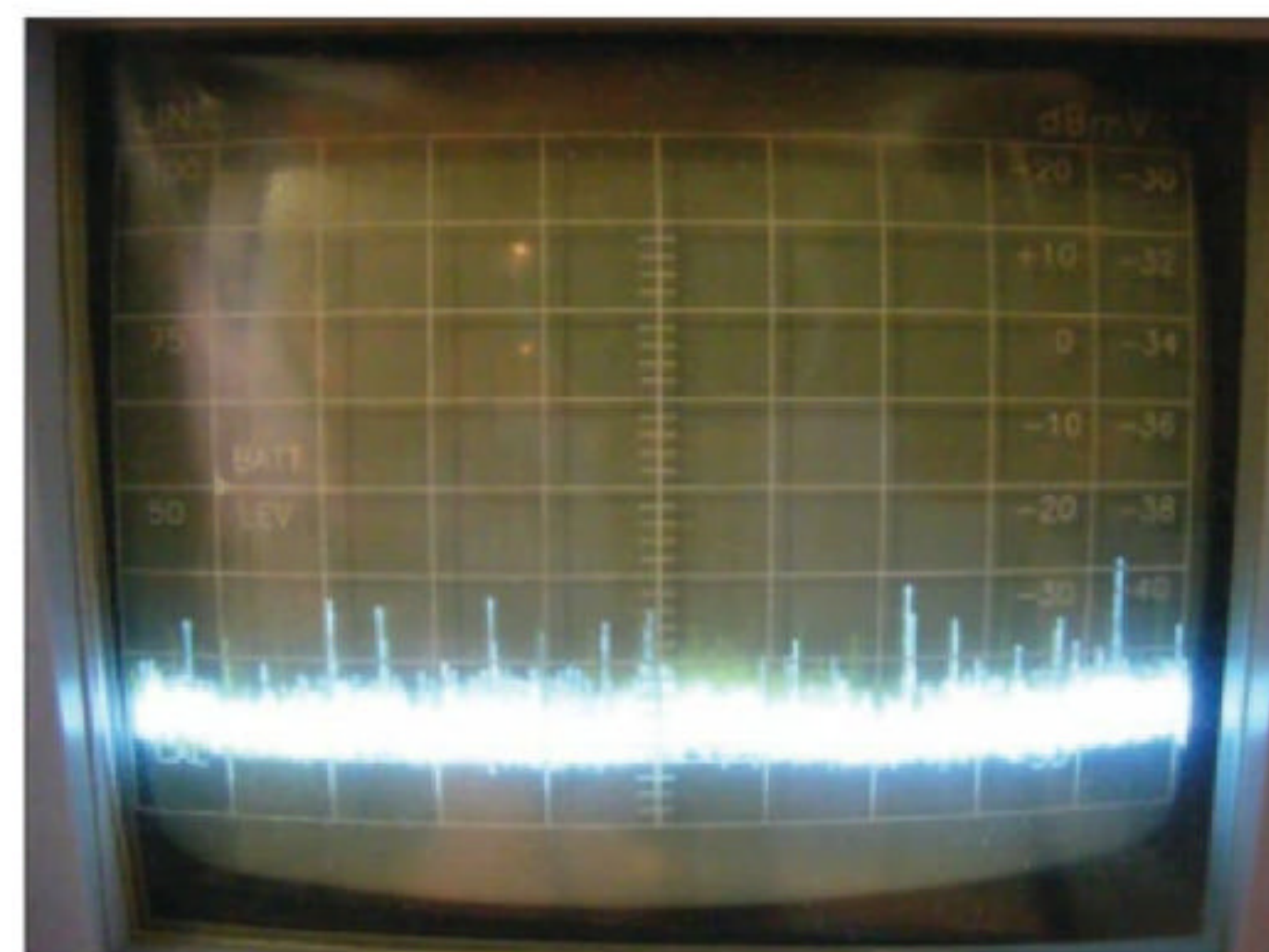


Figure 9: The noise level dropped after I made a simple power supply with a center-tapped half-wave bridge rectifier and used a single 4700 μ F capacitor for the filter.

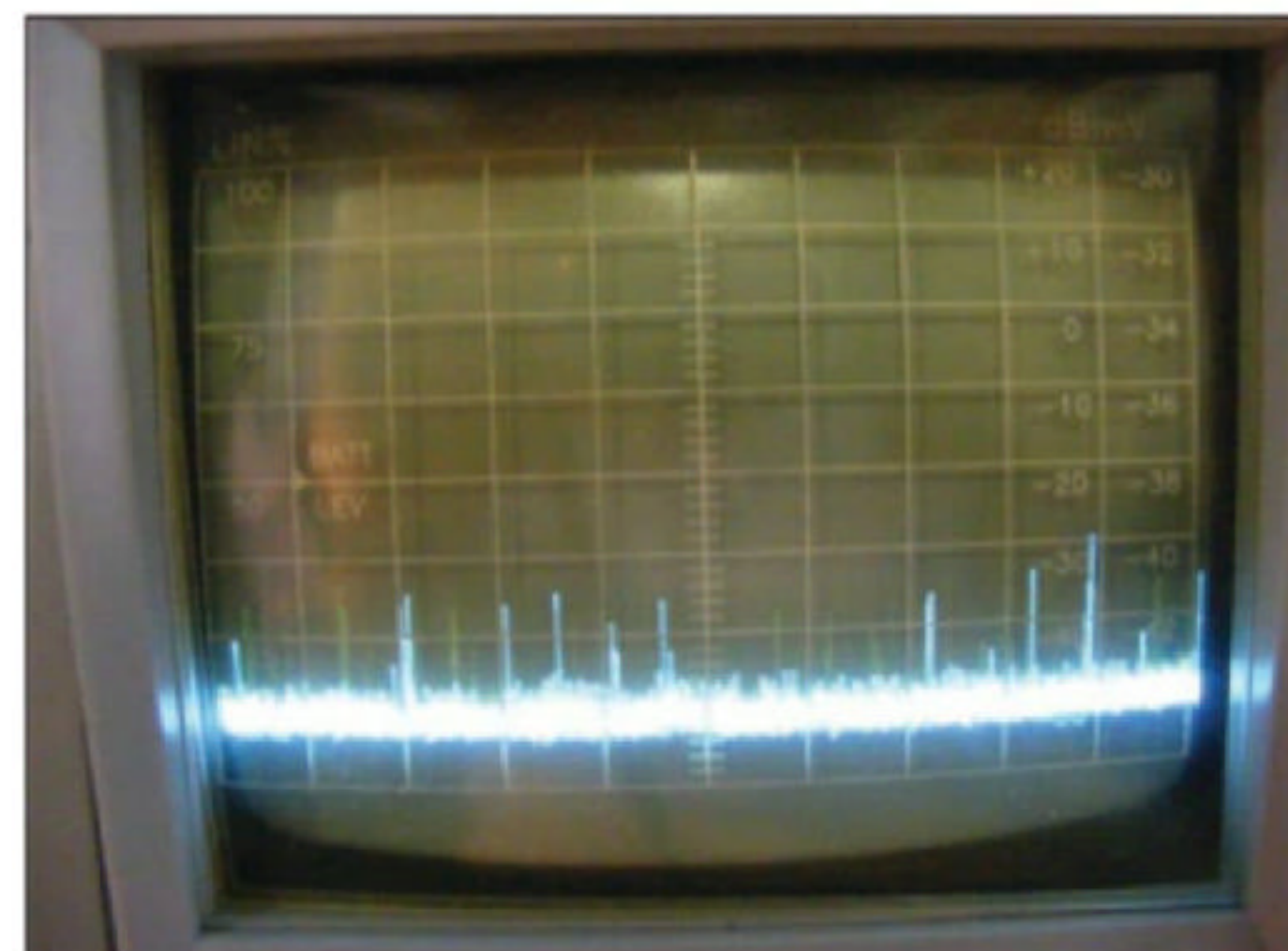


Figure 11: The image shows the results after we placed a small capacitor across the main filter capacitor.

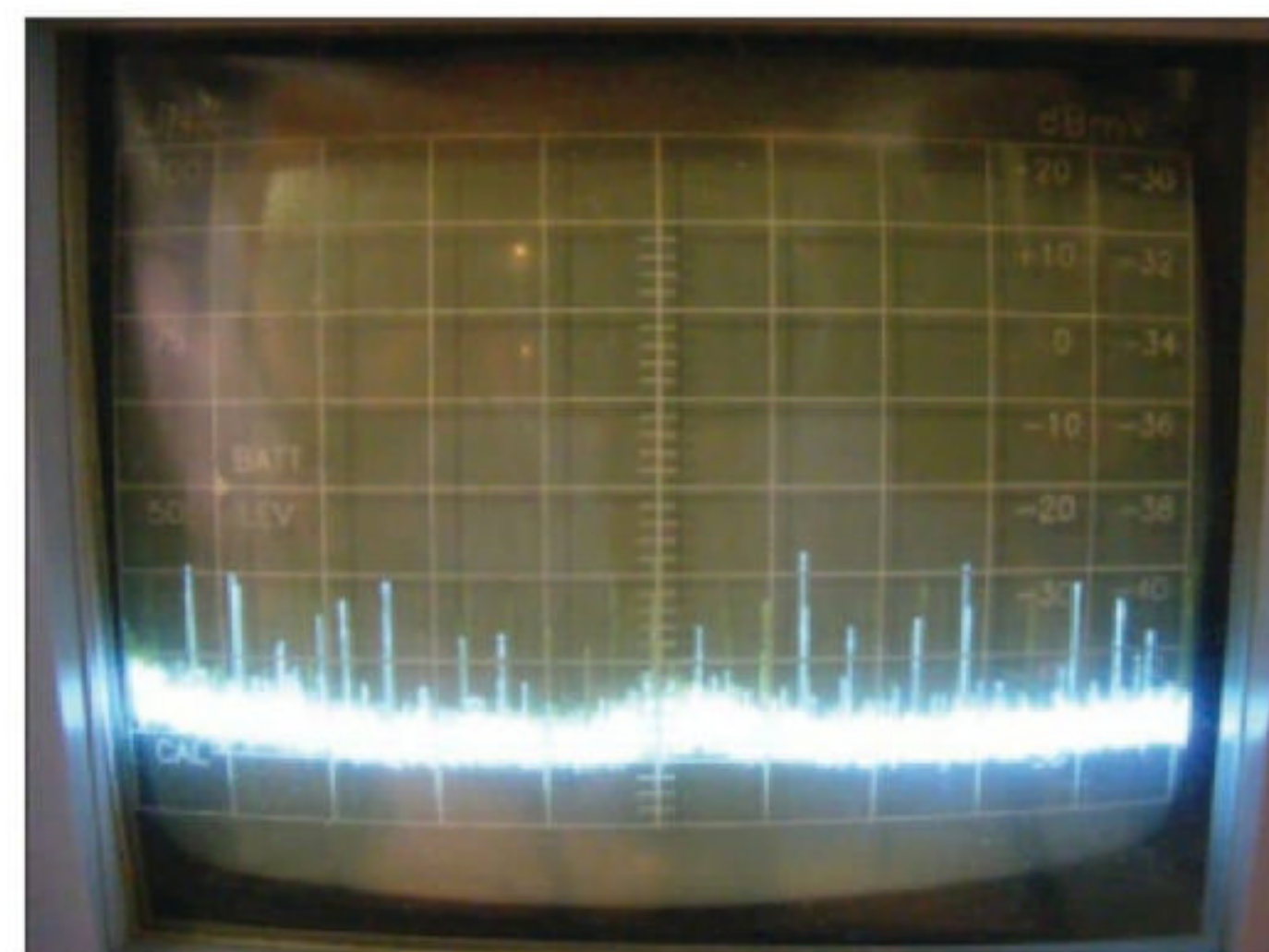


Figure 10: The next circuit we tried was a full-wave bridge, which showed more diode switching spikes but the noise floor dropped.

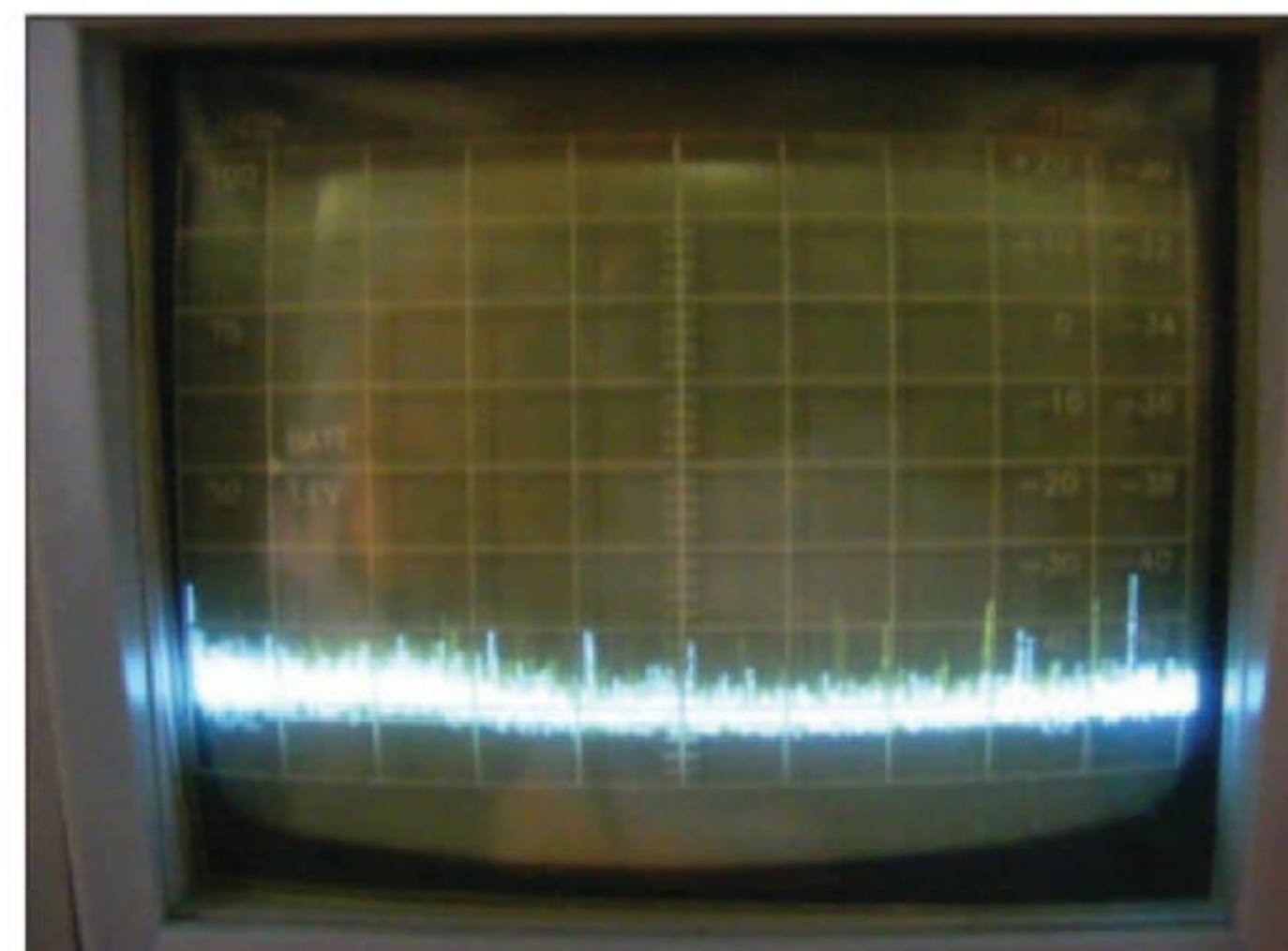


Figure 12: Here, we tried using small capacitors across the rectifier diodes to reduce the switching noise.

noise begins to show up—at some points, it is even above the incoming noise.

Figure 10 shows the next circuit we tried, which was the full-wave bridge. As you can see, there are more diode switching spikes but the noise floor has dropped. That is for two reasons. There is no longer a direct connection from the transformer leakage noise and the diodes in the diode bridge only conduct both power and noise for a fraction of a power cycle.

Next up, we placed a small capacitor across the main filter capacitor to handle frequencies above where the internal inductance of the main filter began to decrease the filtering. I used a 0.01 μ F capacitor for this.

As you can see in **Figure 11**, this lowered the noise a bit and seemed to take out a bit of resonant rise.

	Safety Grid to Chassis		Safety Grid to Center Tap	
	AC Normal	AC Flipped	AC Normal	AC Flipped
R-Core	19.3 μ A	3.3 μ A	18.9 μ A	2.7 μ A
Toroid	8.3 μ A	7.3 μ A	9.6 μ A	17.1 μ A
14A-20-24	1.6 μ A	0.9 μ A	2 μ A	1.2 μ A

Table 1: We compared the current leakage from the AC power line safety ground to both the chassis and to the center tap of the transformer under test secondary.

Next, we tried using small capacitors across the rectifier diodes to reduce the switching noise (see **Figure 12**). As you can see, the diode switching noise has dropped. However the capacitors across the diodes have let in a bit more of the AC Line noise.

Now, one trick to reduce this added noise is to use higher voltage diodes. They will have a greater forward voltage drop along with less capacitance. The second technique is to use a small value capacitor (typical 0.01 μ F) across the output of the power supply. This capacitor is actually best placed not in the power supply but right at the point of load. That way the power supply leads will also act as an inductor and provide a small bit of additional filtering.

Chassis Noise

Now we have looked at the line noise that goes through the transformer, but what about the noise that couples straight into the chassis? I looked at the current leakage from the AC power line safety ground (also called a grounding bond) to both the chassis and to the center tap of the transformer under test secondary (see **Table 1**).

Note it is not just different for each transformer, it also changes when you switch which lead of the primary goes to the AC line hot lead. That is because the primary coil has some windings starting closer to the core and ending away from it. When the AC hot side is connected closer to the core there is more leakage current.

Next month, in the final article in this series, we will focus on power transformer ratings—the all-so-common question of what size transformer is required to power a known load—and we will also discuss fusing and safety. 

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

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Compactrons

By
Richard Honeycutt

(United States)



In this article, our hollow-state columnist provides some “off-label” applications for a unique group of tubes known as Compactrons, which he hopes will provide new options for experimenters in hollow-state audio!

Photo 1: Over the years, tubes of many sizes have been made.

The history of electronics has been summarized as an ongoing quest for miniaturization. **Photo 1** shows some tubes of various sizes that have been manufactured over the years. The very earliest tubes were much larger and looked a lot like light bulbs (see **Photo 2**). I once owned a 1960s vintage H. H. Scott sound level meter (see **Photo 3**) that used subminiature triodes about the size of a jelly bean.

In the last years before transistorization, TVs provided by far the largest market for vacuum

tubes. In 1961, General Electric (GE) introduced a novel way to decrease the size of the tube-based TV chassis: the Compactron. Rather than being a single smaller tube, the Compactron was a device containing two or three vacuum tube structures in the same case (see **Photo 4**). Use was prevalent in televisions because transistors were slow to achieve the high-power and high-frequency capabilities needed particularly in color television sets. In the first portable color television, the General Electric Porta-Color television, 10 of the 13 tubes were Compactrons.

Characteristics

Compactrons had a characteristic 3/4” diameter circular pin pattern dubbed “duodecar.” The envelope diameter of most Compactrons ranged from 1 1/8” to 2 3/4”. Variations of the Compactron design were made by Sylvania and by some Japanese firms. Some Compactron types are:

- 6AG11 double diode similar to the 6AL5, plus double triode high-mu similar to the 12AT7
- 6BK11 triple triode—two are similar to the 12AX7 and one is similar to the 5751
- 6C10 high-mu triple triode
- 6M11 twin triode - pentode
- 6K11 triple triode
- 6LF6 beam power pentode with anode cap
- 8B10 twin triode - twin diode

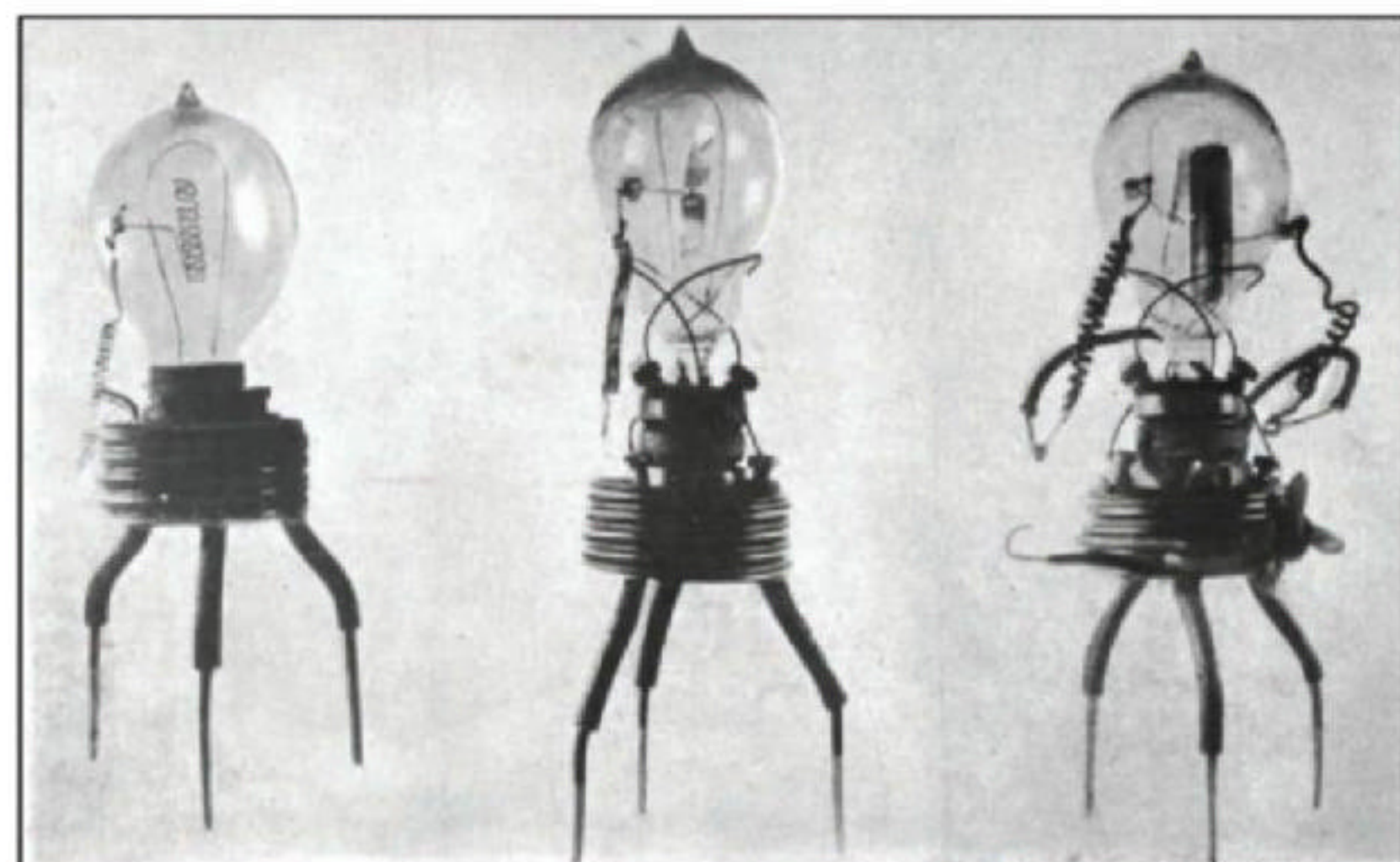


Photo 2: The Fleming Valve, which was the first electron tube (not counting the CRT), looked much like a light bulb and was of a similar size.

- 12AE10 twin pentode
- 38HK7 pentode diode
- 1AD2 high voltage diode

A more comprehensive listing of Compactrons can be found on Jeff Duntemann’s Junkbox website (see Resources). Due to their specific applications in television circuits, many different Compactron types were produced. Almost all were assigned using standard US tube numbers. A few sample models, sorted by basing diagram, include:

12EZ Basing (pentode & beam power tube)

- 6AD10
- 6BF11 plus 12 V, 17 V, and 24 V versions
- 6T10 plus 12 V version
- 6Y10
- 10T10
- 12AE10
- 13V10
- 18AJ10

12BT Basing (pentode and power pentode)

- 6J10 plus 10 V and 13 V versions
- 6Z10 plus 10 V and 13 V versions
- 17AB10
- 17X10

12DZ Basing (triode and power pentode)

- 6JZ8 plus 13 V, 17 V, 24 V, and 25-V versions

As you can see, although there may be quite a few Compactron model numbers, there are relatively few variations of what’s actually in the tube envelope.

Virtually all the power pentodes and beam power sections in Compactrons are limited to plate dissipations under 10 W each—often 6.5 W to 7 W, with a few handling up to 12 W, meaning that for Class-A amps, you can only get about 2.5 W (typical) output: absolute maximum of 4 W. Class AB will increase that to a theoretical maximum of about 40 W, but you’ll find it challenging to get more than 20 W. (The highest-power example given in the GE datasheet for two 6BQ5s running in Class-AB₁ is 17 W, using 300V B+. The specs for the beam power section of the 6D10 Compactron are quite similar to those of a 6BQ5, so the 6D10 should work in the GE circuit. (See the circuit description at <https://frank.pocnet.net/sheets/093/6/6BQ5.pdf>.)

Compactrons Current Status

By the mid-1980s, Compactrons were considered obsolete. Manufacture of Compactrons was discontinued in the early 1990s. New old stock (NOS) replacements for almost all Compactron

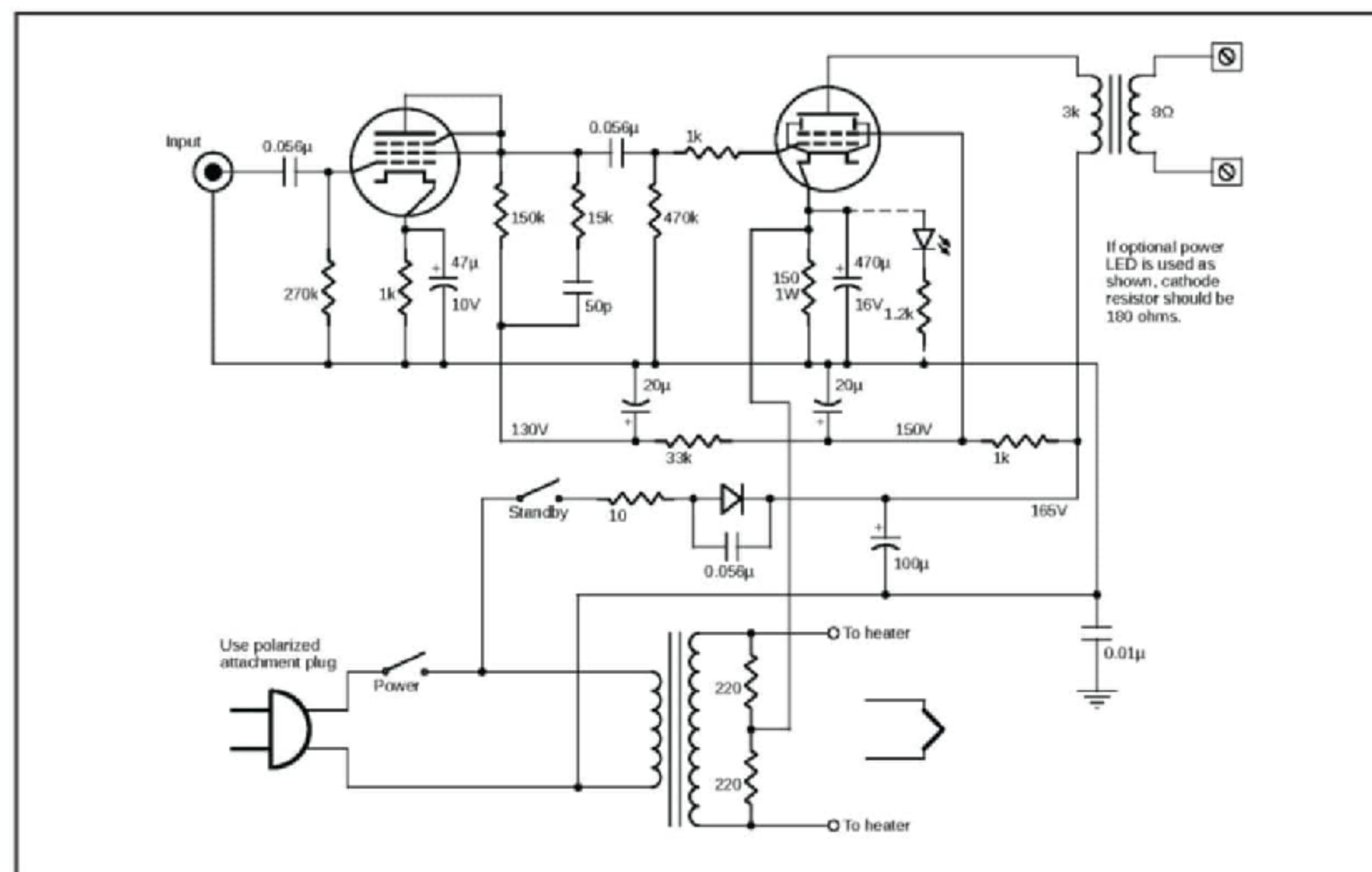


Photo 3: This H. H. Scott sound level meter used jelly-bean-sized triodes.



Photo 4: The Compactron achieved miniaturization by combining multiple tube functions within a single envelope.

Figure 1: This 2 W power amplifier from Wade's Audio Page uses a single Compactron.



types can be found for sale on the Internet.

The Compactrons' integrated design helped decrease power consumption and heat generation, which in itself allowed more compact chassis to be used. Compactrons were also used in a few high-end hi-fi stereos. They were also used by the Ampeg guitar amplifier company in some of their guitar amps. The only modern tube-based home-stereo system known to use this tube type is Linear Tube

Audio's 20W Ultralinear, which uses four 17JN6 Compactron tubes as the power tubes in the amp. Except for the 17 V filament rating, the 17JN6 is basically a 6JN6 beam power tube designed as a horizontal deflection amplifier for TV sets.

A few tube audio enthusiasts have successfully experimented with using Compactrons for audio amplifiers. Wade's Audio and Tube page (www.angelfire.com/vt/audio/compactron.html)

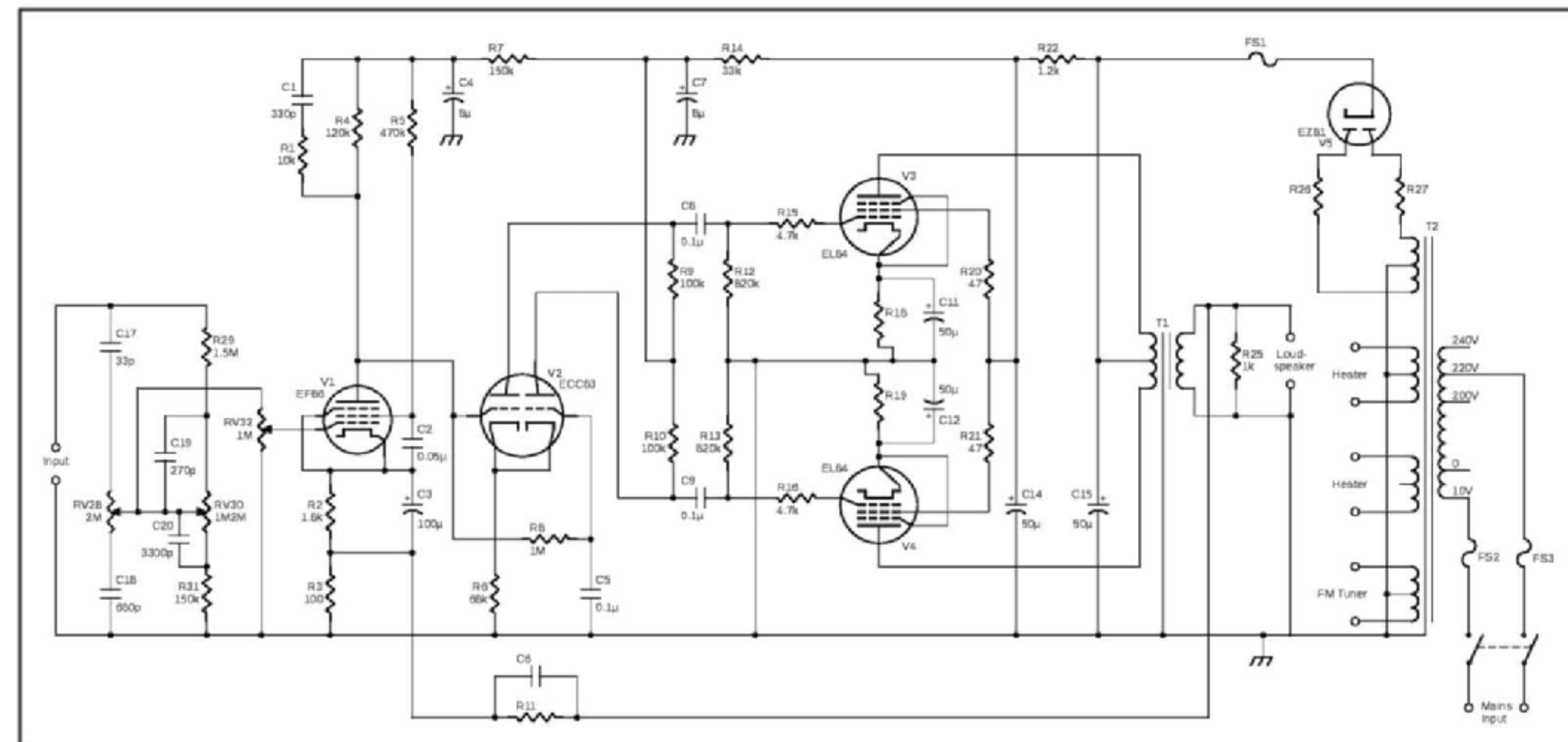


Figure 2: This circuit by Mullard can be built using two 6T9 Compactrons and one EF86 pentode.

discusses a single-ended power amplifier using a Compactron containing a pentode and a beam power tube (see **Figure 1**). The designer's intentions seem to have included using available free parts and keeping things simple. In this case, "simple" means no B+ power transformer (only a filament transformer), and minimum parts count. He mentions that, even with the low power output, it sounds better than circuits using the LM386, a DIP IC power amplifier.

If we use a 6AD10, the beam power tube section seems to perform very similarly to a 6BQ5, having a maximum plate dissipation of 12 W and a transconductance of 6500 μ S. Using data for a 6BQ5, we find that we should be able to get at least 10 W output from a push-pull pair of the beam power sections of two 6AD10 Compactrons. The output transformer would need to be 8,000 Ω plate-to-plate, and the B+ and screen grid supplies would need to be 250 V. Maximum-signal plate current would be 79 mA; and maximum screen-grid current, 13 mA. Peak grid-to-grid signal voltage to achieve maximum output would be 30 V.

The pentode of the second section is used as a triode in this circuit. Another Compactron that is useful in audio amplifiers is the 6T9, which is a power pentode and a triode in the same envelope.

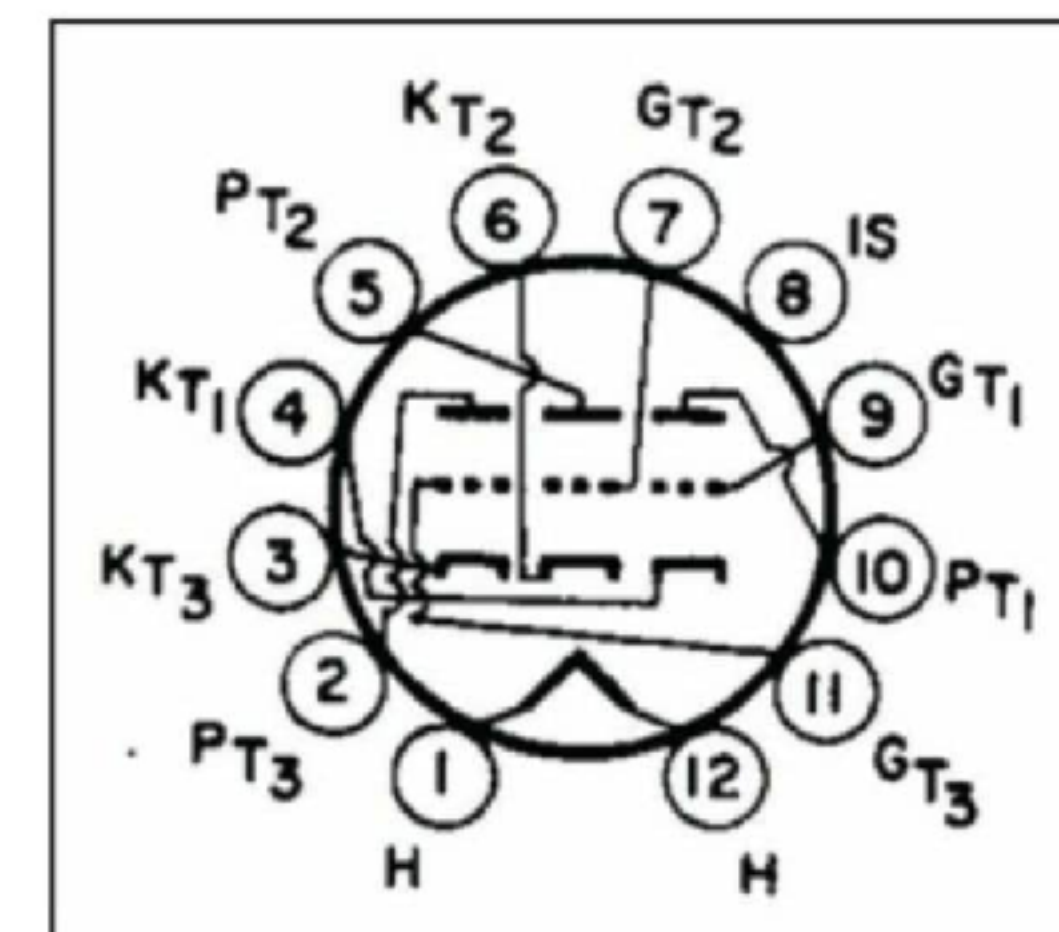


Figure 3: This is the 12BY basing diagram of the 6K11 triple triode Compactron.

Even though the power section of the 6T9 is a power pentode, and that of the 6AD10 is a beam power tube, both are similar in performance to the 6AQ5, 6BQ5, 6V6GT, and EL84 tubes. The triode section is comparable to the 12AX7, but with a μ of 95 rather than 100—an insignificant difference.

Figure 2 shows a 10 W audio amplifier circuit by Mullard that can be built using fewer tubes if a pair of 6T9s is substituted for the EL84 output tubes and the long-tailed pair phase splitter. (The 6T9's datasheet can be found at: <http://frank.pocnet.net/sheets/201/6/6T9.pdf>). The

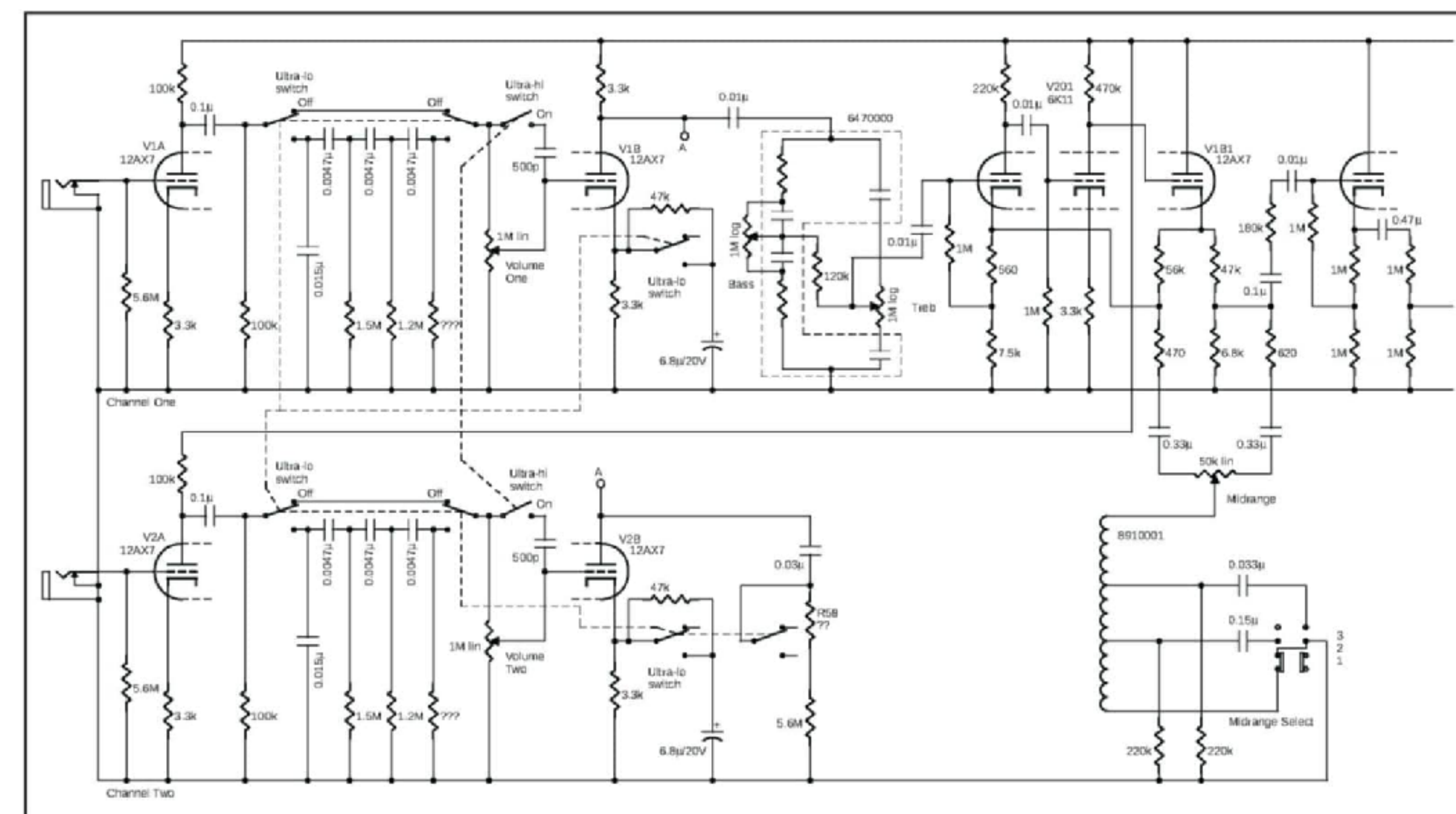


Figure 4: The 6K11 triple triode Compactron was used in the active tone control of certain Ampeg amplifiers.

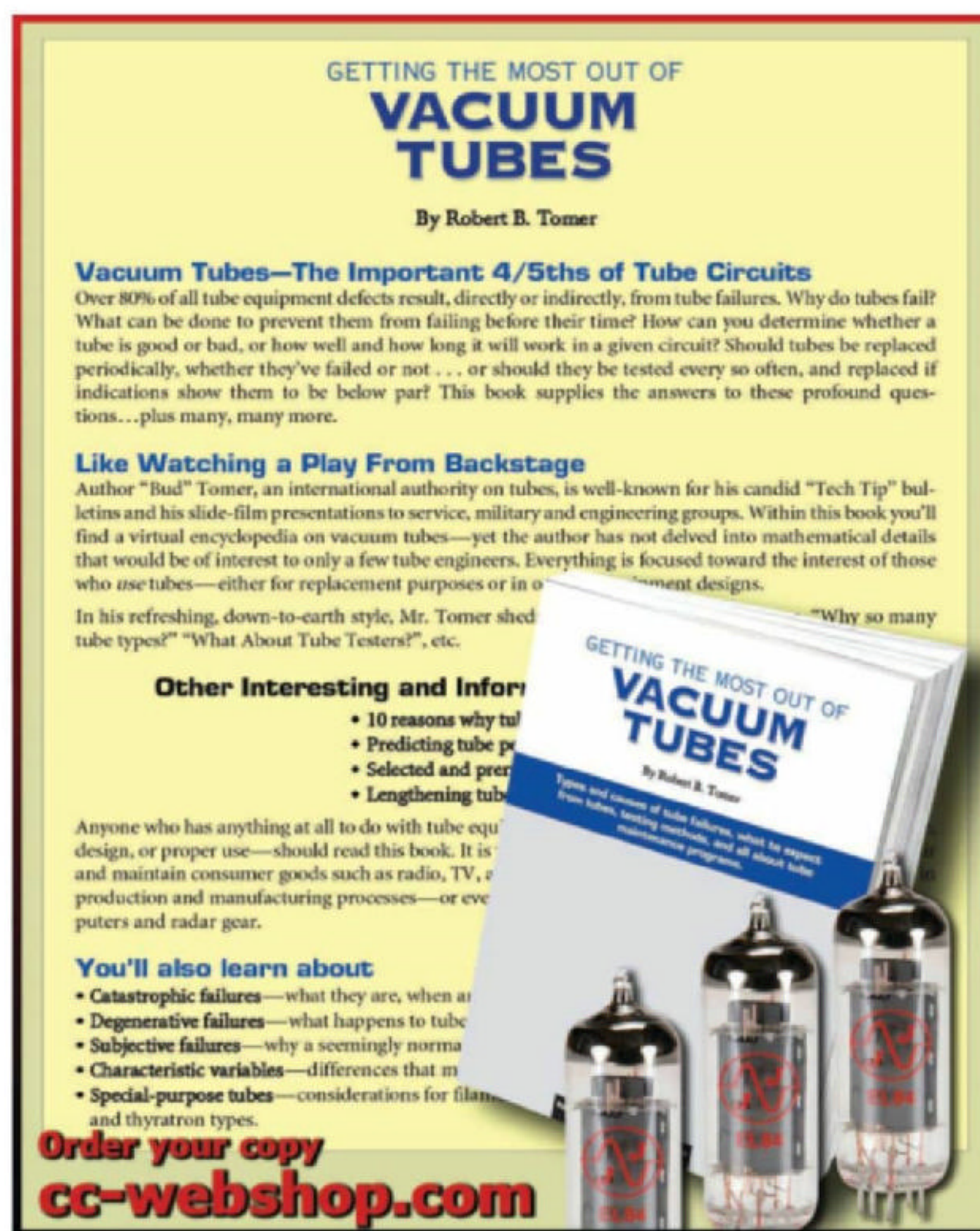
EF86 pentode will be needed as well, plus the EZ81 full-wave vacuum rectifier. Or a solid-state rectifier could be substituted, in which case, for maximum tube life, you'd want to make sure the 275 V maximum plate voltage for the power pentode is not exceeded, since less voltage is dropped across a solid-state rectifier than across a hollow-state one. Actually, tubes are very forgiving of small (<10% to 15%) increases of plate voltage above the maximum, as long as the maximum plate power dissipation is not exceeded.

Resources

J. Duntemann, "Compactron Tubes: Pinouts, Circuits, Projects and Tips," Copperwood Media, LLC, www.junkbox.com/electronics/CompactronTubesIndex.shtml.

Mullard Circuits for Audio Amplifiers, by the Technical Services Department of Mullard, Ltd., London, England, 1965, <https://cc-webshop.com/collections/books-amplifier-related/products/mullard-tube-circuits-for-audio-amplifiers>

Wade's Audio Page, www.angelfire.com/vt/audio/compactron.gif



RV28, RV30, and RV32 are the treble, bass, and volume controls, respectively, and should be audio-taper units. R18 and R19 should be 270 Ω , 5%, 3 W units. All capacitors should be rated at 350 V, except for C3 (12 V), and C11 and C12 (25 V). The original circuit (see the Mullard book listed in Resources), included B+ feeds for a specific tuner and preamp, plus suggested power transformer models that presumably were available in the UK in 1965. Who knows whether they are available today!

If you want to try your hand at designing audio circuits using Compactrons, and you plan to use a SPICE simulator, you'll probably have trouble finding models to use. The workaround is to find "standard" tubes that are similar to the units in each section of the Compactron, as mentioned in connection with the 6AD10 and the 6T9 Compactrons discussed earlier. The output section of both of these tubes is similar to a 6BQ5, and the triode of the 6T9 is very similar to a 12AX7.

When finding comparable tubes, look at the maximum ratings and the plate resistances, plus the μ of triodes or the transconductance of pentodes. For useful comparisons, make sure the plate resistance values you compare were measured at the same plate voltages for both the Compactron and the "standard" tube. Also, be warned that not all the vacuum-tube SPICE models you find on the Internet are good. Sometimes a simulation will be halted, and you'll get an error message; then if you try the same circuit using a different model for one or more of the tubes, the simulation will run perfectly.

When the time comes to build your circuit, you may wonder about the availability of the 12-pin duodecar sockets they use. The good news is that even if your junk box has none of these, they are available at reasonable prices from a number of Internet vendors.

Aside from experimenter/DIY projects, there is one particular repair issue regarding Compactrons that you may encounter. The "Magnavox-era" (1971-1980) Ampeg V4, VT22, and V4B musical instrument amplifiers used 6K11 Compactrons (see **Figure 3** and **Figure 4**) in their active tone controls. These were triple triodes: unit 1 had a μ of 71, and units 2 and 3 had a μ of 100. These tubes became unavailable for a few years, but can now be found on the Internet. Fortunately, they very seldom failed. In fact, I don't think I ever had to replace but one in all my years of servicing amps.

I hope this survey of "off-label" applications for an oddball group of tubes has widened options for experimenters in hollow-state audio!



ELECTRONIC TRANSFORMERS

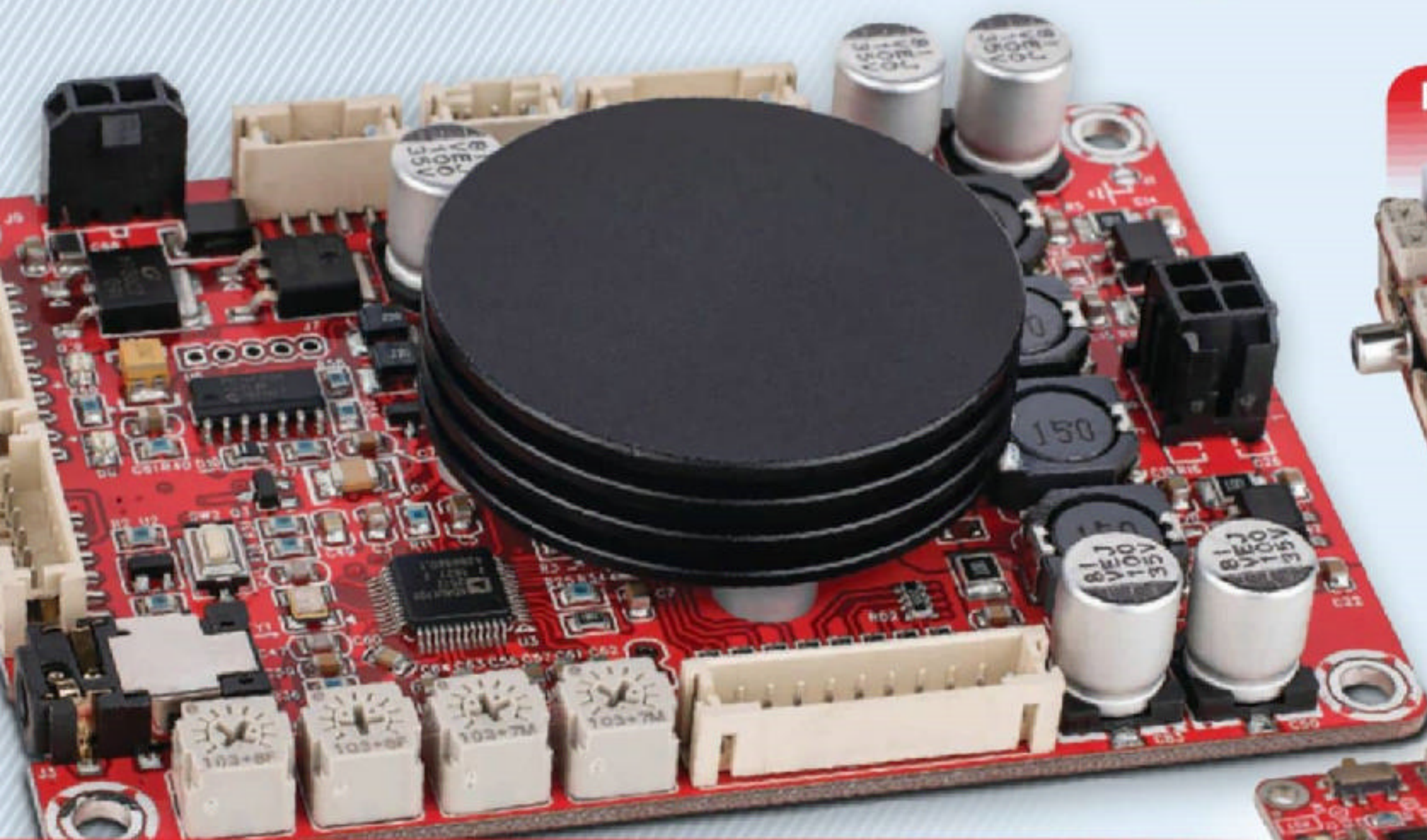
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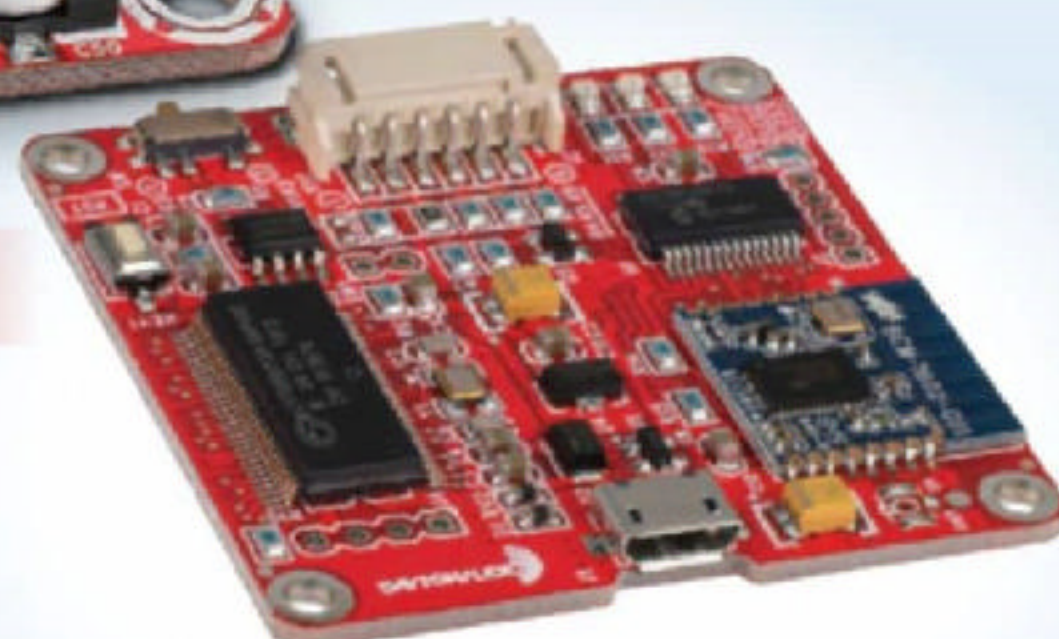
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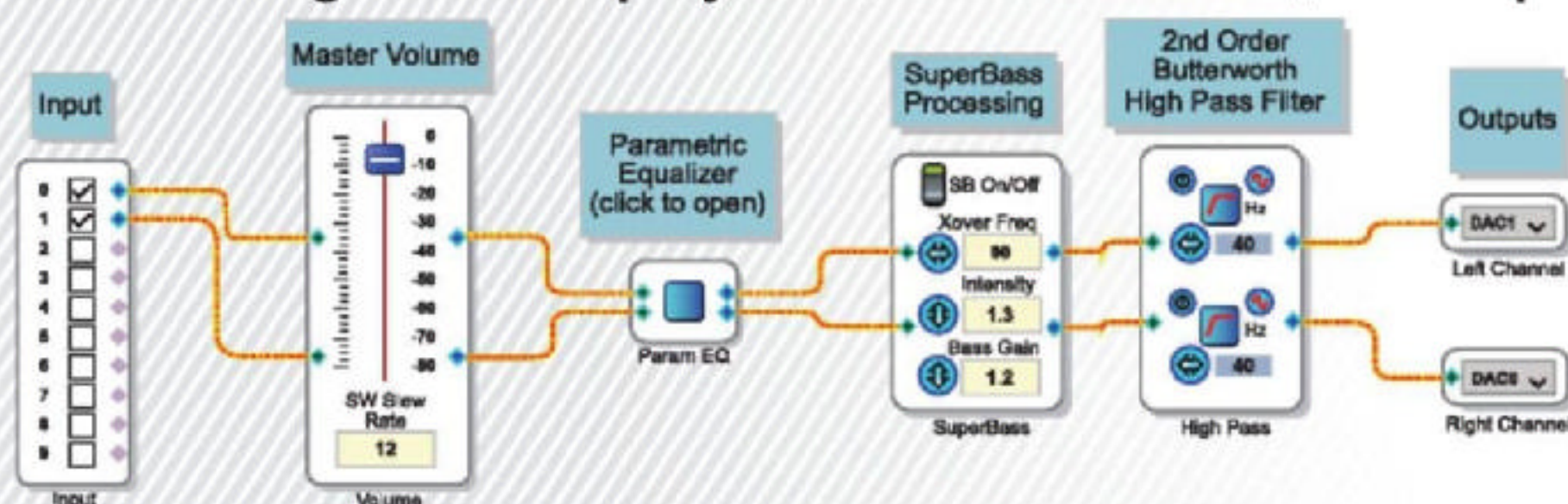
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