

# audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

## Market Update

### State of the Hearables Industry

## Market Update

### Headphones Test and Measurement

By J. Martins

## Simulation Techniques

### Acoustic Topology Optimization for Vibroacoustics Applications

By René Christensen

## Audio Electronics

### Vintage Op-Amp Noise Simulation Part 1 — Deviation from Realistic Noise Figures in Datasheets

By Burkhard Vogel

## You Can DIY!

### Building a Soundbase

By Ken Bird

## Book Review

### Remote Recording: David Hewitt's *On the Road* By Scott Dorsey

## Sound Control

### Wrong Things “Everybody Knows” About Acoustics By Richard Honeycutt

## You Can DIY!

### Monoblock Power Amplifier with Precision Bias Control Part 1 — The Design By Bruce E. Gillingham



# ANATOMY OF A 300B

PRESENTED BY *Western Electric*

New, patent-pending graphene-coated anode improves thermal emissivity, electron affinity, and eliminates secondary emission

Molybdenum grid creates negative coefficient of resistance

Ultra-high vacuum finishing at  $2 \times 10^{-7}$  Torr, made possible with state-of-the-art turbomolecular pumping systems

Proprietary mica coating increases dielectric strength, optimizes higher voltage operation

Filament core's proprietary oxide coating allows for 40,000 hour lifetime and superb impulse response

Quality guaranteed with a five-year extended warranty



There's more to it than cathode, anode, and grid. Each Western Electric 300B is assembled at our new facility by the hands of our highly trained staff using the highest quality materials and modern technology—all while honoring the original 1938 specifications and tooling.

What's the result? What's in it for the listener? Unmatched manufacturing quality and the most powerful, full-bodied sound there is. When you're through compromising, order a matched pair online or from Authorized Dealers worldwide.

## AVAILABLE NOW

[westernelectric.com/300b](http://westernelectric.com/300b)



# MANUFACTURING / DESIGNING TRANSFORMERS & ENCLOSURES SINCE 1917





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

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

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

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

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

Advertising rates and terms available on request.

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

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



## New Frontiers in Wireless

Often I find myself explaining why I remain optimistic and enthusiastic about what we do at *audioXpress* and the audio industry overall, particularly to those who insist that the “good old, golden days” are behind us and/or that “nothing is as it used to be.” At the beginning of the year when I returned from CES full of energy and inspiration, I repeatedly heard the same statements, “...but, there were hardly any audio companies left at CES!”

And that’s the reason why I explain—and strongly recommend—not to limit one’s perspectives. That’s why I need to get out there, and follow the many professionals in the audio industry who are thriving and living the “golden days” while others are just starting. Yes, nothing stays the same. Things evolve and in a world of technology, that’s precisely what you would expect.

The audio industry is currently as vibrant as it ever was, and in some application segments bigger than it ever was, attracting and employing more people than ever, and evolving with some of the best technology we have ever had.

That’s part of our mission, to share the multitude of perspectives that make audio technology always exciting and vibrant. We cannot restrict ourselves to one single—often misrepresented—perspective.

It’s not always obvious. Sometimes things are not in plain sight. The most important takeaway from CES regarding home entertainment has to do with wireless. At CES 2023, Xperi, in cooperation with Philips and LG, promoted the (re)launch of DTS Play-Fi supporting whole-home audio streaming and Dolby Atmos wirelessly over standard Wi-Fi. And LG was basically promoting an “everything wireless” future, showcasing the world’s first consumer TV with Zero Connect technology—a wireless solution capable of real-time video and audio transmission at up to 4K 120Hz.

Yes, it’s unfortunate that we needed to attend another show—ISE 2023 in Barcelona—to actually see two halls with half the area of the LVCC full of audio companies showcasing quality audio. It was at ISE that I saw finally a working demo of a Dolby Atmos system using the Wireless Speaker and Audio Association (WiSA) technology. And it wasn’t even from WiSA Technologies but in a very small booth from Hansong and Platin Audio, demonstrating the Platin Monaco 5.1.2 speaker system, which is a WiSA HT Certified, Dolby Atmos enabled, shipping product. The best thing in the world? No, but great to see the realization of the promise of WiSA supporting Dolby Atmos and also great to see that in Europe there is an increasing number of great companies, led by Bang & Olufsen, enthusiastically embracing WiSA technology, while in the US WiSA Technologies struggles to stay afloat in the stock market even though the company is riding the wave of the future of audio.

Future of audio? Yes, I truly believe this is the last frontier to reconnect hi-fidelity and home audio with the consumer’s evolving lifestyle. It’s obvious and right in front of everyone.

Bluetooth speakers, Smart speakers are everywhere. And yet, neither is even close to be the future. The latest generations of TVs and soundbars are all now streaming audio wirelessly but users don’t know how to configure 10% of the features.

We need to evolve “smart” technologies that deliver on the promise of higher-quality, high-resolution, low latency, immersive, simpler-than-cables audio.

Later this month, I’m meeting with some companies that are now working on what we can do with Wi-Fi 7 for better home audio, while others are dreaming about upcoming 6G cellular networks streaming spatial audio directly to our home speakers. Yes, for real!

At CES 2023, I also had the chance to spend some time with MQA and Bob Stuart and hear for myself the latest SCL6 scalable codec, created specifically for high-quality wireless audio and efficiently scaling from below 200kbps to 20Mbps connectivity. A technology that is ready to be deployed and break the compromises of wireless personal audio, taking off where Bluetooth stalled.

And finally, it was at CES this year that we gained the insight needed to write this issue’s Market Update on True Wireless Earbuds evolving to Hearables. An updated perspective on the many reasons why working in audio technologies has never been more exciting... and a market worth \$14 billion USD in 2023.

J. Martins  
Editor-in-Chief

### The Team

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

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

# OUR NETWORK



# SUPPORTING COMPANIES

|                             |       |                                   |    |
|-----------------------------|-------|-----------------------------------|----|
| ACO Pacific, Inc.           | 53    | Menlo Scientific, Ltd.            | 27 |
| COMSOL, Inc.                | 9     | Parts Express International, Inc. | 68 |
| DSP Concepts                | 23    | QuantAsylum USA LLC               | 45 |
| GRAS Sound & Vibration      | 30-31 | Redco Audio                       | 53 |
| Hammond Manufacturing, Ltd. | 3     | SB Acoustics                      | 25 |
| HEAD acoustics              | 13    | Spectral Measurement              | 55 |
| InfoComm International 2023 | 19    | Triad Magnetics                   | 67 |
| KAB Electro-Acoustics       | 65    | Western Electric                  | 2  |
| Lone Star Audio Fest 2023   | 43    |                                   |    |

## NOT A SUPPORTING COMPANY YET?

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

# The Audio Voice

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



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



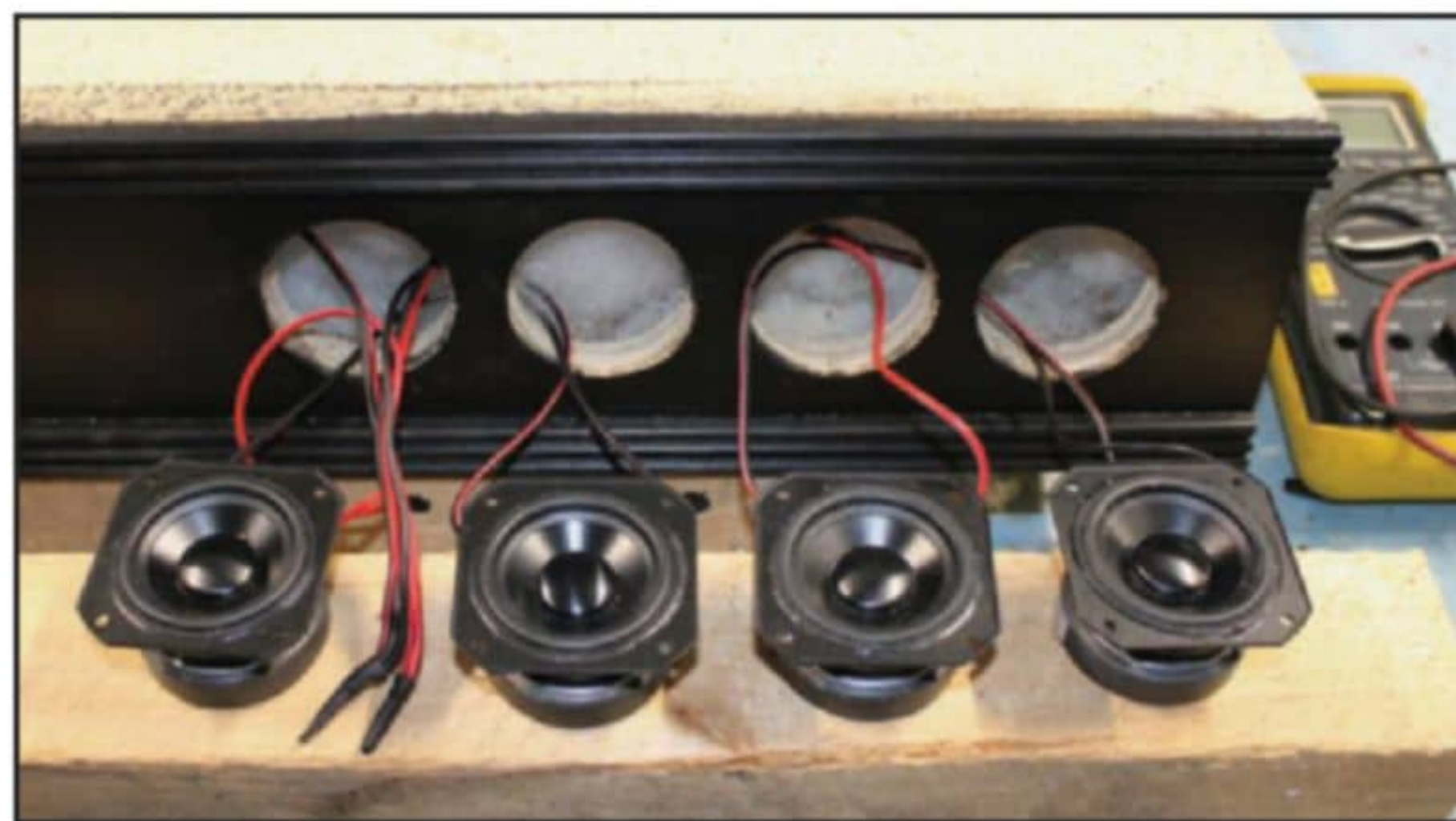
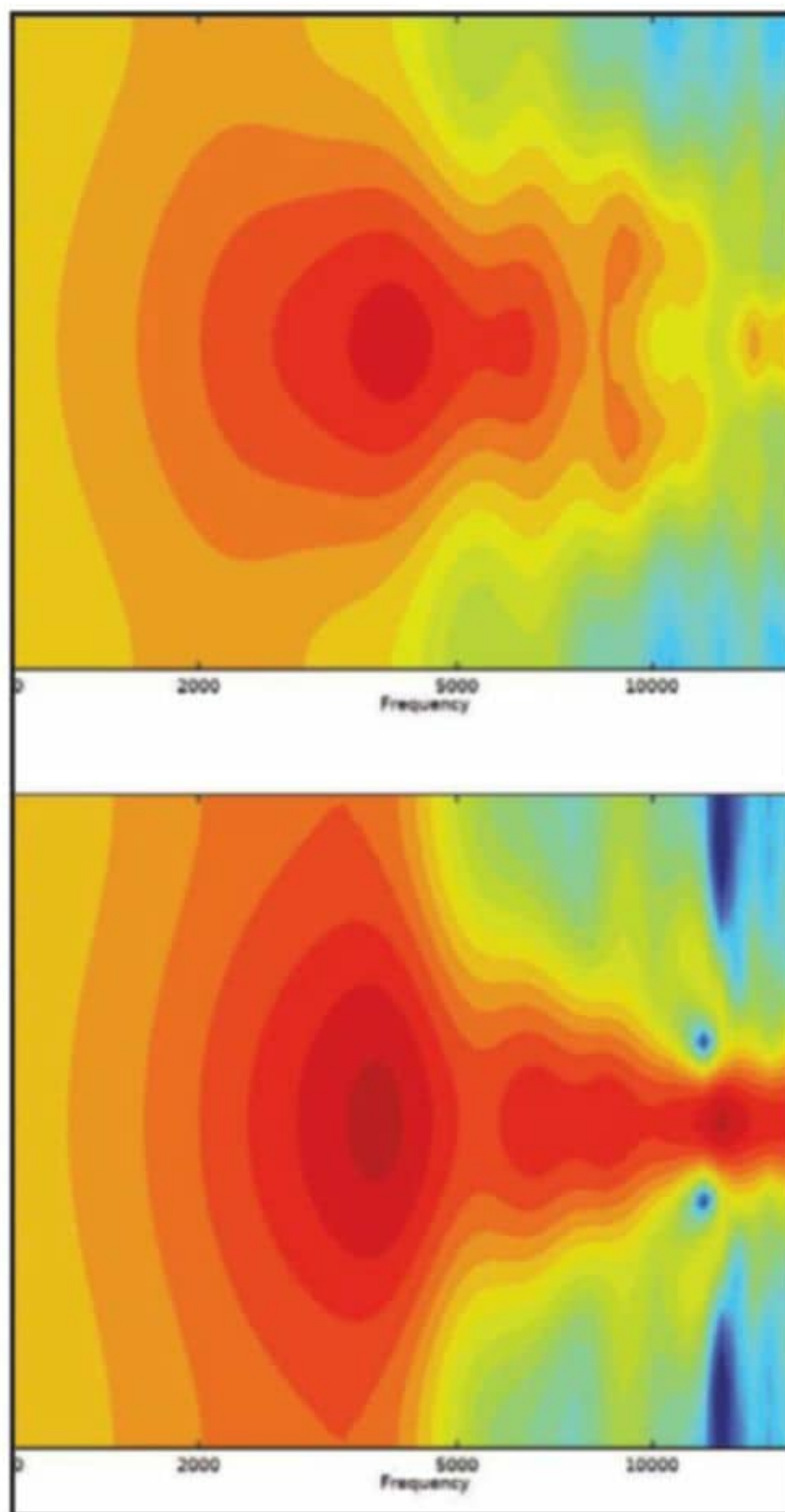


## Part of the Loudspeaker Industry?

Designed to help you locate the resources that are essential to loudspeaker manufacturing operations, the *Loudspeaker Industry Sourcebook* keeps you informed and connected with the companies making it happen.

**Don't miss out! Sign up for your FREE listing!**

[www.loudspeakerindustrysourcebook.com](http://www.loudspeakerindustrysourcebook.com)



## ● Features

### 28 Remote Recording in the Rock Era David Hewitt's *On the Road*

*Review by Scott Dorsey*

There were many well-known engineers who made their livings doing live recordings of rock concerts during the 1970s and 1980s. One of the most popular engineers was David Hewitt, who wrote a book about his experiences. After a thorough read, Scott Dorsey shares his review of the book.

### 32 Acoustic Topology Optimization for Vibroacoustics Applications

*By René Christensen*

Acoustic topology optimization is one of the most interesting new technologies in vibroacoustics, and is a powerful tool that enables engineers to find optimal solutions to problems related to their applications. In this article, several acoustic topology optimization examples are shown.

### 40 Vintage Op-Amp Noise Simulation Part 1 — Deviation from Realistic Noise Figures in Datasheets

*By Burkhard Vogel*

This article is divided into two parts. Part 1 presents the deviations from realistic noise figures in datasheets, based on measurements and calculations for real op-amps.

### 46 Monoblock Power Amplifier with Precision Bias Control Part 1 — The Design

*By Bruce E. Gillingham*

The project described here is a high-power (300W into 4Ω), bipolar, Class-AB, monoblock audio power amplifier. The design utilizes voltage feedback, a wide open-loop bandwidth (> 20kHz) approach, and an instrumentation amplifier architecture. It also features a novel precision Bias Controller that the author developed.

### 56 Building a Soundbase

*By Ken Bird*

One of the most successful speaker categories today, soundbars were originally intended to compensate for the poor sound quality of thin-display TVs. This project takes the concept up a notch in sound quality and fidelity for any living room application.

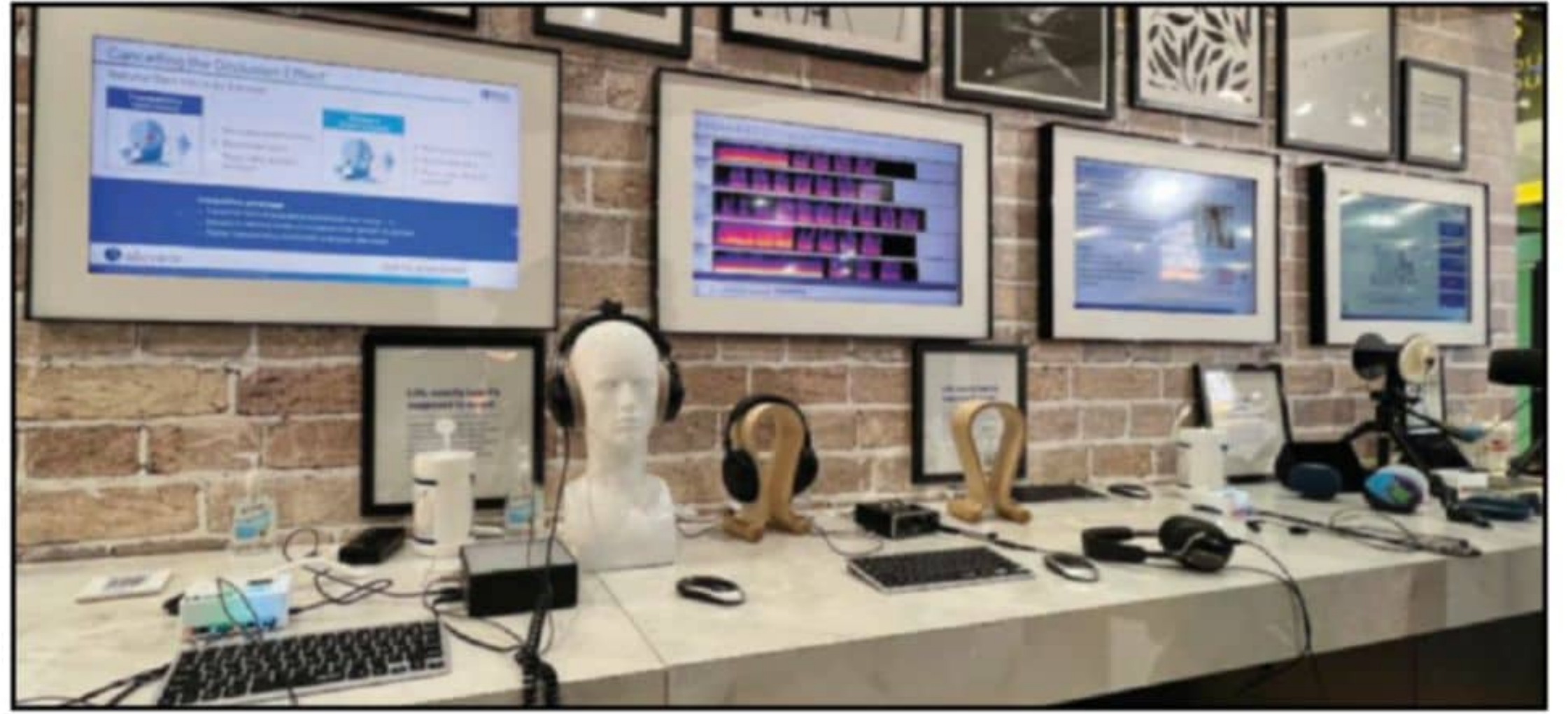
### 62 Power Transformer Parameters, Selection, and Testing Part 6 — Quality, Reliability, and Design

*By Chuck Hansen*

The initial group of articles detailed the history of transformer cores, their construction methods, materials, testing methods, and their losses. Part 6 focuses on commercial AC power quality, electrical component reliability, and the design of current transformers.

#### COVER ART

beyerdynamic Free  
BYRD True Wireless  
Earbuds



## ● Market Updates

### MARKET UPDATE

#### 8 State of the Hearables Industry

*By J. Martins*

From April 2022 to April 2023, true wireless earbuds and all the efforts to create product concepts in the hearables class have evolved dramatically. That's why this year's article focuses on projecting the evolution and the new possibilities.

### MARKET UPDATE

#### 20 More Choices for Headphones Test & Measurement

*By J. Martins*

This article is a brief update on test and measurement solutions for headphones and earphones that we couldn't fit in our March 2023 issue. An update on solutions that perfectly complement our hearables-focused article.

## ● Columns

### SOUND CONTROL

#### 36 Wrong Things "Everybody Knows" About Acoustics

*By Richard Honeycutt*

Our acoustic expert sets the record straight about some of the more common misconceptions about acoustics that he has heard over the years.

### SPONSORED

#### 30 New Microphone Technology —Flat Response Curve, Set Sensitivity

*By GRAS Sound & Vibration*

## ● Departments

### 4 From the Editor's Desk

### 5 Supporting Companies

## ● Websites

[www.audioxpress.com](http://www.audioxpress.com)

[www.voicecoilmagazine.com](http://www.voicecoilmagazine.com)

[www.cc-webshop.com](http://www.cc-webshop.com)

[www.loudspeakerindustrysourcebook.com](http://www.loudspeakerindustrysourcebook.com)

[www.circuitcellar.com](http://www.circuitcellar.com)

[www.linuxgizmos.com](http://www.linuxgizmos.com)



@audioxp\_editor



audioxpresscommunity



linkedin.com/company/audioxpress



audioxpress



# State of the Hearables Industry

By  
**J. Martins**

From April 2022 to April 2023, true wireless earbuds and all the efforts to create product concepts in the hearables class have evolved dramatically. That's why this year's article focuses on projecting the evolution and the new possibilities.



The true wireless stereo (TWS) product category is currently one of the hottest trends in consumer technology. It's also one of the most important product segments in growth and revenue and that happened faster than anyone involved expected, even though the product itself and its use cases are still evolving.

True wireless currently applies to both standard consumer TWS earbuds and hearing aids, with the two very distinct markets benefiting from common technologies that support independent wireless links to each ear-device. That breakaway from wires is fundamentally what proved critical for consumer adoption and it's been a technical challenge that in large part explains why some products have done much better than others. But let's not jump ahead.

In our own reports and contributed articles, *audioXpress* has illustrated that very intense and fast evolution, revealing how from solving technical challenges to a relatively simple use case, the audio industry has been transitioning to a completely new class of audio and voice applications and technologies.

Hovering above both the consumer-oriented TWS earbuds and medical hearing aids is now a completely new class of truly wireless devices with embedded intelligence, which we can easily predict could quickly expand to a complete ecosystem of wearables.

We are currently witnessing the quick evolution of crossover products that provide hearing assistance and are sold over the counter, while in the less-regulated consumer technology space we are seeing the rise of hearables and augmented hearing.

It was in 2014 that Nick Hunn (Chair of the Bluetooth SIG Hearing Aid Working Group, Vice Chair of the Generic Audio Working Group and a key contributor to the Bluetooth LE Audio specifications) coined the word "hearables." And yet, the original definition of hearables continues to constantly evolve from its original focus on sensors and "voice," to becoming software/app defined platforms, or "wearable computing devices" as ABI Research has defined them.

In the original text where Nick Hunn defined the transition from the wrist in wearables to the ear or "hearables" he writes:

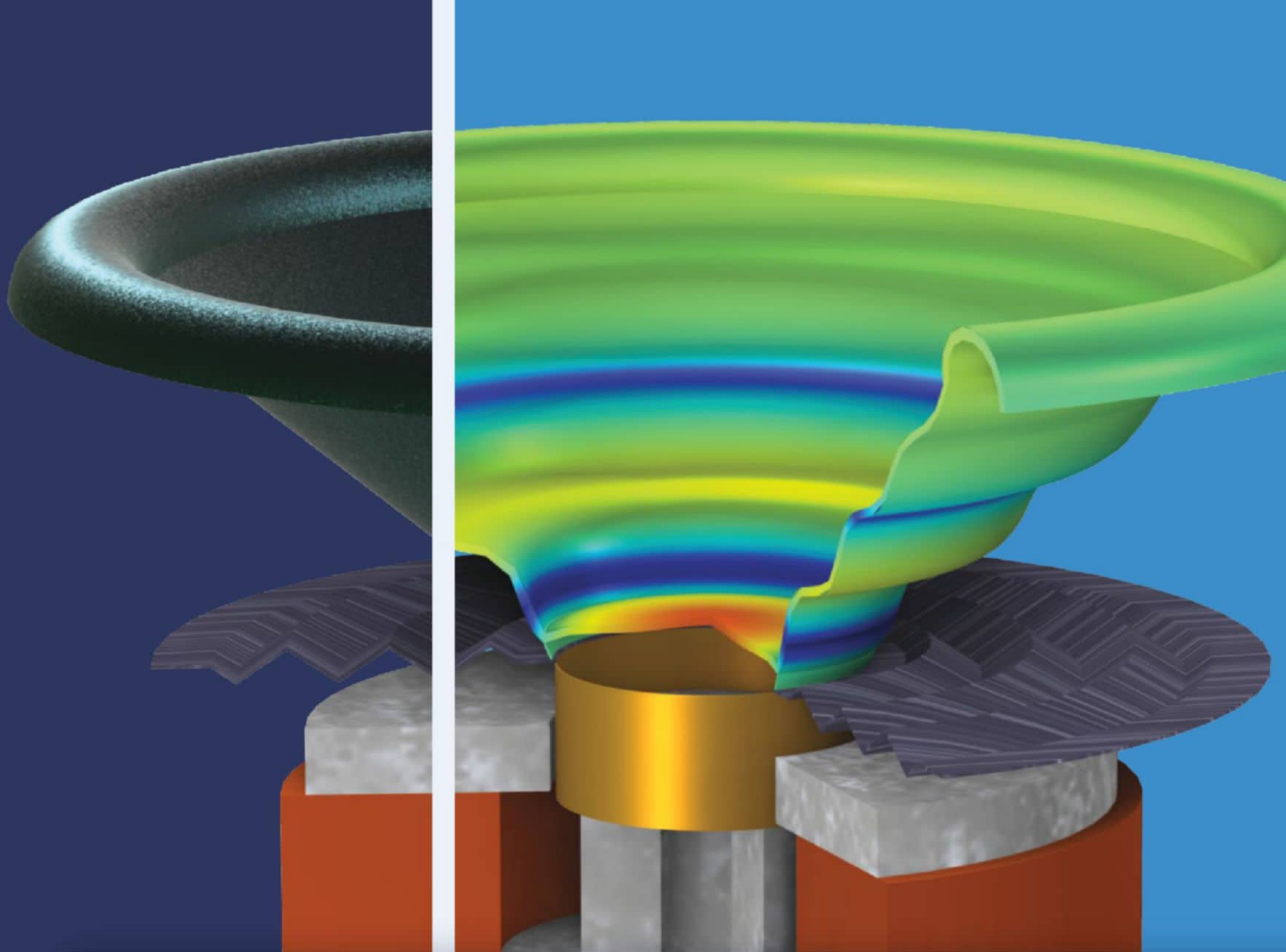
Analysts (...) almost invariably assume the wearable market is intrinsically linked with the smartphone market (...). To them, wearable seems to be mostly about smart watches and phones, which extend small parts of the phone experience to something we wear. They ignore the fact that we still purchase smartphones to make calls. All of those calls send audio to our ears. As well as voice, hundreds of millions of people use their phones for music, as evidenced by the ubiquitous cables trailing from their ears. Sound drives the bulk of our technology use and earbuds are the only piece of wearable tech to have gained ubiquity and social acceptance. These devices are about to undergo a revolution in capability, getting rid of their cables and giving them the opportunity to be the standard bearer for wearable technology."

He also noted: "We're seeing the first few signs that developers are beginning to understand that the ear has more commercial value than the wrist."

At the same time, true wireless stereo or TWS technology — largely based today on Bluetooth, although better technologies and approaches existed (including Near Field Magnetic Induction) — also quickly evolved from the original vision of simply replacing headphones with compact wireless, miniaturized devices that could be worn at all times.

Today, smartphones are hardly used to make calls, and likewise, true wireless earbuds evolving toward hearables are also being used for many other things—none of which determines the category, much like smartphones.

Currently we have a wider definition of hearables, combining smart processing features, all types of sensors including health-related sensors, voice recognition (voice assistants moved to the background a bit), and of course built-in computer and edge AI processors with user-defined software apps. And in that regard, Nick Hunn's original analysis of the "appcessories," which he predicted in 2013 for Bluetooth technology, is actually happening and is helping create what we have been broadly defining as



# Take the Lead in Acoustics

with COMSOL Multiphysics®

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

» [comsol.com/feature/acoustics-innovation](https://comsol.com/feature/acoustics-innovation)

“hearables.” Some companies, such as Gary Spittle’s Sonical envisions a new ecosystem of ear-computers—touted as Headphone 3.0—where users can not only pick and choose apps, but those apps help define the device itself and its use cases, including personalization of hearing for augmentation and health monitoring.

## Eight Years of Change

The growth and evolution in the true wireless category was relatively fast, even if it doesn’t seem so for those of us following the product evolution up close.

It has only been only eight years since this product category was created and six years since it became a commercial reality. At the foundation, the purpose was to listen to music anywhere and without wires. Apple applied its usual strategy to the new category and entered the space. After having already created a gigantic business with wired EarPods, the company decisively entered the true wireless space in 2016, when it decided to remove the headphone jack from the iPhone 7 in order to improve the



Launched in 2016, the original Apollo 7 true wireless earbuds from Erato Audio can still be found on Amazon and online retailers. Unlike the Noble Audio, Kanoa, HearNotes, and others who never shipped and faded away, Erato managed to design what remains one of the smallest TWS earbuds ever, and send it to market. It allowed listening to music and making calls. Simple.



Since Bragi announced its Bragi OS and #Bragienabled software strategy for true wireless earbuds and hearables, the company has been scaling its technology and investing more in advanced AI and embedded software. It currently supports a range of third-party apps in commercially available products.

critical ingress protection rating of the smartphone.

Many companies had pioneered the concept before the Apple AirPods, and all were essential to identify the key technology bottlenecks and encourage the industry to move forward with the design of critical electronics components.

German company Bragi, one of the few surviving pioneers in the true wireless space, anticipated the announcement of Apple’s headphone-jack-free iPhone 7 with an announcement of its own. While some audio companies were still replacing the 3.5mm headphone jack with a Lightning connector and the benefits of direct digital connection, Bragi tried to make the best use of its technology already developed for The Dash true wireless earbuds to create a new simpler model—The Headphone—focusing on compatibility with any Bluetooth device... including the iPhone. And yet, making a simpler model than the Dash just to wirelessly stream music proved challenging, as many other companies were learning. Choosing where to focus their resources in a very complex design proved challenging for many companies that lacked the experience and industrial connections that others such as Erato Audio had in place.

Another important pioneer in the space, Australian company Nuheara, launched its flagship product, IQbuds, at CES 2017. The vision was to create intelligent, truly wireless earbuds that would go beyond the convenience of truly wireless music listening by promoting the benefits of assistive listening. Nuheara wanted to be the first to market this augmented hearing concept, which was only somewhat realized two years later. Not much time considering the ambitious technology, but a long time given how the market evolved in that space in that timeframe, with countless other companies doing many other (different) things better.

Like most of those TWS pioneers (including Nuheara), Bragi started with a Kickstarter campaign and over time it built considerable engineering resources on a field that was highly sought after, leveraging more than 150 unique patents in approximately four years. Bragi anticipated the vision for “hearable concepts,” exploring the array of sensors and technologies that it mastered for Bragi Dash, which was also the first product in the category to allow software updates. Somewhat surprisingly, in February 2018 the German company decided to pivot into developing “the first edge-computing AI suite” using its breakthroughs in ultra-efficient artificial intelligence. During Mobile World Congress (MWC) 2018, Bragi offered a software suite formed by the building blocks BragiOS, nanoAI, BragiNET, and Hardware Reference Design, proposing a platform centered around machine learning techniques to the industry and enabling manufacturers to gain deeper insights into the usage and needs of their users.

In March 2019, Bragi sold the product business to a third-party company to focus on scaling its technology to transform the company into a software and service integrator. Bragi also anticipated that its efforts could be applied in connected vehicles, wearables, sports, and healthcare since its software suite enables companies to equip their Bluetooth audio products with advanced natural user interfaces and new experiences.

And Bragi anticipated the importance of user uploadable (or activated) apps, starting with personalization features from Mimi

Hearing Technologies. A vision that would be further realized with future implementations of “Smart Features” for Skullcandy and Klipsch, inspiring the idea of a marketplace for hearables software apps.

The challenges in this approach are multiple. Starting with the need to support multiple chips and hardware configurations, while at the same trying to see which “killer-apps” are the ones that manufacturers and brands want to add. Because it’s not as simple as loading software when the “apps” involve things such as gesture control, voice commands, trigger a remote selfie, improving ANC, or finding where the earbuds case was left the last time we used it (Find My). All those are features that have been implemented and that users want or appreciate. Others yet to be recognized almost always imply access to hardware resources.

When we look at the available products in the market in 2023 we find a variety of platforms powering true wireless earbuds, but the features in the latest products are all fairly common, in great part due to the fact that companies such as Qualcomm made it possible for manufacturers to quickly gain access to complete reference designs based on its QCC chips and growing Snapdragon Sound platforms.

Not surprisingly, products in the top price categories now combine a Bluetooth 5.3 radio (supposedly able to be upgradable to support Bluetooth LE Audio), two or more microphones per earbud to support ANC with smart ambient or transparency features, sensor fusion, and an ultra-low-power processor and edge AI engine. Consumers now also expect more from their devices, and higher quality streaming options (lossless, hi-res audio, and adaptive streaming), together with spatial audio features with head-tracking are all common to a growing number of premium products (even if in reality users are not able to explore those features for lack of access to content, or unwillingness to be tied to subscriptions models). Improved voice capture with smart noise DSP suppression, which might be separated and filtered with the aid of a bone conduction transducer, is now a popular feature for improving calls, web conferencing, and voice recognition applications. And all the associated processing is now a requirement, along with heavy data usage over 5G networks, while battery life expectations are now expected to reach at least six hours with ANC active, or eight hours of continuous audio streaming.

The combination of these systems is also finally enabling new levels of hearing personalization and augmentation of super-hearing. Users not only want the system to measure and compensate for their own hearing loss or sensitivity—preferably automatically adaptable to any usage situation—but they also want to be able to clearly hear conversations, even in noisy environments or at increased distances, gaining extra-human capabilities. Those speech enhancement features, largely resulting from the convergence with the best medical hearing aids technologies, are going to be a great selling point going forward for hearables.

Finally, and as a result of the evolution of Bluetooth standards, users now expect to have multi-pairing and multi-connectivity with their multiple devices, and very soon Bluetooth LE Audio will make Audio Sharing extremely popular. Auracast or broadcast



The second-generation AirPods Pro earbuds seemed to be almost the same as the first-generation product. And yet, everything about these TWS earbuds was far superior and established a very high reference for the industry.

audio will also offer some potentially interesting applications, but since the use cases depend upon installation efforts that cannot be anticipated, it will not be a key consumer driver until clear use cases become visible and popular.

Lingering in the background, voice recognition applications and voice assistants in earbuds will continue to be a very promising field. For those interested in voice as a key feature, from voice communications to voice interfaces and voice enhancement, our 2022 Market Update article remains reference reading material.

As mentioned, using our earbuds to access and connect to our virtual assistants (or personal digital assistants), in relative privacy quickly moved from a “killer app” in 2018 to a nice to have. Priorities changed mainly due to privacy concerns due to the always-on connectivity requirements, but also because of the lack of progress in the voice assistants per se, and language support.

This is bound to change as our platforms and ecosystems evolve in parallel with the smartphone and “personal computers,” which are inherently always-connected, and the Bluetooth-tethered



In the premium segment of the market, Bang & Olufsen offers a made-to-order collection of its Beoplay EX true wireless earbuds, with prices reaching \$699 USD. Selling points include superior sound reproduction with aptX Adaptive, multipoint connectivity, and adaptive ANC with three microphones in each earbud, in a fully waterproof design.

devices, such as hearables also gain capabilities with edgeAI. With better AI and machine learning, there is a new opportunity for using assistants for augmentation and we can expect to see great progress on that front in the next year. OpenAI GPT language models can change all that in a matter of months as we've seen from ChatGPT. And from the enthusiasm we are seeing in using those tools, we can only guess as to what would happen to hearables if consumers could ask these engines anything and hear the replies directly on earbuds. Also, we can expect exponential progress in translation engines and their applications on edge devices.

Consumers are not worried about “carrying too many gadgets” as long as they use them and need them. It is possible that smartphones will gradually transform into devices so compact that we just wear them, potentially transitioning to our wrists, collars, glass frames, or any wearable jewelry (although I would never disregard the appeal of a great display).

TWS earbuds can connect when needed to enable augmentation provided by our AI-powered assistants. No need for a true wireless earbud to be connected to an Auracast channel at the airport lounge to receive gate change announcements. Our Personal Digital Assistant, connected to our smartphone or smartwatch, constantly monitors flight information and any airport announcements since it knows we are traveling and that we are located at the airport. Any changes or notifications are transmitted to our ears or any of the screens we wear only when appropriate.

The “broadcast” uses of Bluetooth Auracast in that regard can quickly be made redundant by our own connectivity without requiring further installations and infrastructure investments. Although, tuning to an audio channel for hearing assistance, real-time translation, or even for entertainment, remains a valid proposition—but that can be done to any of the connected-devices we already carry. The charging case for our earbuds, with its large capacity battery and space for a display, is an attractive platform to improve upon with “smart” features. But with energy harvesting and wireless charging technologies quickly evolving, maybe carrying a charging case when out of home will no longer be a priority in the future. Staying connected remains the core priority and whatever device is used for mobile connectivity will remain at the core of our personal network ecosystem, of which hearables will be a critical extension.



Low-power wireless and DSP platforms—even edge-AI processors that are able to process natural language—are now available, specifically optimized for TWS earbuds. Qualcomm is pursuing that strategy with its Snapdragon Sound platform, currently in its second generation, designed to support Bluetooth LE Audio.

Completely untethering our hearables? Yes, it could be important in some cases, but consumers will likely want the option to decide when to be connected or not. For now, it is just too unpredictable to think that any category will “win” over another and that any single device will rule over the others.

True wireless earbuds and hearables will remain a strong proposition.

## Development Bottlenecks

What we know has changed since our last Market Update article on this topic is that Apple has launched a second-generation AirPods Pro, which is much better than what we had last year—although at first the industry seemed to disregard it. What didn’t change is the need for most of the whole consumer electronics and mobile industry to wait until it learns about what Apple is doing and what they will be competing with.

With a market share between 29% and as much as 56%, depending on who does the research and how, Apple leads the overall true wireless earbuds market globally. We are not focusing on unpredictable statistics in this report, but we can reliably estimate that if we compare between what consumers bought and what consumers use two months after purchase, Apple holds two-thirds of the true wireless earbuds market in North America and much of Europe. Elsewhere in the world, such as India, Apple is just too expensive, and many of the contenders in those markets—from India and China—just try to copy all they can from the AirPods. In that effort, companies such as Boat, are making an effort to differentiate and focus on some innovations, consolidating its gained market share.

But even if the different AirPods models (and not forgetting those from Beats) continue to lose market share given the number of competitors that enter the segment, Apple still has increased sales volumes, as the market overall continues to grow. As it happens with the iPhone, more important than the units sold, Apple gains more revenue in the product category than all the profits of all its competitors added up.

As everyone is trying to compete in the space, with the AirPods, as with any other product segment that Apple enters, its products combine unique custom hardware, a dedicated OS and a software ecosystem. And since no other player in the market combines all those fundamental pieces, no one can optimize products to the level Apple does.

Obviously, manufacturers want to differentiate new models and Bragi is just one of the companies offering a complete software platform for the category, with its Bragi OS that includes multiple service integrations like Amazon Alexa and embedded AI. This effectively offers a sort of “marketplace” for product developers and manufacturers to “shop” and implement in their products. But that choice, App Store style, is still not available to the consumer.

For developers and manufacturers, designing new true wireless earbuds can be an exercise in frustration, given the tremendous platform constraints. Even for those brands that are able to design from a blank canvas, it’s hard to predict all the different combinations of features they can put into a single product and make it market fairly quickly. A few examples of new efforts in

# **ViBRIDGE**

*In-ear device testing – faithful to the bone*

Bone conduction simulation – the game changer



**REAL-LIFE-PROOF.**

[www.head-acoustics.de](http://www.head-acoustics.de)   

that perspective serve as a reminder of how hard that journey can be—and how tempting it becomes to just use the Qualcomm platform and make it quickly to market to dispute a small share among all the other small fish.

At CES 2023, this was evidently clear as most of the demonstrations in the hearables space—mostly focusing on adaptive ANC—depended upon one of the leading Bluetooth SoC platforms and most simply emphasized a specific edgeAI or DSP platform.

That is precisely where a new company is seeing an opportunity. Sonical, based in the Silicon Valley of California, wants to solve this dilemma for manufacturers by creating a fully integrated approach that could revolutionize the product development of headphones and earbuds, while at the same time helping consumers to gain more choice. The company is designing a new RISC-V-based “ear computer” that is able to leverage powerful AI and an advanced audio processing platform, running a dedicated real-time operating system (based on FreeRTOS).

Working to fulfill this vision since early 2020 is Gary Spittle, Sonical’s CEO and founder. Spittle is a technology and product strategy expert, with a long career in the audio industry, where he led companies in the music, professional audio, consumer electronics, and mobile segments. And he has worked directly with most of the world’s leading audio product brands for more than two decades, defining, designing, building, and shipping products on a large scale. Before Sonical, he worked with Knowles, Dolby, CSR (before the acquisition by Qualcomm), CEDAR Audio, and others, piling up multiple patents in the process.

As Spittle details, the opportunity exists for an open platform that is software defined, allowing developers to write high-level apps and plug-ins and rapidly deploy them. “Having just a software framework is not enough, it will always be restricted by existing silicon performance. The optimal solution (proven by CSR) is a

combination of hardware and software that is well supported for product manufacturers and developers to innovate. The next-generation of products will not only have improvements for how we listen to audio, they will also listen to the world around us and present an enhanced version using augmented reality algorithms, and will also listen to our bodies to monitor our health and wellness using AI,” Spittle told *audioXpress*.

Spittle is very familiar with all the challenges of developing Bluetooth products and designing products and solutions for headphones, hearing aids, and hearables. That led him to understand that the industry “needs a new chip.” And that was even before Apple did just that with the H1 processor. For now Sonical has partnered with Dolphin Design, an embedded AI silicon IP company, and a subsidiary of semiconductor manufacturer Soitec, to design this new computing platform. The ability to run apps in small, ultra-low power hearable devices involves a new generation of ASIC/SoC that combines greater computing power and artificial intelligence, without requiring extra power consumption. The processing IP design platforms provided by Dolphin Design are supposed to bridge this gap. These new platforms are built around RAPTOR, a Convolutional Neural Network (CNN), and PANTHER, a software programmable RISC-V-based AI/DSP accelerator. At CES 2023, Dolphin supplied a working evaluation board that Sonical used in its demonstrations, mainly to convince software companies to write code optimized for the platform.

It’s early to say how much traction this will have in the industry, but for now Sonical is encouraging a powerful ecosystem of technology companies to work together. Spittle is also exploring other market opportunities, including in pro audio applications, where clearly the current limitations of Bluetooth for high-quality, low power wireless audio are too restrictive.

## The AirPods Reference

As mentioned, the major event that determined the direction for most manufacturers and the strategy for most of the technology companies this year has to do with the launch by Apple of its second-generation AirPods Pro in September 2022.

Much as no one knew exactly what to expect from the original AirPods Pro introduction, which surprised everyone in the midterm with the introduction of OTA software updates and the launch of Spatial Audio support (leveraging the head-tracking that was already built-in), very little information leaked about what Apple intended to do with the second generation. And the possibility of future updates caused more anxiety and confusion for competing efforts.

It turned out that this announcement had way more impact than anticipated by reinforcing what Apple had previously already done right.

We don’t know which model Apple sells the most, if it is the updated “loose-fit” standard AirPods or the in-ear and more expensive noise cancelling AirPods Pro. The AirPods’ third-generation were launched in October 2021. Nearly a year after that the flagship AirPods Pro were announced, delivering 2x better ANC, longer battery life, better drivers and a new and improved charging case.



The second-generation AirPods Pro earbuds offer 1.5 hours of additional listening time over the first generation, for a total of up to six hours with ANC. The most important innovation is in the area that separates the AirPods Pro from the standard AirPods: ANC. The second generation of AirPods Pro brings major upgrades to ANC and Transparency mode, thanks to the new H2 chip that is able to cancel up to twice as much noise over the previous generation AirPods Pro. Also powered by the new H2 chip, new Adaptive Transparency processing reduces loud environmental noise for more comfortable listening while still enabling users to be aware of the surrounding environment.

What many competitors found most disturbing was the fact that the second-generation AirPods Pro kept the same—already iconic—design, available only in white. And inside, they are much improved products with better ANC, canceling up to twice as much noise than the original model, and offering an improved Transparency mode, making it adaptive with active processing. As Apple promoted, the new AirPods Pro are able to minimize the intensity of loud noises using computational algorithms, promoting safety against very loud noises, while leaving the ANC or transparency settings untouched. The company also brings Personalized Spatial Audio, or loading HRTF profiles, as announced in the Apple iOS 16 update preview.

To enable Personalized Spatial Audio, users just need to point the TrueDepth camera on iPhone (front) to their ears. The captured information is used to create a personal HRTF profile for Spatial Audio, which is then enhanced even more with dynamic head tracking for content viewed on any iPhone, iPad, Mac, and Apple TV. Those are mostly software-enabled features that are part of iOS, much as the Accessibility features for people with disabilities, or Headphone Accommodations and Conversation Boost for hearing enhancement. This is an area that Apple has been improving directly through software, firmware, and OS updates.

The second-generation AirPods Pro also feature a more powerful H2 processor and support for Bluetooth 5.3. The previous H1 is now built into the new MagSafe charging case to enable Precision Finding, which means that users with an iPhone can locate their charging case with guided directions.

For those listening to music, the drivers are much improved, lower distortion units that translate even better the dynamics of a new custom amplifier. The well-documented experience from countless users was unanimous in praising the sound quality of the AirPods Pro second-generation, matching “audiophile” wired in-ear monitors costing four times more. A very high threshold to reach.

## Form Factor and Comfort

As Andrew Bellavia wrote in his article analysis for *audioXpress*, “Hearables in 2023: Comfort Must Come First,” to unlock the full potential hearables, which need to be comfortably worn for long stretches of the day, the form factor is a key factor, including being something that people are willing to wear in social settings. Currently, as noted in the article, users do not wear earbuds for long continuous times due to discomfort, including pain, and irritation. “Addressing this must be job one in 2023 if advanced applications are to be successful,” writes Bellavia.

In this same article, occlusion is mentioned as another major contributor for comfort, an area that explains why adaptive transparency modes are seeing the same or even more acceptance by consumers than ANC.

But let’s start by looking at the form-factor, which continues to be mainly defined by two base designs that Apple adopted for its AirPods: the loose-fit shaped earbuds in the standard AirPods, and the closed in-ear in the AirPods Pro.

“The 2022 State of Sound Report,” a global analysis of audio consumer behaviors and desires compiled annually by Qualcomm,



Skyline’s DynamicVent technology will enable TWS engineering teams to deliver a new generation of more capable adaptive transparency and adaptive ANC functions. The new solution from xMEMS enables system designers to balance the benefits of open and closed-fit earbuds, improving comfort and providing solutions that better adapt to a variety of listening environments.

help us understand how consumers use and the qualities they seek in a number of audio products, including true wireless earbuds. Not surprisingly, in the 2022 report ANC remains the highest-ranked rich audio feature, joined by lossless audio as a heavily desired feature.

Audio purchase drivers for TWS earbuds continues to be price, comfort, and sound quality as leading factors. And due to consumers wearing true wireless earbuds and headphones for longer periods of time throughout the day, comfort has moved up to number two. ANC is now in the middle of the purchase drivers list, and context awareness and hearing enhancement are also mentioned higher compared to the “2021 State of Sound” report.

Also significant is the fact that the true wireless earbuds market is still experiencing growth. According to the report, 41% of respondents said they currently own a pair, up from 35% in 2021 and 20% of respondents indicate that they plan to purchase true wireless earbuds in the upcoming 12 months. Among the trends, users are saying they want both better ANC and more long-term comfort, which is the biggest challenge for designers of these products.

Sony Electronics has recently addressed new avenues in earbud comfort with its latest LinkBuds, which introduce an open diaphragm ring design that delivers improved audio transparency, improved fit, and crystal-clear sound and call quality. A platform that Sony says is intended for long-term use and applications



The Bose QuietComfort Earbuds II introduced an all-new design for the brand. Each earbud is about one-third smaller than its predecessor, weighs less, and because they are much smaller, come in a pocketable charging case.

including augmented reality (AR) content, navigation, and work-from-home. Combining a loose-fit design with the open 12mm ring transducer, the Sony LinkBuds let users tune into the outside world whenever they want, without compromising the listening experience. At approximately 4 grams, these earbuds are also small and lightweight, so users can wear them for extended periods of time. Unlike conventional earbuds, the protective surface covering the speaker diaphragm is integrated within the housing, saving space. The overall shape has also been crafted using extensive ear shape data, collated since Sony introduced the world's first in-ear headphones in 1982.

Finally, the LinkBuds also offer precise voice pickup technology and advanced voice signal processing, combining Sony's noise reduction algorithm—developed with AI machine learning, using more than 500 million voice samples to suppress ambient noise and extract voice clearly. This allows making clear calls even in noisy environments. The trade-off from the miniaturization is that the LinkBuds will work for a maximum of 2.5 hours before they need to be recharged.

Designers of new earbud products will be happy to know that there is now a new approach to the occlusion problem that not only implies any of those compromises, but even improves the response and consistency by using MEMS microspeakers instead of conventional dynamic drivers. Part of xMEMS Labs' announcements at CES 2023, the company introduced Skyline, the world's first discrete solid-state MEMS DynamicVent, a specific design with a valve that opens or closes by sensor fusion input from the earbud DSP and enables the benefits of both an open-fit (spatially aware) and closed-fit (occluded) earbud design. This enables smart TWS earbuds and hearing aids that create best-of-both-worlds user experiences in the same product.

The new Skyline already represents xMEMS's second-generation DynamicVent technology following last year's introduction of the Montara Pro. Unlike Montara Pro's vent, which was integrated



In addition to creating a near perfect seal thanks to their small size, the 1MORE ComfoBuds Mini use an AI-powered DNN (Deep Neural Network) algorithm that helps to recognize, learn, and amplify the user's voice to improve the phone calls experience and effortlessly remove ambient noise.

on the same die with a MEMS speaker, Skyline is a discrete, stand-alone solution and can be paired with xMEMS's solid-state microspeakers or traditional coil-based speakers. This technology can enhance popular adaptive transparency and ANC features in advanced TWS earbuds, by enabling the earbud system DSP to intelligently open or close the valve based on ambient noise levels detected by microphones or the listener's activity from motion sensors. And Skyline can also eliminate the need for oversized speakers in earbuds that are commonly used to compensate for low-frequency loss caused by persistent, static vents.

### The DSP Route

The other approach is to use the same systems for ANC earbuds and transparency modes, applying both better analog-front ends and DSP to help improve the undesirable effects of occlusion.

That's the approach followed by Bose, which recently updated its family of ANC products with the now-flagship QuietComfort Earbuds II. For the ANC pioneering company to offer "the world's best noise cancellation" in TWS earbuds required a large engineering effort. The Bose QuietComfort Earbuds II leverage the most recent available technology to move to the front of the highly disputed TWS ANC category, using Qualcomm's most advanced S5 audio SoC combined with its own CustomTune sound calibration technology that intelligently personalizes audio and noise cancellation performance to the unique shape of every ear.

Starting by completely closing the ear canal in order to provide isolation, the new Bose design uses four microphones in each earbud—one on the inside and three on the outside—to sense, measure, and send unwanted noise to the dedicated processor, loaded with Bose's exclusive algorithms. The CustomTune sound calibration is triggered every time the earbuds are removed from their case and placed in the user's ears. A proprietary tone is played, and a microphone inside the earbud measures the ear canal's acoustic response—a technology that Bose has been perfecting for its OTC hearing aid efforts. That information is then used to precisely tailor both audio and noise cancellation performance especially for the user—all within less than half a second.

The more advanced noise reduction also targets frequencies that were previously difficult to reduce, and CustomTune also continuously adapts to changing noises in the surrounding environment. The intelligent personalization provided by CustomTune also enhances Aware Mode with ActiveSense. In Aware Mode, it delivers improved transparency, and ActiveSense responds with just the right level of tuned noise cancellation to reduce noise as it happens, and only for as long as it lasts.

Measuring and fitting the acoustic signature of the ear canal and adjusting the noise cancellation to fit the response at every frequency is not exactly new. One way or another, Apple and other manufacturers are applying the same approach in the latest Adaptive ANC designs. What is new with CustomTune is the level of optimization that is completely focused on just noise cancellation optimization, counteracting the ear canal's unique resonances.

Another manufacturer that deserves mention in this context is 1MORE, a Chinese manufacturer with recognized expertise in

personal audio, which launched the ComfoBuds Mini, combining ANC and personalization. Weighing no more than 3.7g each and sized at just 17mm×13mm, these mini earbuds are packed with all the key features expected in modern-day true wireless earbuds, with a combination of attractive features, convenience, and an original form-factor for totally wireless earbuds.

The 1MORE ComfoBuds Mini feature ANC effective up to 40dB, based on the signal that is captured by four microphones, which is then processed internally by a deep learning algorithm. Inside these small earbuds, the processing power is able to power multiple listening modes that combine more or less cancellation or transparency, while also supporting SoundID personalized sound from Sonarworks. The earbuds use 1MORE's QuietMax ANC Technology with both feedforward and feedback microphones, also providing a Transparency Mode and a Wind Noise Resistance Mode that helps to filter out the annoying harsh frequencies caused by high winds or while cycling at great speed.

## The Road to Augmentation

At CES 2023, Analog Devices had a large demonstration area, both for in-house developments and technology partners and clients, focusing on the possibilities to actively enhance our listening abilities and comfort while wearing earbuds. Using its ADAU1860 programmable FastDSP low-power codec and audio processing engine, with a Tensilica Hifi 3z DSP core, the company showcased how to adjust ANC filters, create hearing augmentation applications with speech enhancement and noise control, or manage adaptive pass-through filters for eliminating the occlusion effect.

The Analog Devices team was focusing on what it calls Ambient Isolation Management, a complete solution where ANC is always a foundation running on the FastDSP, with extra control/features leveraging the Tensilica DSP. This solution covers maximum isolation (maximum adaptive ANC) to transparency with occlusion cancellation, and the whole range of Hearing Augmentation starting with a hear-through mode with beamforming plus noise suppression, with applications able to meet the OTC-specification cases.

In one of the demonstrations, client Austrian Audio was able to address tuning an ANC earbud design for professional audio applications, while Jean-Marc Luneau (co-founder of 7 Sensing Software and now part of Analog Devices) showcased how to augment conversations in noisy environments with low latency audio processing.



The large demonstration area promoted at CES 2023 by Analog Devices, technology partner companies, and even clients focused completely on Ambient Isolation Management applications for hearables.

Another key demonstration in that regard was from Elevear, which addressed how to eliminate the occlusion effect in in-ear ANC earbuds, running on the Analog Devices ADAU1787 audio codec. The German company has developed its Occlear solution for occlusion cancellation and intelligent transparency, implementing a wideband approach that is a step-up from existing approaches. Occlear uses a patented Idle Noise Reduction (INR) technology that reduces the noise floor of hearable microphones by 10dB to 15dB to improve the sound of the user's own-voice and perceive the surrounding environment more naturally. Occlear requires no specialized hardware and can directly be deployed to ANC-ready hearables via a firmware update.

Another company with extensive experience in this domain, and offering multiple technologies for both speech recognition, voice enhancement, and hearing enhancement is Alango Technologies. At CES 2023, Alango promoted demonstrations of its systems and algorithms in collaboration with Cadence, Realtek, and Knowles. Alango offers solutions that help enhance both the surrounding sound through advanced noise-suppression algorithms, but also offers advanced solutions to separate and process the user's voice separately from the rest of the processed signals, which is fundamental in order to address the unnatural effect of occlusion.

In order to expand on those applications for hearables, hearing aids, and communications, Alango demonstrated at CES the Only Voice sound pickup technology, running on the Knowles IA8201 AISonic edge processor and using Knowles SiSonic microphones. The advanced voice processing technology is ideal for TWS devices

## Resources

A. Bellavia, "Hearables in 2023: Comfort Must Come First," originally published in *The Audio Voice* newsletter, (#404), January 5 2022, <https://audioxpress.com/article/hearables-in-2023-comfort-must-come-first>

Dolphin Design, [www.dolphin-design.fr](http://www.dolphin-design.fr)

Elevear, <https://elevear-tech.com>

"Hearables: Emerging Opportunities, Competitor Leaderboard & Market Forecasts 2022-2026," Juniper Research, [www.juniperresearch.com](http://www.juniperresearch.com)

N. Hunn, "To Ubiquity and Beyond: Bluetooth Smart and the Growth of Appcessories," WiFore Consulting, [www.nickhunn.com/wp-content/uploads/downloads/2013/03/to-ubiquity-and-beyond.pdf](http://www.nickhunn.com/wp-content/uploads/downloads/2013/03/to-ubiquity-and-beyond.pdf)

N. Hunn, "Hearables—the new Wearables." Creative Connectivity, April 2014, [www.nickhunn.com/hearables-the-new-wearables](http://www.nickhunn.com/hearables-the-new-wearables)

K. Shin, "TWS Biggies Focus on Differentiation to Beat Competition," Counterpoint, December 2022, [www.counterpointresearch.com/tws-biggies-focus-differentiation-beat-competition](http://www.counterpointresearch.com/tws-biggies-focus-differentiation-beat-competition)

Sonical, Inc., [www.sonical.ai](http://www.sonical.ai)

since it allows combining signals from external microphones, an in-ear microphone, and a bone conduction sensor, allowing applications that perfectly filter the users' voice from other signals, and help to deliver completely noise-free voice capture.

Another approach for the same problem was developed by Swiss company AVAtronic, and acts purely on the signal processing domain. Having experience in extremely noisy applications in industrial, aerospace and military applications, AVAtronic offers unique active noise cancelling technology, that both combines hearing protection and personalized ANC earbuds.

AVAtronic has developed a unique wideband ANC technique, which benefits from deep Neural Network techniques to adapt the ANC system to the environmental noise or specific application scenario. AVAtronic uses a totally different approach from all the current adaptive ANC techniques used for earbuds, and as a result, it's enabling earbuds and headphones manufacturers to approach the market with wideband ANC devices that work optimally in any noisy environment.

AVAtronic already implemented its wideband ANC technology in multiple technology platforms commonly used for earbuds and headphones with the capability to run sophisticated processing algorithms in real time.

Clearly, the future in this domain seems to combine approaches with the possibilities now offered by AI and ML, directly running on-device, to create fully adaptive systems, which are critical for the hearables and hearing aids. After all, the ultimate goal for hearables, like in hearing aids, is precisely to approach the user experience from the natural human listening experience, which is not conditioned by preset filters. From the opposite perspective, the combination of improving microphones, sensors, and advanced DSP, allows us to go beyond the natural limitations of even the better human listening capabilities. Much like our brain adjusts to the information captured and transmitted by our hearing systems, hearables will be able to augment our auditory system.

And we are getting very close to those possibilities being widely available in this exciting consumer category.

## TDK InvenSense

# VibeSense360 Development Solution for TWS Designs

TDK had a lot to showcase at CES 2023, with major announcements for developers and brands working on cutting-edge audio designs, including true wireless, and hearables. Expanding its important portfolio on the sound capture side, TDK announced availability of the InvenSense T5838 and T5837 MEMS microphones, which push the boundaries of microphone acoustic performance, power efficiency, and advanced feature sets all in small package footprints.

Also new is the SmartSound One Development Platform, a USB plug-and-play microphone interface module that enables rapid evaluation and development of TDK Analog and Digital MEMS microphone solutions.

But the major announcement from TDK at CES 2023 was the InvenSense VibeSense360 TWS solution suite. This is a comprehensive portfolio of TDK solutions comprised of hardware

and software dedicated to the TWS market to help solve some of the most complicated problems OEMs face, all while enabling state-of-the-art features. Among the new solutions included in this development package is a low-power six-axis IMU for accurate head tracking, an on-chip accelerometer that captures vocal vibration, and a machine learning on-chip solution with significant power savings on main processor/MCU.

TDK's VibeSense360 solution suite gives all TWS vendors the ability to include significant feature value in their end products by taking advantage of TDK's full array of expertise in this market.

Features that are enabled by VibeSense360 include "Always On" Spatial Audio, combining the world's lowest power six-axis IMU at 280µA with TDK's software suite to provide industry-leading head tracking accuracy. This is the most essential piece for a truly immersive 360° spatial audio experience. Another major feature is Transparency For Talk (TFT), where VibeSense360 determines when the user is speaking via vocal vibration detection and can automatically shift from ANC to transparency mode and pause any audio that may be playing.

VibeSense360 can differentiate between user audio and ambient noise, assisting the microphones in the TWS system by ignoring surrounding audio. This not only creates less false keyword triggers but provides secure keyword detection. And using TDK's machine learning algorithms built to run inside the IMU, VibeSense360 can train the device to recognize certain head gestures and assign commands or actions to allow the user's head motion to control the TWS system.

TDK confirms that the VibeSense360 TWS-M601 and TWS-M602 development solutions are sampling and will be available in 2023 alongside even more feature enablement from other sensors.

[www.invensense.tdk.com/tws](http://www.invensense.tdk.com/tws)



ic23

EDUCATION / JUNE 10 - 16  
EXHIBITS / JUNE 14 - 16  
ORLANDO, FLORIDA



**Brittany DiCesare, CTS**  
Client Success Manager  
Control Concepts, Inc.



**Steve Greenblatt, CTS**  
President  
Control Concepts, Inc.

**“InfoComm is the place to be for everyone in the AV industry. There’s no substitute for what can be accomplished in just three days by attending InfoComm.”**

**CLAIM YOUR  
FREE PASS**



**REGISTER NOW**

**infocomm**  
Infocommshow.org

# More Choices for Headphones Test and Measurement

By  
**J. Martins**

This article is a brief update on test and measurement solutions for headphones and earphones that we couldn't fit in our March 2023 issue. An update on solutions that perfectly complement our hearables-focused article.

Testing, measuring, and characterizing headphones and in-ear monitors is increasingly a challenge. While conventional headphones are all more or less understood and correctly characterized, they are now increasingly wireless, featuring sophisticated active noise cancellation (ANC) systems, microphones, and digital signal processing (DSP). They are also supporting binaural rendering of immersive multichannel and object-based spatial audio sources, which requires new methods of evaluation yet to be standardized. The growing range of features to be tested and possible variations of different modes supported by these products, increase exponentially the number of challenges in headphone test and measurement, be it for frequency response verification or tuning, or for quality-control checks at the production lines.

At the same time, the whole class of in-ear devices equipped with ANC and active hear-through systems, plus all the DSP that's associated with noise suppression, voice separation, personalization, and now all the real-time adaptive algorithms, poses even greater challenges. Never mind that true wireless earbuds are now the largest category by a wide margin in personal audio products. As our Market Update in this same issue discusses, from true wireless stereo (TWS) earbuds to hearing aids and now the converging class of hearables, these designs add a variable number of features that are increasingly user-dependent, since it's now possible to upload or activate new software directly on the device.

No wonder the audio test and measurement companies are keeping busy trying to catch up and support all these needs. There's even enough of a market for a new class of non-scientific products that meet a range of needs.

## Understand and Measure

Measuring and characterizing the audio quality of headphones, earbuds, and headsets presents unique challenges to the design

engineer in an R&D lab, the test engineer on a manufacturing line, or even the reviewer tasked with understanding and evaluating the unique features of new products.

Measurement systems designed to answer the needs of a wide variety of applications need to take into account that headphones can be circumaural designs that completely surround and isolate the outer ear (around the ear), supra-aural headphones that rest on the ear (on-ear), supra-concha for the smaller on-ear headphones (Walkman style), or intra-concha for models intended to rest within the concha cavity of the ear (a typical example being Apple's original wired Earpods).

From here, we evolve to in-ear or insert earphones that are intended to enter the ear canal partially or completely. In this category we can now find examples of loose-fit designs that simply "tunnel" sound to the ear canal (AirPods), others (including hearing aids) that are inserted in the ear canal but remain acoustically open or unsealed, and finally in-ear monitors and in-ear earbuds that are designed to completely close/seal the ear canal (AirPods Pro).

All those various types and designs will have a different sound when pressure is applied against the ear, causing variations in the fit, which should also be analyzed and compared. Now that we have earbuds and earphones with "transparency" modes, and even new passive designs that are intended to avoid occlusion (Sony LinkBuds), characterizations become even harder to perform and compare. As our colleague Brent Butterworth confessed when testing the latest Apple AirPods with its range of adaptive systems, its evaluation becomes an endless, almost impossible task.

Basically, all the audio test and measurement companies try to offer and continuously adapt solutions for these tasks and challenges. And that's probably why most manufacturers specialize in supplying a part of the test system and simply recommend using a third-party hardware or software. No single solution in the market is able to cover all the needs in the R&D lab or a

reviewer’s desk. Only when it comes to quality control or simple characterizations, where restricting the number of variables is decisive, does it become advisable to settle for a single contained solution (and test environment). Often, users simply have to do with the investments they already made and that is when it’s critical to understand its boundaries and limitations.

**Different Needs, Different Solutions**

Although not an immediate reference when it comes to headphone testing, there is no reason why a lab equipped with a Klippel Analyzer and software cannot adjust to those challenges. As Klippel explains, the key is to obtain meaningful and reproducible results. And the German company is known to conform its applications to the methods and recommendations that are established in relevant standards (e.g., IEC 60268-7). Not being a manufacturer of dedicated ear test fixtures, Klippel supports head-and-torso simulators, ear simulators and ear couplers (also called artificial ears) from GRAS, Brüel & Kjaer, HEAD acoustics, Larson Davis, Cry Sound, and others. The Klippel QC or R&D software frameworks, and a Klippel Analyzer 3 or a third-party audio interface can be used in quality control applications of headphones, headsets and in-earphones, for passive, digital (USB), and wireless products, including being able to perform noise attenuation tests in active systems.

Similar to loudspeaker measurement with microphones, ear simulators measure the acoustical response of earphones and headphones that are placed in or over the ear. For in-ear applications, the use of testing systems that emulate the dampening and impedance of the average human ear canal is essential.

As the headphones market continues to grow, and more companies are working on frequency response correction and emulation for headphones, the requirements for reliable and calibrated test fixtures are also increasing. Expanding its already extensive catalog of headphone test and measurement solutions, now as part of Axiometrix Solutions, GRAS Sound & Vibration has recently launched four new configurations for its GRAS 45CC Test Fixture. The new variants enable headphone and headset manufacturers to obtain accurate and reproducible test results, dual-channel testing, and multiple configuration options that are easy to use and will offer an efficient and inexpensive testing solution.

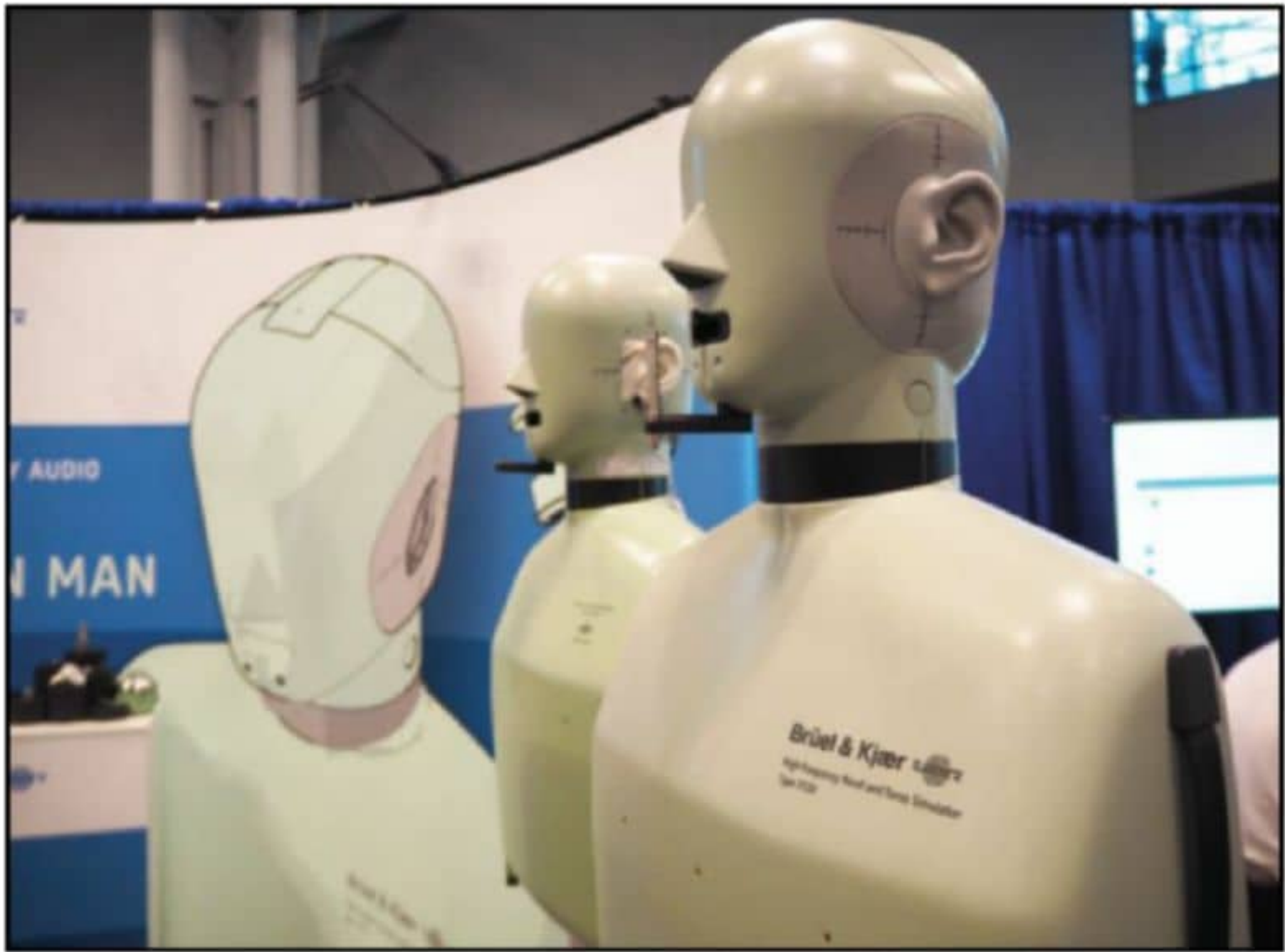
The GRAS 45CC platform is a desktop solution designed to be part of a complete test and measurement modular system, including high-frequency ear simulators and anthropometric pinnae, while meeting all applicable ANSI, IEC, ITU-T and other industry standards. The four new configurations to the 45CC test fixture series are ideal for research and development and for aligning testing on the production line.

With these new 45CC-14, 45CC-15, 45CC-16, and 45CC-17 configurations, GRAS offers a total of 10 variations that target different testing needs and are configurable for any headphone test setup and test environment, with a mouth simulator option for all of them.

Ease of use and functionality are maximized when working with combined GRAS and Audio Precision solutions, such as a measurement chain with TEDS-enabled ear simulators and a

GRAS 12Bx power module or an Audio Precision analyzer such as APx517 or APx1701. The standardized TEDS format also means that any test and measurement setup with analyzers supporting TEDS will benefit from using GRAS couplers, ear simulators, and head-and-torso-systems.

But not everyone needs to use a high-frequency ear simulator such as the GRAS RA0401/02 that extends the useful frequency



The High-Frequency Head-and-Torso Simulator (HF HATS) from Brüel & Kjaer (now called HBK—Hottinger Brüel & Kjaer) provides unprecedented realism for electroacoustic testing of the latest advanced headphones, hearables, and hearing aids. The Brüel Type 5128 HATS family supports testing according to all international standards, including evaluation of intelligibility in two-way communications in the presence of background noise and feedback.



The 45CC platform already has a solid footprint in the testing market and GRAS now offers four new versions that are ideal for test scenarios that require or benefit from realistic human-like acoustic loading and ear geometry when testing any headphone or earphone type (around ear, on ear, in ear, and insert).



The GRAS 43BA ear simulator is available in both a LEMO version, a CCP version, and a high sensitivity CCP version.



The popular Earphone Audio Response System (EARS) from miniDSP is increasingly used by enthusiasts who want to EQ their headphones based on measurements, which are then loaded to a headphone amplifier DSP capabilities or directly to a smartphone app.



The Audix TM2 Integrated Acoustic Coupler for In-Ear Monitors (IEMs) was designed to test the performance of in-ear monitors increasingly used by musicians on stage, but also by monitor engineers who need to check the exact mix that is being used by each performer.

range to 20kHz, and for quality control and evaluation, the GRAS 45CC and 43 AC/AG test fixture being excellent options can be simply not affordable for large operations where multiple units are needed.

## Simpler Needs, New Solutions

The fact that there are increasing applications where scientific instrumentation is not required, and a simple measurement of a system's response is all that it is needed, explains why Hong-Kong-based miniDSP has had so much success selling its Earphone Audio Response System (EARS).

Since the launch in 2018 (and review by *audioXpress*), the miniDSP EARS has proven its worth in bringing headphone measurement capability to "the masses." This affordable measurement system is fit for headphones and in-ears and uses the same technology of the company's UMIK-1 calibrated measurement microphone. Apart from its use by many amateur reviewers and enthusiasts who like to compare frequency response curves and create equalization filters, many EARS are also being used in the labs and studios as a binaural microphone with USB audio streaming.

Each pair of miniDSP EARS has two microphone capsules mounted in a minimal "dummy head." Measurement output is via a driverless USB connection, and power is also supplied via USB so there is no need for an external power supply. The supplied calibration file corrects for the response of the microphone capsule and the ear. Of course, the miniDSP EARS is not an industry-standard measurement system (it says so in the manual), but even expensive test systems can produce measurements that are different to each other depending on the type of ear simulator. The EARS is contributing to making a lot more users aware of the possibilities and limitations of these test systems, and have very likely encouraged many to acquire more serious measurement solutions.

But there are other requirements where a simple test and measurement can be enough, and that is why US-based microphone manufacturer Audix decided to create a dedicated division where it sells the TM2 Integrated Acoustic Coupler for In-Ear Monitors (IEMs). As the company explains, the TM2 was designed as a readily available measurement solution to complement software such as Rational Acoustic's SMAART or Studio Six Digital's Audio Tools, allowing monitor engineers to easily confirm the functionality of each performer's IEMs. The coupler will appeal also to IEM enthusiasts obsessed with comparing measurements.

Headquartered in Wilsonville, OR, Audix already sells the TM1, a 6mm pre-polarized omnidirectional condenser microphone designed for test and measurement applications—also available as a complete kit with calibration files (TM1 Plus). In 2020, the company, which has meanwhile been acquired by The Vitec Group, started shipping the TM2, approaching the functionality of a lab-type testing equipment into an integrated compact package. Featuring precision-machined brass and aluminum components, the TM2 is built for the road and includes adapters to fit a wide range of IEMs, including custom molds.

With its included adaptors to fit any size of in-ear monitor, the TM2 retails for \$1299, and Audix already offers the TM2SP, a stereo pair of TM2s with a mounting base. A calibration file is included with each unit, and the TM2 can also be connected to an acoustic calibrator for greater measurement precision.

For the more demanding applications, many companies, from Klippel to NTi Audio, which have solutions for headphone measurements and QC, recommend a third-party solution such as the Larson Davis Binaural Test Fixture Model AEC206. The test system supports circumaural, supra-aural, supra-concha, intra-concha, and insert-type headphones. Perfect for tasks such as performing quality assurance testing on an assembly line, the Model AEC206 comes preassembled with IEC 60318-4 (711) occluded-ear simulators and TEDS-compatible preamplifiers that provide electronic calibration information. Additionally, calibration is done with a custom adapter, which is included for the Larson Davis Model CAL250 acoustic calibrator or similar device.

NTi Audio offers an almost self-contained solution for headphones and earphones quality testing combining its FX100 Audio Analyzer with the Larson Davis AEC206 test fixture. The FX100 is one of the most powerful audio analyzers in the market, with the advantage that it can combine different connectivity and interface modules, allowing to connect the two microphones of the binaural test fixture.

For these applications, the FX100 Audio Analyzer is controlled by the FX-Control Software, which allows users to select and start measurement sequences using typical glide sweep, noise, or sinusoidal signals. These can be transmitted from the FX100 to wired headphones, or wirelessly using the NTi Audio Bluetooth Box. The sound emitted by the headphones is then picked up by

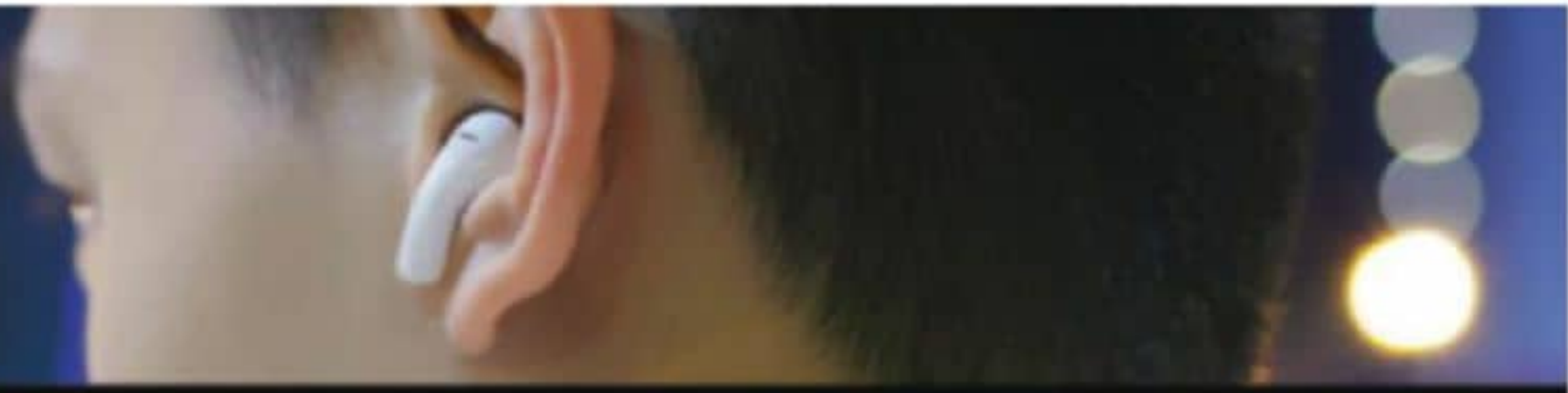


The NTi Audio FX100 Audio Analyzer combined with the Larson Davis AEC206 Headphone Test Fixture is a practical and compact solution for quality assurance testing of headphones. The Liechtenstein-based manufacturer can supply and support the complete systems.

the built-in microphones, and passed to the FX100 Audio Analyzer. The FX100 is able to measure sensitivity, frequency response, distortion, and phase response, as well as active, passive, and total sound reduction for models with ANC.

**The Software**

Some of the most common applications for headphones measurements are simply associated with calibration of the response, as popularized currently by Sonarworks' SoundID Reference software that is used by musicians and music producers.



Consumer demand for advanced playback and voice features is growing across Bluetooth products

Get the complete solution on  **Audio Weaver** development platform



**DSP**  
Concepts

# BLUETOOTH AUDIO

- Highly efficient code for supported SoCs
- Fully modifiable example layouts
- Support for Bluetooth LE Audio

- Wireless tuning and optimization
- Best-in-class 3rd party IP
- Real-time system debug





THD+N results using the FastSweep stimulus, available with APx V8 in Continuous Sweep, Acoustic Response, and Loudspeaker Production Test modes.

In this case, everything is done in the software, with the user wearing the headphones/IEMs and the sound source being fed from the computer running the software, or dedicated equipment that runs the embedded SoundID software. The evaluation is made by the user listening to the results with real music, although the software provides a basic visual guide of the “response”—a simplified overview with smooth lines that are easy to interpret.

Of course, there is software, including almost free software that provides actual measurement capabilities, such as the popular Room EQ Wizard (REW), which can even rival professional software in the quality of the graphics and some of its unique tools. As long as the user doesn’t mind tweaking with PC/Windows, and doing things one by one not knowing if it can repeat exactly the same measurements over time, that is perfectly fine.

For those applications that require the most advanced and reliable tools in the lab or manufacturing, where actual measurement graphs need to approach a consistent, data-rich representation of the signal being measured, there is no question that using software from Audio Precision, Listen, or HEAD acoustics is the only advisable option. And the good news is that it is now possible to use the software with third-party hardware interfaces (e.g., Echo Test + Measurement) and more affordable USB audio interfaces or sound cards.



Multichannel Real Time Analyzer (RTA), upgraded with real-time calculations, is now able to display multiple channels simultaneously, and offers real-time calculation and display of channel addition, subtraction, maximum, minimum, and power average.

Audio Precision recently released version 8 of its APx500 Measurement Software, introducing significant test time savings and much faster reporting of important measurements in the upgrade. Version 8 also features software performance optimizations that reduce unnecessary computation and speed up sequence execution. This results in much faster reporting of important measurements, such as THD+N.

This update follows the introduction of APx500 Version 7 in November 2021, which was a major update adding new enhancements suited to a wide range of audio test applications and the introduction of software subscriptions. V7 enabled the use of FastSweep in open-loop testing scenarios, which is essential in current wireless audio products.

The company is committed to growing the cadence of enhancements to its APx500 software, and Version 8 was announced precisely a year later. It gives users access to THD+N results using the FastSweep stimulus in Continuous Sweep, Acoustic Response, and Loudspeaker production test. As its name implies, this extremely fast stepped frequency signal sweep offers several advantages over a logarithmically swept sine (chirp) signal, including an ability to sweep from low-to-high or high-to-low frequency, a lack of transient ripples at low or high frequencies, support for measuring harmonics above the 20th, and precise control of the number of discrete points used in measurements.

Listen also recently updated its SoundCheck audio test software, now in version 20. Listen traditionally introduces a new version of SoundCheck each year, and the release of SoundCheck 19 in 2021 included a much tighter integration with Listen’s newest AmpConnect 621 hardware interface, supporting multichannel, multifunction, and high-resolution audio functionalities.

Traditionally, SoundCheck was always a very powerful solution for loudspeaker testing, but the latest update expands the features required for audio measurement and full characterization of smart speakers, automotive infotainment systems, headphones and hearables, and other voice-activated and multichannel devices, including communications testing.

The multichannel Real Time Analyzer (RTA), initially introduced in 2020, has now been upgraded with real-time calculations, which is useful for real-time observation of ANC, or validating the headphones seal on a test head prior to measurement by comparing left and right earphone response, and more.

Live curves from the RTA are displayed in the memory list and can be dragged and dropped into any graph for easy comparison to limits and reference curves. This is useful for quick comparisons to reference standards or golden units.

There’s also been a complete overhaul of the signal generator, which can now play any wav file, any memory list file, and any complex waveform created by the stimulus editor. The level can be referenced to Peak, decibels (dB), or RMS. The signal can be equalized in real time, and custom EQ curves applied, which is useful when using a non-flat source such as a mouth simulator.

Of great interest for perceptual measurements of speech degradation in communications applications is a new optional module with Perceptual Objective Listening Quality Analysis (POLQA) right into SoundCheck. This module may be used to

assess the impact of noise reduction algorithms, evaluate Bluetooth degradation due to packet loss, or to analyze distortions introduced into the audio path. Naturally, it can be used within sequences to accelerate and simplify test procedures.

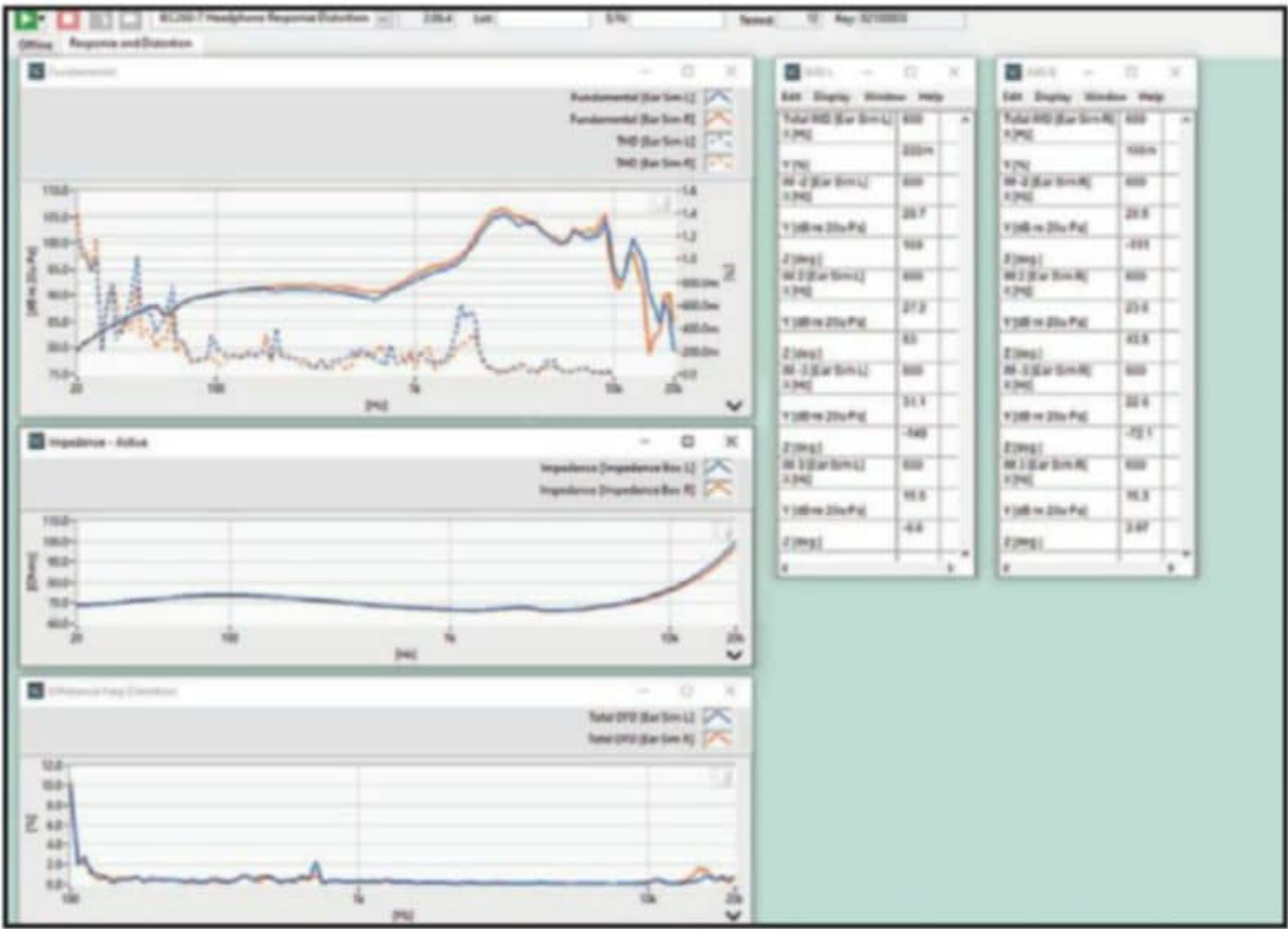
And while SoundCheck has offered batch processing of collected data since 2012, version 20 extends this with the ability to automatically load and process external data. This is useful in any application where it is desirable to automate the testing of a large number of external data files via a sequence, for example wav files of speech recordings. This has many uses in communications testing, for example in testing the Mean Opinion Score (MOS) of degraded speech files.

For communications testing, users can now explore the new stimulus steps and post-processing capabilities such as a silence stimulus step, and RMS level versus time and histogram post-processing steps. This allows measurement of doubletalk and other speech transmission parameters.

Testing Structure-Borne Sound

Responding to the growing popularity of bone-conduction headsets, and the use of bone conduction sensors integrated in TWS earbuds to improve source separation, HEAD acoustics created a complete test solution called ViBRIDGE.

The German manufacturer of audio test and measurement solutions is the first to come to market with a complete suite to



Using Listen’s SoundCheck to characterize the performance of headphones according to the IEC-60268-7 international standard. SoundCheck provides separate sequences, each designed to measure specific characteristics.

support product development combining structure-borne sound. Such products are able to perform better separation of the user’s own voice from the noise or concurrent talkers in the surrounding environment, improving noise cancellation and speech quality in noisy environments.

Increasingly used for advanced hearing assistance and audio augmentation solutions all processed on-device, the industry’s ability to evaluate device performance using bone conduction was limited since no appropriate test equipment and methods were commercially available until now. HEAD acoustics conducted

**AXPONA**  
AUDIO EXPO NORTH AMERICA

APRIL 14-16, 2023  
Renaissance Schaumburg Hotel and  
Convention Center, Schaumburg, IL

Find us in room No. 584  
Experience the sound of SATORI TeXtreme!

Find us on:

SB Acoustics

sbacoustics

sbacoustics\_

SB Acoustics

sbacoustics.com

audioxpress.com | April 2023 | 25

thorough measurements of the individual spectra of many talkers' air-borne and structure-borne sound. This allowed the company to create an overview of the spread of individual differences. These studies were the foundation to derive average transfer functions that can be used to create a realistic structure-borne sound simulation. Structure-borne sound simulation is crucial when testing this type of design, and its evaluation and optimization for different conversational situations can only be achieved with the complementing simulation of air-borne and structure-borne sound generated by the human voice.

HEAD acoustics created an integrated solution in its ViBRIDGE technology, combining lab-based background noise simulation and dedicated test suites for its ACQUA analysis system, which are available as an upgrade. To create an "all-in-one" solution for testing, optimization, and validation of devices using bone-conduction sensors, HEAD acoustics combined a bone conduction

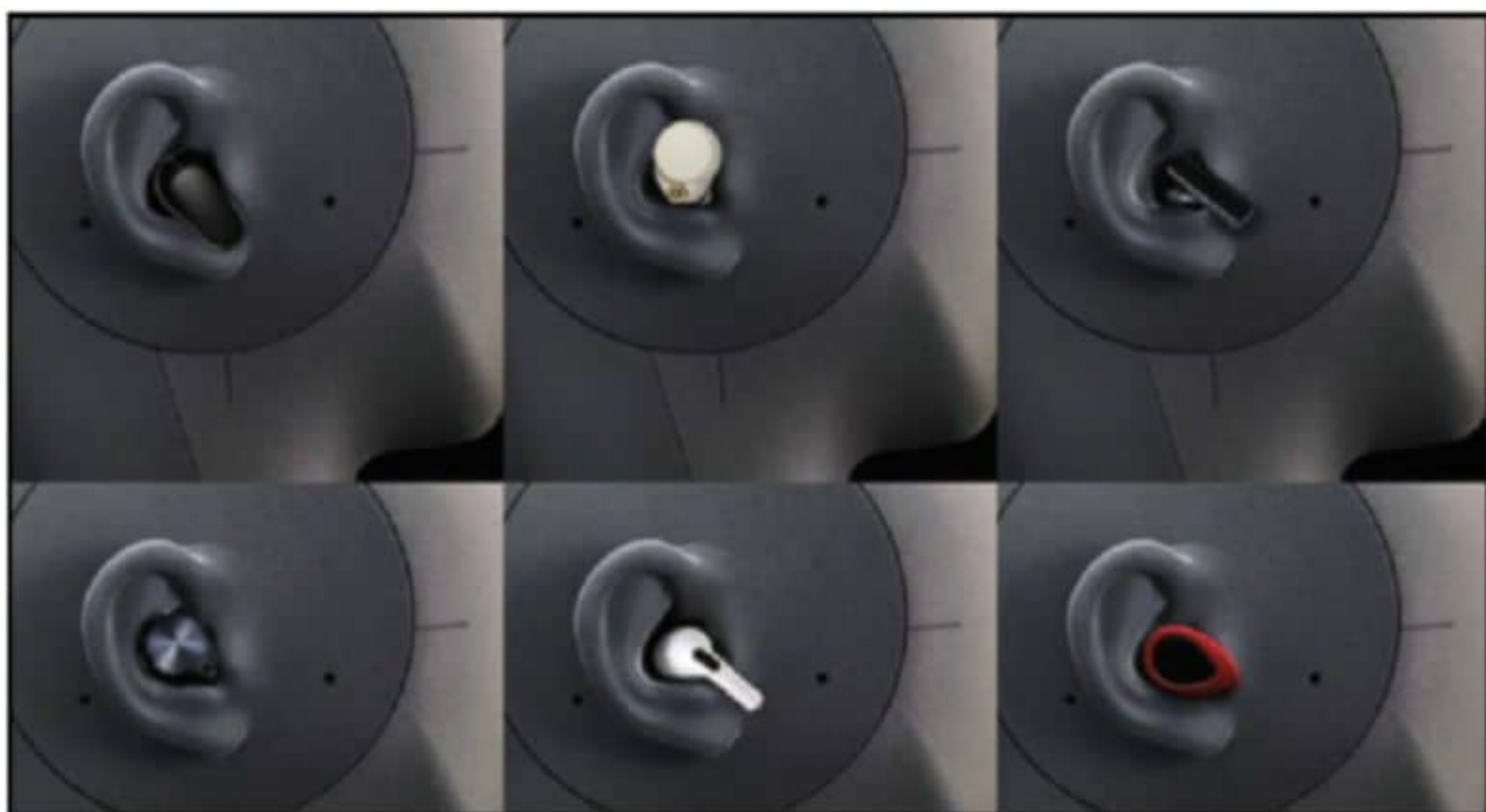
actuator integrated into its HEAD acoustics HEL/HER 4.4 Artificial Ear, in order to induce structure-borne sound. This integration is now available as a standard configuration, with the launch of the new Head-and-Torso Simulator HMS II.3 ViBRIDGE.

The ViBRIDGE solution also combines HEAD acoustics' ACQUA measurement and analysis software, and its labCORE modular interfaces to generate the equalized signal for both the Artificial Mouth and the HEL/HER 4.4 ViBRIDGE pinnae. The combined solution allows developers to have a realistic evaluation solution of the human structure-borne voice signal for testing suitable in-ear designs.

Using high-precision electro-mechanic actuators, the new HEL/HER 4.4 ViBRIDGE pinnae realistically simulates bone-conducted near-end speech to any intra-concha- or insert-type device with integrated bone conduction sensors. This novel measurement technology offers reliable and proper testing of such devices, including evaluating echo cancellation and double-talk performance in sending direction.

HEAD acoustics also offers the valuable Multi-Dimensional Audio Quality Score (MDAQS), a new binaural perception-based software tool that replaces human evaluation with novel metrics in the assessment of audio quality. This is a much-needed approach for characterization of headphones and earphones and, according to HEAD acoustics, could be instrumental for audio device development and control/adjustment of audio quality, comprising the key criteria for human-based, perceived audio playback quality—timbre, distortion, immersiveness—along with an overall mean opinion score (MOS) value to allow for quick assessment and good comparability of listeners preferences.

MDAQS runs as an option in the ACQUA test software, the powerful speech and audio quality measurement and analysis software from HEAD acoustics, and can be fully integrated in its existing measurement process (a topic that *audioXpress* intends to detail in an upcoming dedicated article).



HEAD acoustics created ViBRIDGE, an integrated solution for testing, optimization, and validation of headphones and in-ear devices using bone-conduction sensors.

## Resources

B. Fallon, "Headphone Testing (Part 2): Digital Headphones (USB & Bluetooth) and Noise-Canceling Headphones," *Voice Coil*, April 2012, <https://audioxpress.com/article/Headphone-Testing-Part-2-Digital-Headphones-USB-Bluetooth-and-Noise-Canceling-Headphones>

H. W. Gierlich, "The Benefits of Using Structure-Borne Sound in Headset Signal Processing," *audioXpress*, January 2023, <https://audioxpress.com/article/practical-test-measurement-the-benefits-of-using-structure-borne-sound-in-headset-signal-processing>

Klippel, Quality Control of Headphones Application Note AN73 for the Klippel Analyzer System, [www.klippel.de](http://www.klippel.de)

S. Olive, "The Perception and Measurement of Headphone Sound Quality: What Do Listeners Prefer?" *Acoustics Today*, Spring 2022, <https://acousticstoday.org/wp-content/uploads/2022/03/The-Perception-and-Measurement-of-Headphone-Sound-Quality-What-Do-Listeners-Prefer-Sean-E.-Olive.pdf>

C. J. Struck, "Electroacoustic Measurements of Headphones," *Voice Coil*, July 2019, <https://audioxpress.com/article/electroacoustic-measurements-of-headphones>

S. Temme, "Headphone Testing (Part 1): The Basics," *Voice Coil*, December 2011, <https://audioxpress.com/article/Headphone-Testing-Part-1-The-Basics>

S. Yaniger, "Examining the miniDSP EARS Headphone Test Fixture," *audioXpress*, March 2019, <https://audioxpress.com/article/fresh-from-the-bench-examining-the-minidsp-ears-headphone-test-fixture>

## Sources

Audio Precision, Inc. | [www.ap.com](http://www.ap.com)

Audix | [www.audixusa.com](http://www.audixusa.com)

Axiometrix Solutions | [www.axiometrixsolutions.com](http://www.axiometrixsolutions.com)

GRAS Sound & Vibration | [www.grasacoustics.com](http://www.grasacoustics.com)

HEAD acoustics | [www.head-acoustics.com/vibridge](http://www.head-acoustics.com/vibridge)

Hottinger Brüel & Kjær | [www.bksv.com](http://www.bksv.com)

Larson Davis | [www.larsondavis.com](http://www.larsondavis.com)

Listen, Inc. | [www.listeninc.com](http://www.listeninc.com)

miniDSP, Ltd. | [www.minidsp.com](http://www.minidsp.com)

NTi Audio | [www.nti-audio.com](http://www.nti-audio.com)

# MENLO SCIENTIFIC

Engineering  
**Success**  
since 1983

## Engineering Consulting for the Audio Industry

We Know

- Product Development

We Provide

- Design and Technical Support

We Deliver

- Liaison, Sourcing, and Guidance

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

MENLO  
SCIENTIFIC

Richmond, California, USA

- tel 510.758.9014
- [mike@menloscientific.com](mailto:mike@menloscientific.com)
- [www.menloscientific.com](http://www.menloscientific.com)



# Remote Recording in the Rock Era

## David Hewitt's *On the Road*

Review by

**Scott Dorsey**

There were many well-known engineers who made their livings doing live recordings of rock concerts during the 1970s and 1980s. One of the most popular engineers was David Hewitt, who wrote a book about his experiences. After a thorough read, Scott Dorsey shares his review of the book.

**Title:** *On the Road: Recording the Stars in a Golden Era of Live Music*

Author: David W. Hewitt

ISBN-10: 1493056174

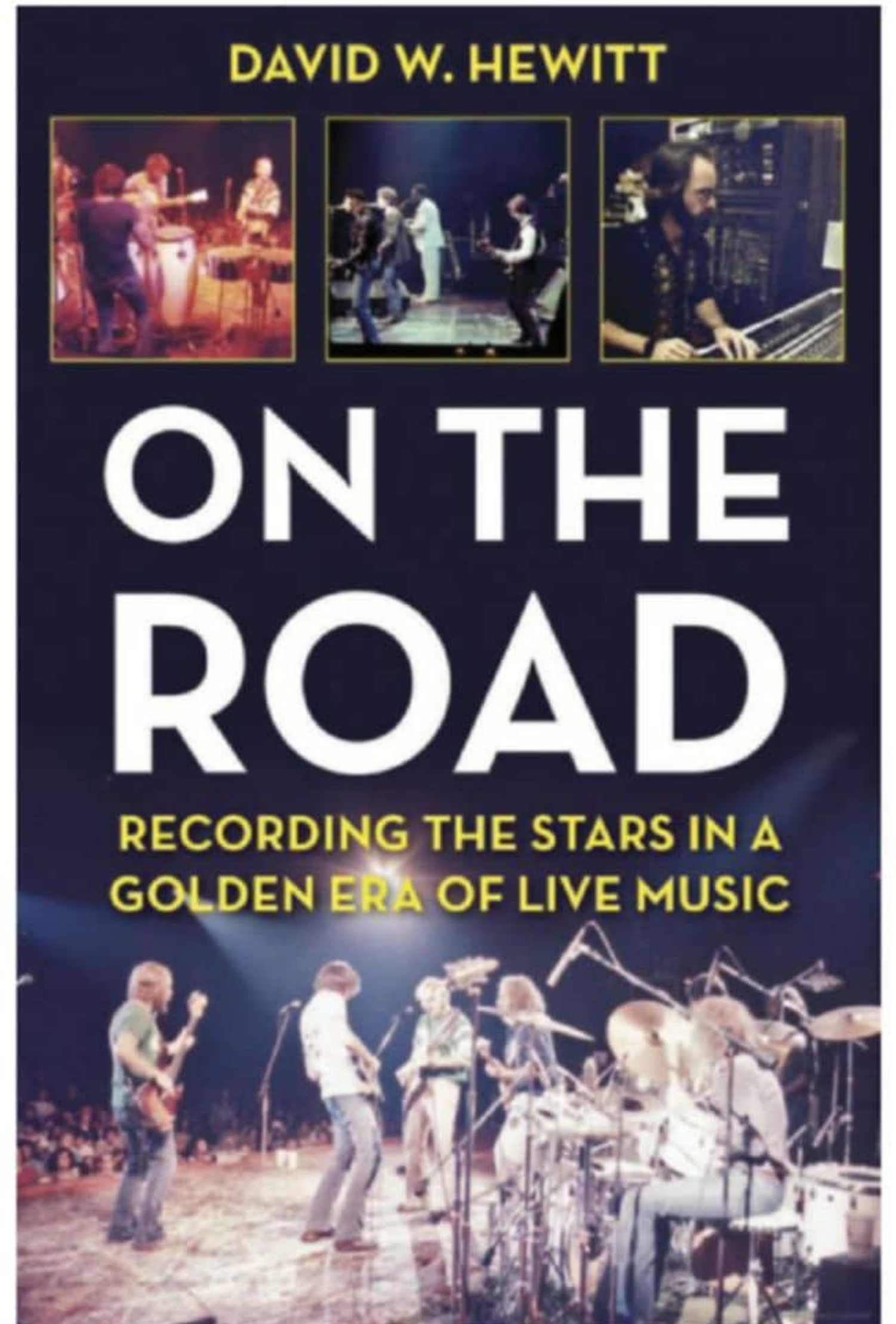
Publisher: Backbeat (November 15, 2021)

Pages: 288

The 1970s and 1980s were great times for live music, and so they were great times for live albums. Technology had reached the point that it was finally possible to get good sound in large auditoriums with live bands, and soon it became possible to get good recordings in those situations.



This image from a Shure microphone ad shows David Hewitt in the famed "Black Truck," the first truly modern mobile recording studio, with which he captured many incredible live performances for artists from Aerosmith to Pink Floyd to ZZ Top.

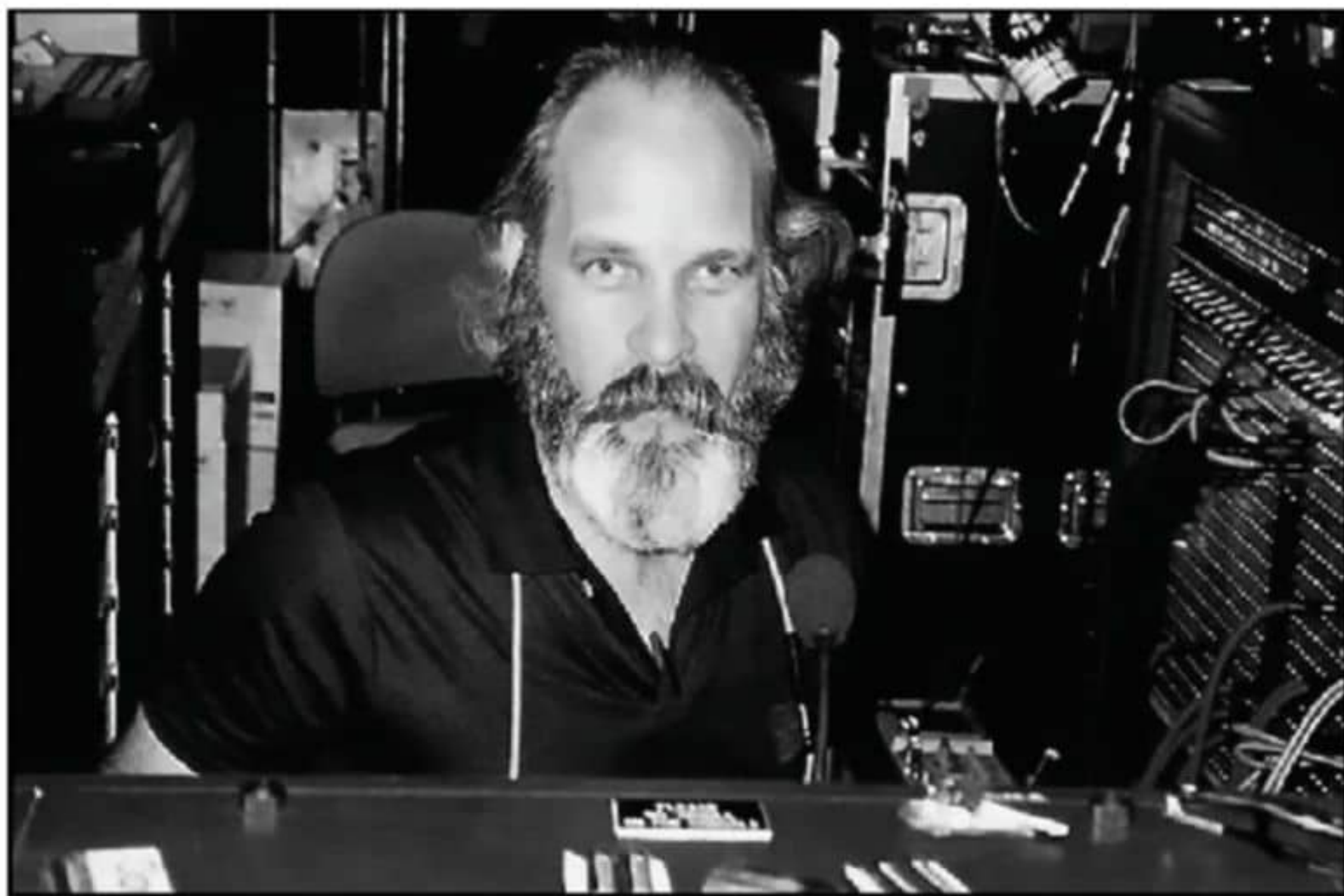


There were many well-known engineers who made their livings doing live recordings of rock concerts in that era, from Wally Heider and Kooster McAllister to the aptly named Steve Remote. But one of the most popular engineers was David Hewitt.

Hewitt worked for 50 years in the industry, starting with Regent Sound Studios in Philadelphia, PA, moving to the Record Plant Remote truck out of Los Angeles, CA, working for many years in the business and eventually buying a truck from the Record Plant and operating a series of remote trucks of his own.

He worked on the Super Bowl Halftime Show, Arthur Fiedler and the Boston Pops, and The Concert for New York City after 9/11. He recorded concerts with Buddy Rich, Wynton Marsalis, Eric Clapton, Neil Young, and stand-up performances by Father Guido Sarducci. When Billy Joel and Nelson Mandela were on stage at Yankee Stadium, David Hewitt was in the truck, just like he was at the Beverly Sills Opera Gala, and for many years of Academy Awards.

Before I talk about what his book, *On the Road: Recording the Stars in a Golden Era of Live Music*, is, let me say what it's not. It's very much lacking in technical details and in discussions about process and philosophy of remote recording. I would like to have seen a lot more of that, not just for the reasons that all *audioXpress* readers probably have, but also because I came up in a very different world of remote recording for acoustic music. I would



like to have been able to compare my philosophy of recording orchestras and his. I would also like to have seen him write more about the early digital era, because there isn't a lot about that and people who live in the modern digital workstation world might have some notions about a world of analogue tape machines... but the world of digital tape machines is not something people talk about.

But what IS it? It's a memoir, as if Hewitt had gone through his journal of concert recordings over the years and tried to write about each one of them and what he remembers about the event. It's not all that well organized, not exactly chronological, and

not exactly by genre but a little of each. Some of the events he talks about extensively, while others (including some of the ones I'd be most interested in such as the jazz gigs of the early 1980s) he just mentions in passing.

If you're at all curious about the concert scene in the 1970s, 1980s, 1990s, and 2000s, you will find this book interesting. You might find it brings up more questions than it answers, but that's not a bad thing because it means it got the reader thinking about concert recording work.

Yes, it would have been nice to have had more technical discussion and more details about how things have changed over the past few decades, but there's a little bit of that there to start off. And it does seem like Hewitt goes out of his way not to talk about *Frampton Comes Alive*, which is probably the most famous and possibly the most controversial album he ever worked on. How much of it is actually live? He's not telling here.

But he is talking about the field hockey game played by members of the Who against a high school girls' field hockey team that was in the same hotel. It is filled with interesting little anecdotes about a touring era that is gone today.

You should get this book and keep it on the coffee table or in the bathroom because it's a fun book to read and you can read it in little bits and pieces. If anything, it will introduce you to new recordings that you didn't know existed, and it has definitely added to my long list of records to buy. For more information, visit <https://davidhewittontheroad.com> 



David Hewitt has been a legendary name in the recording industry and pro audio business since 1970, having recorded and engineered an incredible array of talented artists in both live and studio settings across the world.

## About the Author

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



# NEW MICROPHONE TECHNOLOGY —FLAT RESPONSE CURVE, SET SENSITIVITY

**It's been too long since the last leap in microphone technology.**

WORDS BY:  
**KIM BOLDT**

The new EQ 40PM Miniature Production Line Microphone with EQset™ from GRAS is an industry-standard sized ¼" 20 kHz pressure microphone. But while EQ 40PM has the same dimensions as a typical production line test microphone, key elements differentiate it. This microphone was specifically designed for economy and performance, designed to be extremely environmentally stable, and is the first microphone to include EQset technology, which provides a uniformly fixed sensitivity and a flat response curve across all frequencies in each microphone—out of the box.

The output of every EQset microphone is, in a sense, equalized and provides accurate, valid data without need for additional corrections or calculations. This reduces measurement uncertainty and greatly simplifies microphone setup, monitoring, and replacement. The sensitivity deviation between individual EQ 40PM microphones is  $\pm 0.2$  dB (see the figure showing a random sampling of 100 EQ 40PM microphones), and the flat response curve results in a tolerance window of only  $\pm 0.5$  dB from 20 Hz to 10 kHz and  $\pm 2$  dB from 10 to 20 kHz—compared to traditional production line microphone deviation of  $\pm 3$  dB from 20 to 50 Hz,  $\pm 2$  dB from 50 Hz to 10 kHz and  $\pm 3$  dB from 10 to 20 kHz.



For many years, transducer electronic data sheets (TEDS) were the premium solution for shortening setup time and acquiring the best data. TEDS provided the information needed to speed up calibration, data correction, and microphone identification. But TEDS is a cost-prohibitive solution, and the data acquired via TEDS microphones must still be corrected either by pre-processing, requiring analyzer software capable of the task, or post-processing, a time-consuming task.

EQset microphones provide output at the correct level and do not need pre- or post-processing software. This eliminates the time consumed in processing data and a potential step where human error can shine. Lastly, EQset microphone response curves and sensitivities are so close (see figure) that they can be interchanged without affecting the data. Without a need to calibrate, correct, or identify individual microphones, there is no need for TEDS.

This enables EQ 40PM with EQset to be used effectively in a wide variety of production line setups: from economic systems with microphones powered directly by an audio sound card using 48 V phantom power to large complex systems. While EQset microphones do not use TEDS, they can



*”EQ 40PM will raise the bar for the quality of measured data on production lines. It eases the setup process and will substantially decrease the cost of the overall measurement chain”*

Says **Dan Vandenberg**

Global Strategic Account Manager at GRAS



*”EQset will set a new standard in acoustic measurement because its ease of use and consistency”*

Says **Rémi Guastavino**

Director of Product Management at GRAS



*”GRAS has taken a major step in technology that reduces the complexity of testing by focusing on practicality”*

Says **Niels Kjærgaard**

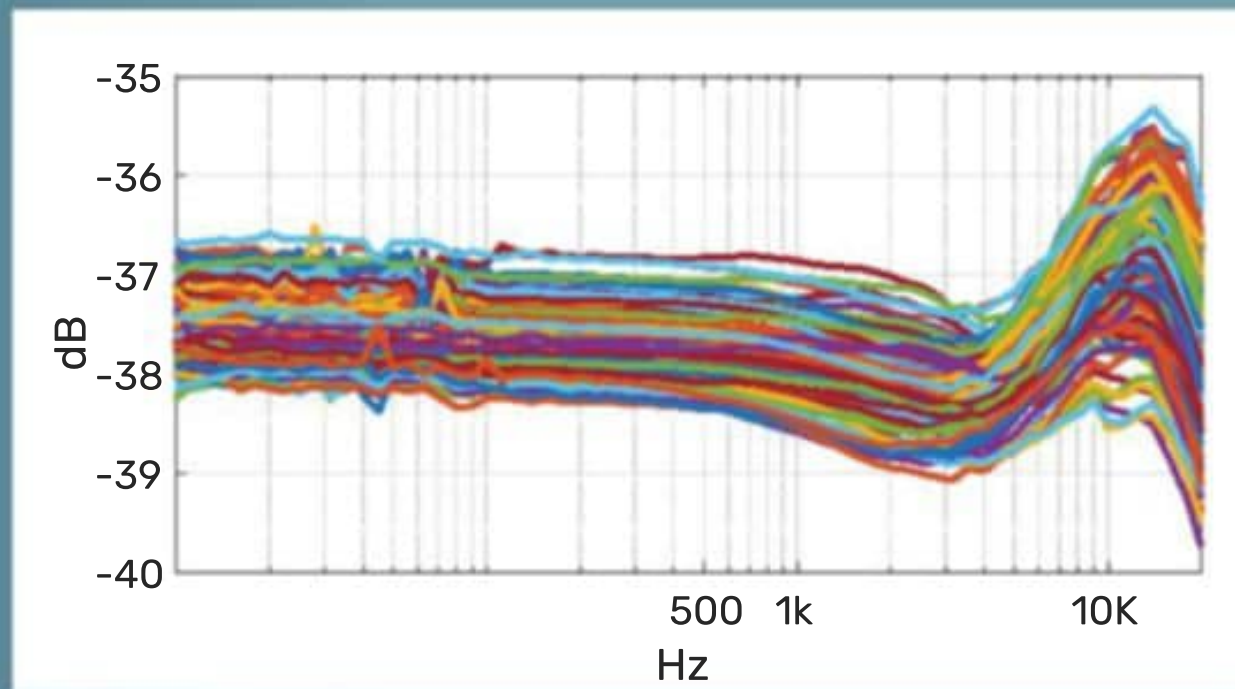
VP Business Development at GRAS

## RANDOM SAMPLE OF PRODUCTION LINE MICROPHONES



*EQ 40PM with EQset technology*

VS



*Traditional production line microphones*

be used in any CCP-based system, so there is no need to acquire new equipment or retool existing CCP-based systems in order to incorporate them.

A key element of EQ 40PM design regarding production lines is its extremely high environmental stability. The environmental stability ( $< 0.3$  dB) provided by the new design of the EQ 40PM reduces sensitivity variation over the course of a normal day to such an extent that, under typical production line conditions, compensation is completely unnecessary. The environmental stability and EQset come together to provide excellent data and mitigate a

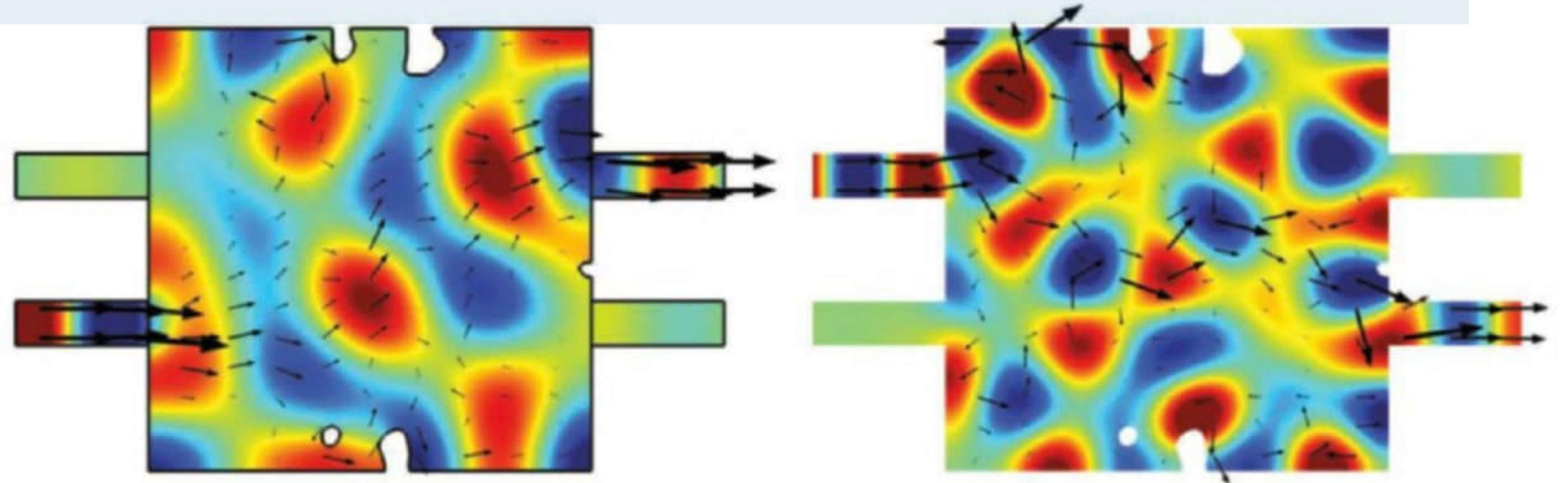
significant risk in production line testing—the risk of false passes and false failures. The uniformly set and fixed gain/sensitivity, the flat response curve, and resistance to fluctuations in temperature and humidity greatly reduce the chance that good units will fall outside the passing window or that bad units will fall inside due to uncertainty in sensitivity or frequency curve adjustments.

For more information, please visit [GRASacoustics.com/EQSET](https://GRASacoustics.com/EQSET).



# Acoustic Topology Optimization for Vibroacoustics Applications

By  
**René Christensen**



Acoustic topology optimization is one of the most interesting new technologies in vibroacoustics, and is a powerful tool that enables engineers to find optimal solutions to problems related to their applications. In this article, several acoustic topology optimization examples are shown.

Engineering is to a large degree the combined output of education and experience. If you are designing transducers, you need academic knowledge about acoustics, electronics, transductance principles, modeling and so on, but you also partly rely on the experience you have accumulated of a more experimental and empirical kind and you use solution strategies from previous projects to solve the problems at hand.

Oftentimes, however, the combination of academic and experimental knowledge is not enough to clear a direct path toward a product or to solve a problem in a straightforward manner. That can be the case when there are many degrees of freedom, i.e., parameters that can be changed in a system and when several targets are sought at once. In such cases, changing one parameter at a time will not be feasible for finding an optimal solution, as the optimum value of one parameter will depend on several or all of the other parameters. A trial-

and-error approach could then be used, but that can cause new issues, as you can easily end up spending a lot of time without getting any closer to a solution. More systematic approaches such as Design of Experiments (DOE) [1] may be helpful if the number of parameters is limited, but many design problems have so many design variables that a simple combinatorics analysis will show that it could literally take years to explore the full solution spaces, even when using simulations instead of experiments.

In cases with a large solution space and several complex targets, it is advantageous to turn to “formal optimization.” By formal, I mean a procedure driven by mathematics with one or more objectives (targets), and one or more constraints, typically defined within some finite element modeling framework. With the optimization problem formulated via mathematics, a computer algorithm can be utilized to solve the problem.

For any type of optimizations, one must have an objective function defined via a mathematical expression that describes the desired targets. This can be quite difficult to achieve. It could be that you are interested in having a flat on-axis sound pressure, and a particular directivity index, alongside one or two more targets, and some of these may be easy enough to formulate in a sentence but not as mathematics. Once defined as an expression, the objective function can be minimized or maximized, such as the difference between some target value  $f_{\text{target}}$  and a variable of interest  $f(\xi)$  being a function of the design

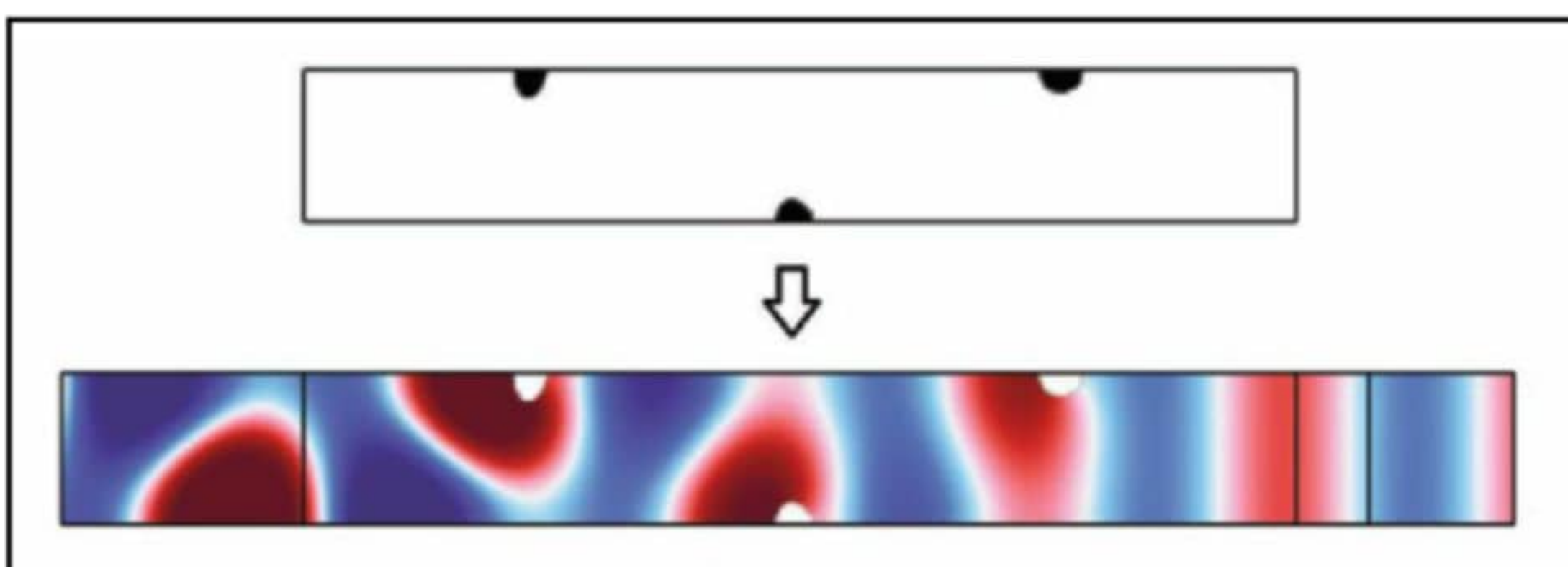


Figure 1: A topology optimization domain with the black parts ( $\xi=1$ ) indicating a “solid” and the white parts ( $\xi=0$ ) being air. Below, the sound field is shown with the solid parts modifying the geometry.

variable  $\xi$  to be explained shortly. A minimization of an objective function could in principle look like Equation 1:

$$\min_{\xi} \left( \left| f(\xi) - f_{target} \right| \right) \quad (1)$$

For wave propagation problems, the objective function will typically be a function of frequency, and if several objectives are to be met, they must be weighted with their individual importance by the engineer.

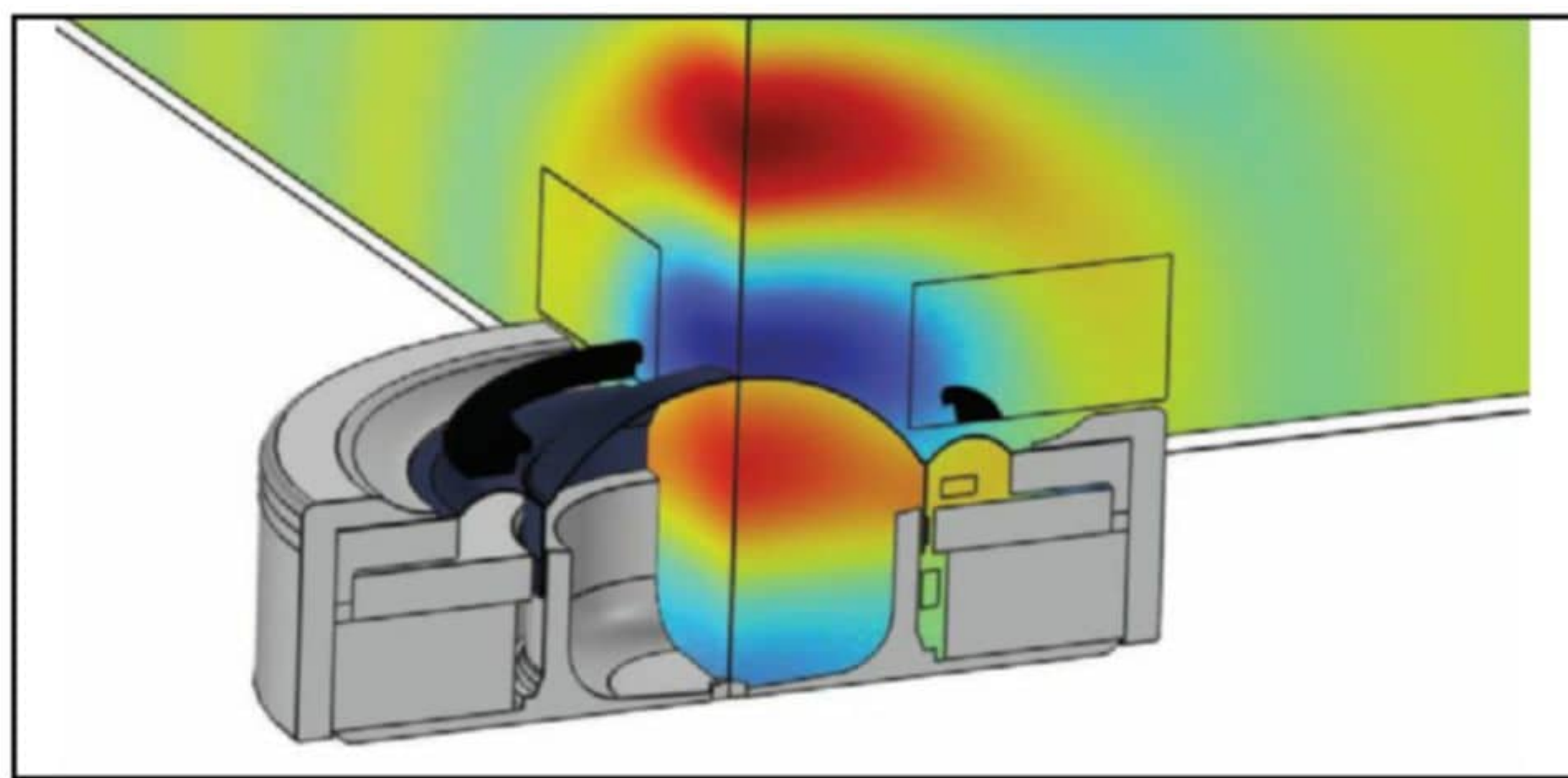
With a proper target function defined, there typically are also certain constraints to consider. That could be how much of the domain assigned for optimization can take on a “solid” form as shown shortly, or other constraints involving the acoustic variables. With the targets and constraints defined mathematically, and the general finite element setup in place, the optimization can be run and while there is no guarantee for finding an “absolute” or global minimum, the optimization routine should at least find a local optimum.

For so-called topology optimization problems, a design variable  $\xi$  is defined via a field that varies in an assigned optimization domain [2]. It can take on values between zero and one, and while it is a continuous field, the optimization routine is set up to prioritize values being either close to zero or close to 1. This will ensure a more binary field where a 0 will be interpreted as being air and a 1 is interpreted as a rigid solid mechanics structure, although in the simulation it is a fluid; a fluid with very high density and ditto bulk modulus; the “stiffness” of the fluid.

For a design variable field that is close to binary across the domain, a solid mechanics structure of the same shape as the field can be expected to result in similar optimized results, but since it is generally not possible to have a completely binary field, the resulting geometry should always be tested by exporting the field and then import it interpreted as a solid structure in a subsequent simulation. In **Figure 1**, an example of such a two-dimensional design variable field is shown with black parts being interpreted as solid parts. The sound field can also be seen, revealing the objective function used in the simulation; to convert a non-planar sound field from the left to a planar one on the right. This already alludes to what can be achieved with optimization and how this very non-trivial problem can be solved with the proper optimization setup.

## Topology Optimization Examples

Several acoustic topology optimization examples are shown now, where the simulation software COMSOL Multiphysics was used.



### Tweeter Phase Plug

The first example is taken from a COMSOL Conference paper [3]. A tweeter geometry was made with realistic parameters and materials, and its resulting sound pressure level (SPL) is sought to be modified via acoustic topology optimization. A domain in front of the tweeter was assigned as the optimization domain, and an objective related to the on-axis SPL was defined mathematically. The resulting “phase plug” geometry is shown in black in **Figure 2**, where the optimization routine has found that this new geometry placed close to the tweeter “surround” will modify the SPL to best meet the set objective.

The optimization field for the design variable was exported as a geometry, and a new simulation was run to test the topology optimized geometry. The on-axis SPL before and after adding the phase plug are shown in **Figure 3** and the overall level is flatter and higher after the optimization.

It should be noted that while your objective(s) may be met to some acceptable degree, one should always investigate if other quantities have changed for the worse because of the optimization. In this case it would be informative to look at radiation patterns and off-axis responses before and after optimization to check that any changes are acceptable. If something has changed negatively, it may be necessary to reformulate the objectives and/

Figure 2: The tweeter simulation setup with the optimized phase plug is shown in black.

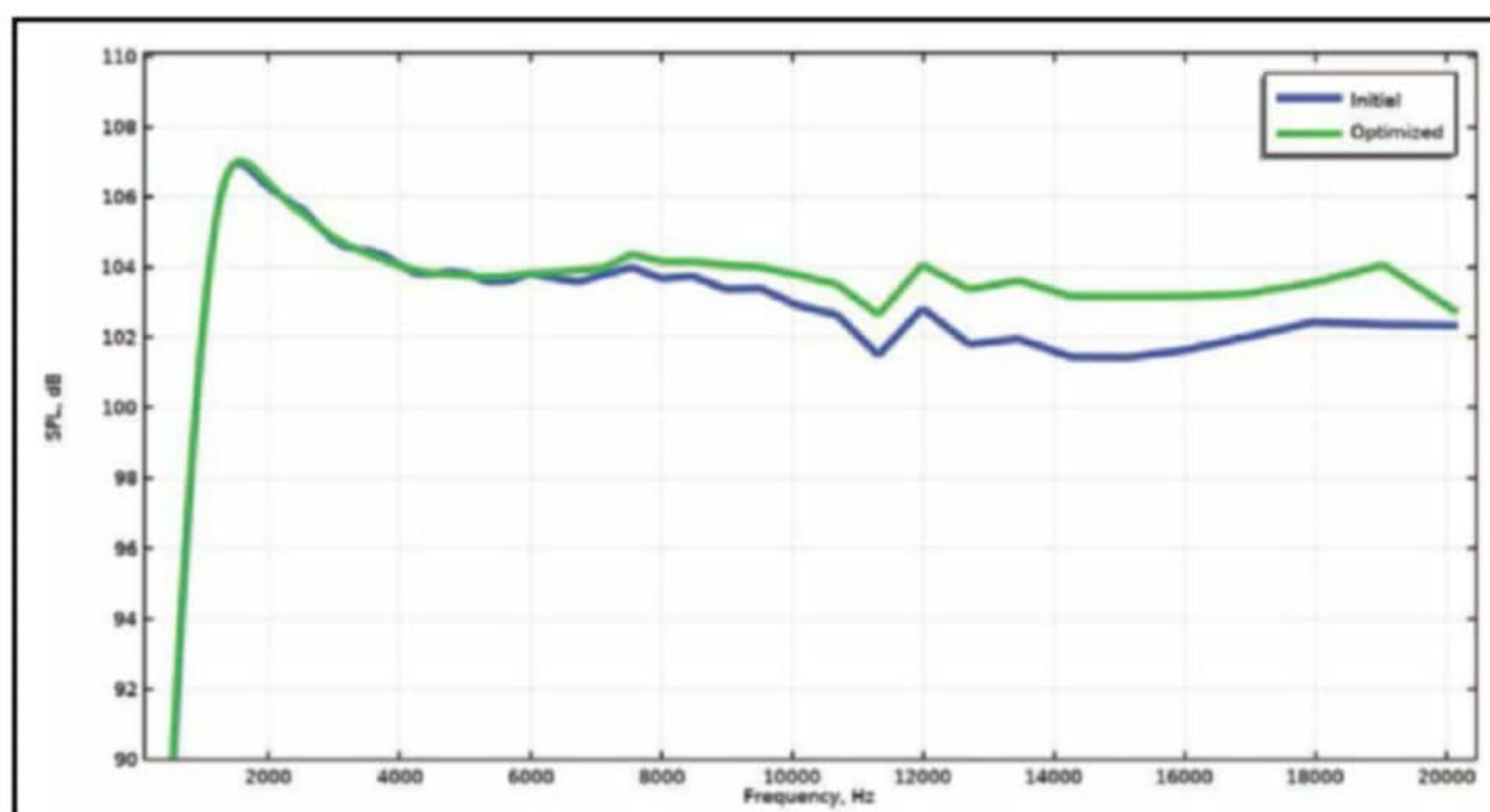


Figure 3: The on-axis sound pressure level for the tweeter before (blue) and after optimization (green)

or constraints to include the new information and run the optimization again. Also, the fact that the geometry is placed this close to the tweeter could point toward an issue with the displacement phase on the tweeter surface [4], and so it might be that the tweeter itself can be modified in such a way that the phase plug is no longer needed. In such cases the optimization will still have aided in the design work, and so it is not always the aim to use the topology optimized geometry directly, but instead as inspiration for simpler changes.

## Speakerphone

A generic speakerphone was made in which the front domain was assumed available as a topology optimization domain. The initial response on-axis is shown to have a dip at around 4kHz, and the objective here is to have a flatter response on-axis without this dip. With the speakerphone sitting on a table, on-axis pressure may be less important than off-axis responses, but it is chosen here anyway to illustrate the principles of the optimization. The speakerphone is shown in **Figure 4** with the geometry resulting from the acoustic topology optimization

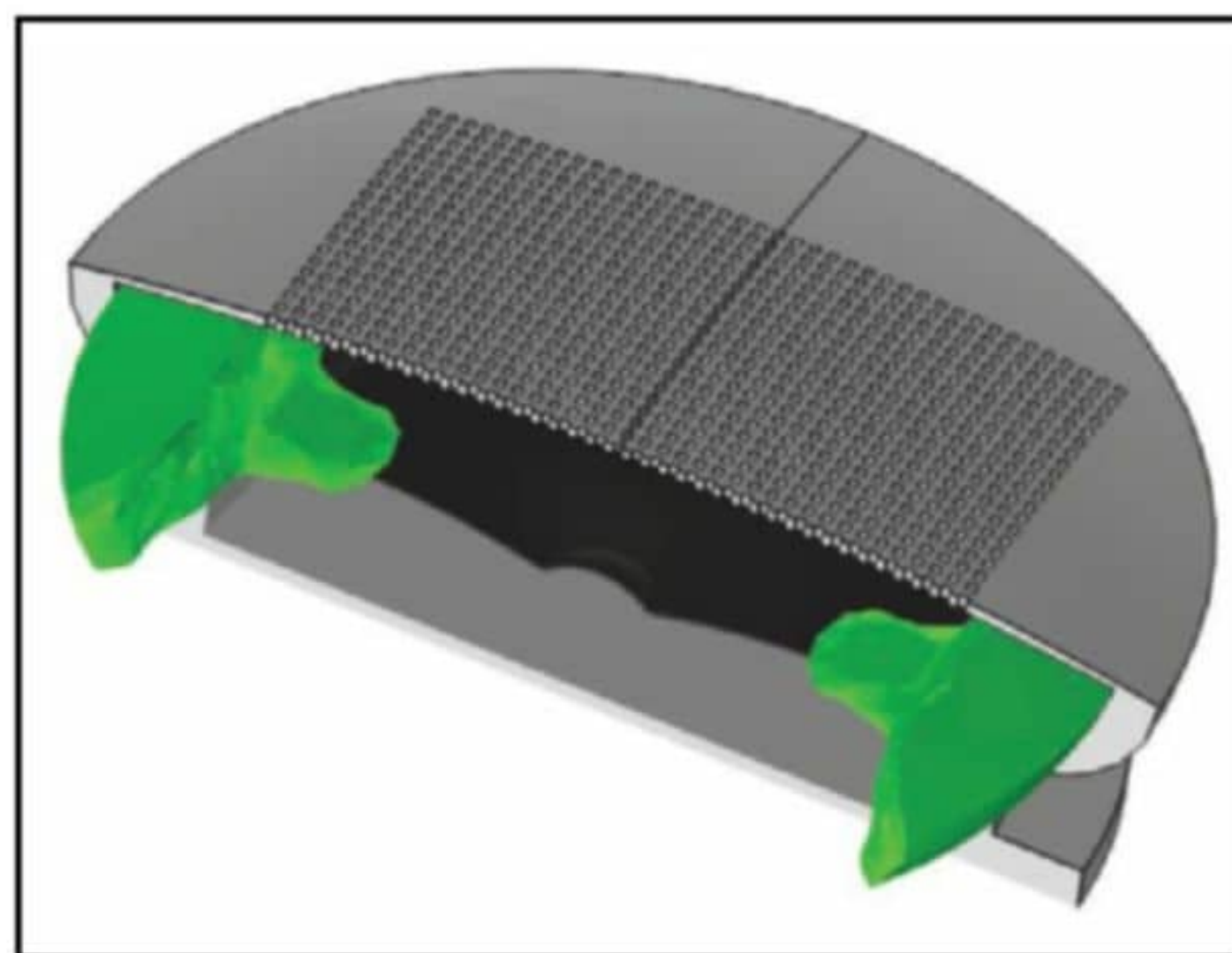


Figure 4: The speakerphone with the green domains being the “solid” geometries emerging from the topology optimization

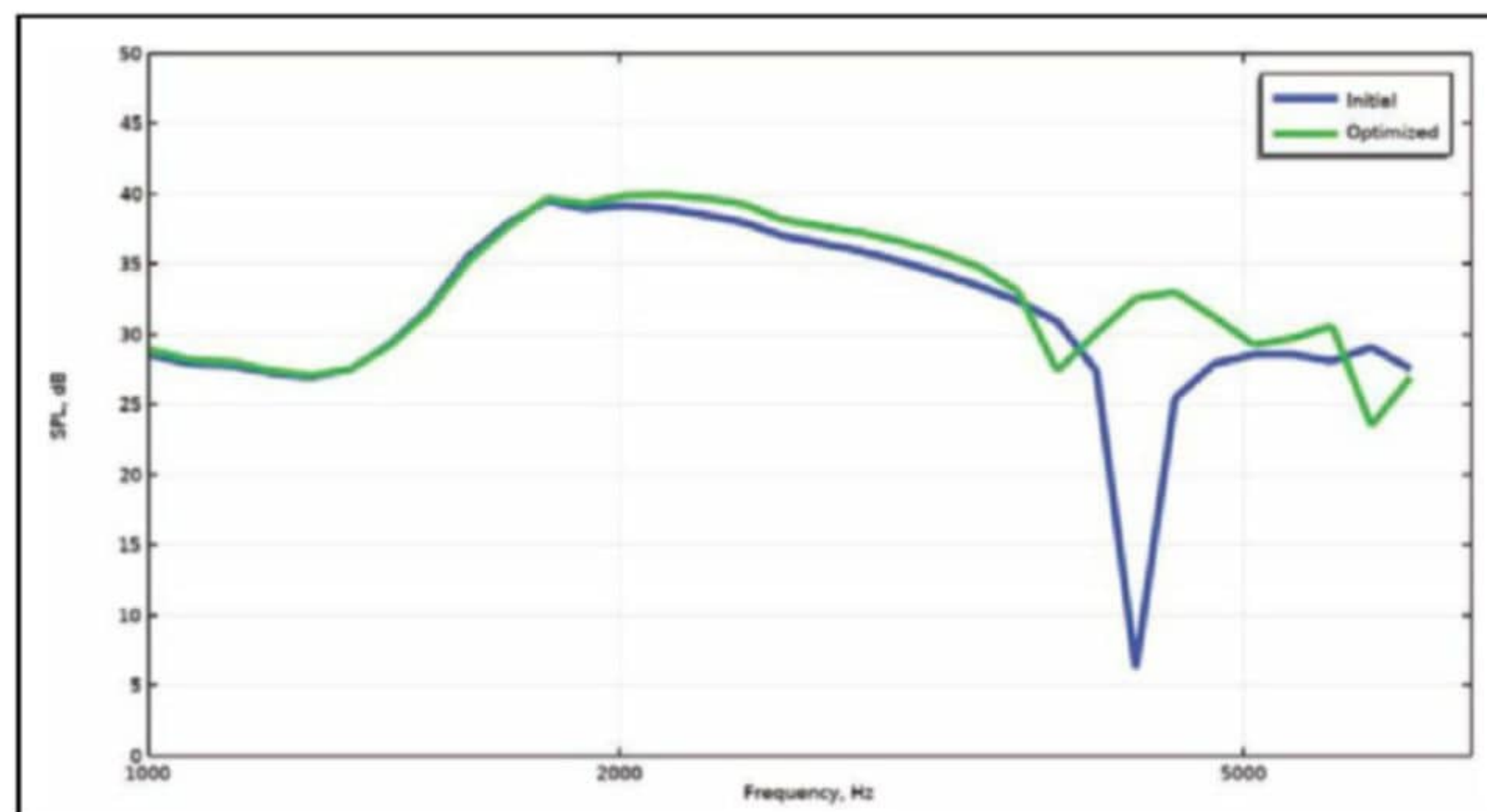


Figure 5: Speakerphone sound pressure level before (blue) and after optimization (green)

## About the Author

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



in green. The SPL is flatter after the optimization as shown in **Figure 5** with the dip removed from the response. Instead of exporting the geometry parts from the optimization and using them directly in a design, their shape and placement could again be used as inspiration for where to place components that anyway must be in the speakerphone, such as the electronics. And for this case it could also be that traditional engineering methods would have revealed the underlying modal acoustical behavior and a similar solution could perhaps fall out of this analysis. As already mentioned, for certain problems the combinatorics involved will make it impossible to find the proper distribution of material by a trial-and-error approach, and one such example is now presented.

## Phase Plug for Waveguide

A tweeter is located at the bottom of a waveguide, and it is assumed that the waveguide cannot have its shape changed. Instead, the air domain in the waveguide is assigned for topology optimization, and so a phase plug can grow out of this domain to obtain certain targets regarding SPLs and directivity. While for a traditional loudspeaker this type of design would probably not be done using topology optimization, one can imagine situations such as loudspeakers sitting deep in a TV or other products with a complex geometry, where there is no other option than to change the topology in an acoustical path. Also, different phase plugs could be found that meet different objectives and delivered alongside the product as a way for the user to change for example the directivity of the loudspeaker.

The initial setup is shown in **Figure 6** with the tweeter indicated in blue sitting in a rectangular waveguide. There are several objectives involved here regarding SPLs and the directivity index, with the air inside the waveguide assigned available for topology optimization. The resulting phase plug has a very distinct geometry as shown in **Figure 7**, which could not have been found using traditional engineering methods, and it is unique to this setup. The directivity is shown in **Figure 8**, where a clear improvement in the radiation at higher frequencies is seen. Changing the objectives even slightly will result in a different phase plug topology, and in that sense a geometry found via optimization may not lend itself well to “reverse engineering,” where the functionality can be deduced after the optimization. That may be discouraging in that it is always best to understand the underlying principle of any added geometry, but this must be weighed against not finding a solution at all. Also, there will certainly be instances where the optimized result will be easier

to deconstruct, and simplified versions that are more easily manufactured may then be made. But in the current case with a complex initial geometry and multiple objectives, one simply must accept the geometry output as it is.

For any topology optimized case, there are several practicalities that need to be considered. As already mentioned, the geometry found is not explicitly defined as distinct geometry. Instead, it is the result of truncating the design variable field, so that above a certain value this field is considered “solid,” and below said value it is considered “air.” If the field is fuzzy, however, there is no clear geometry defined, and this can point toward an issue in the setup or simply that too much is expected of the routine when it comes to meeting the set objectives.

For wave propagation problems such as the acoustics involved in the present cases, every small detail of the resulting topology is important, and so the geometry needs to be very distinctly represented or else the manufactured part will not meet the targets when installed. Also, you might end up with parts of the optimized geometry floating in space without attaching to any other parts. In such cases, it will be necessary to modify the geometry so that these parts are connected. For the present case, all the phase plug parts touch the waveguide and so the waveguide could be made to have this shape, but there is no guarantee for this connectivity in the typical topology optimization setup.

### Conclusion

Acoustic topology optimization represents a new tool in the tool box for vibroacoustics engineers. It does require a lot of knowledge about the topology optimization procedure as many aspects are involved such as filtering needed to obtain a near-binary design variable field. The general acoustics involved should be well understood alongside the traditional engineering methods to determine if a case is suited for formal optimization or not. While still not commonplace, I expect that in the coming years we will see more cases where acoustic topology optimization is used in product development for loudspeakers and vibroacoustics applications in general. 

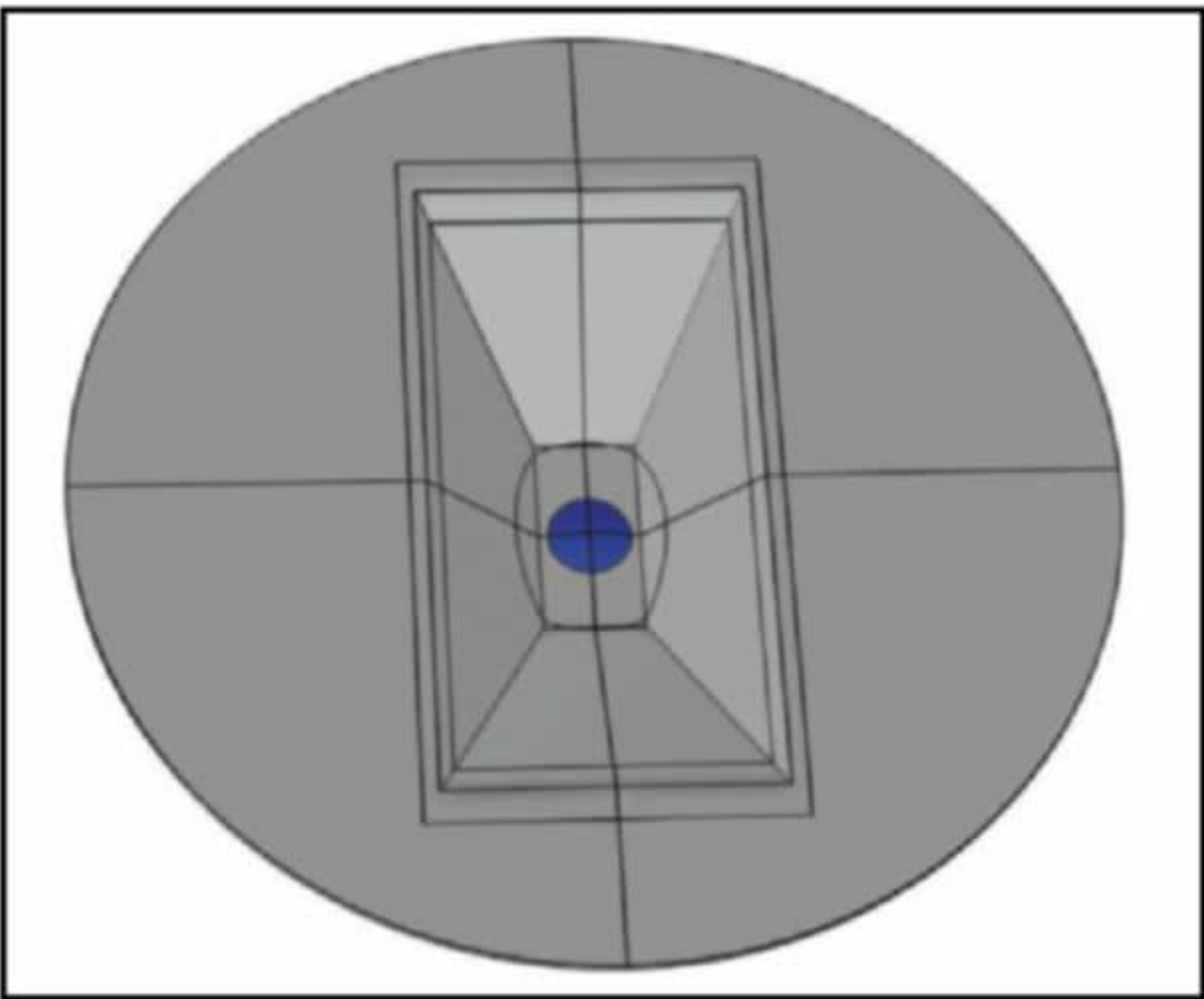


Figure 6: A tweeter (blue) situated in a rectangular waveguide.

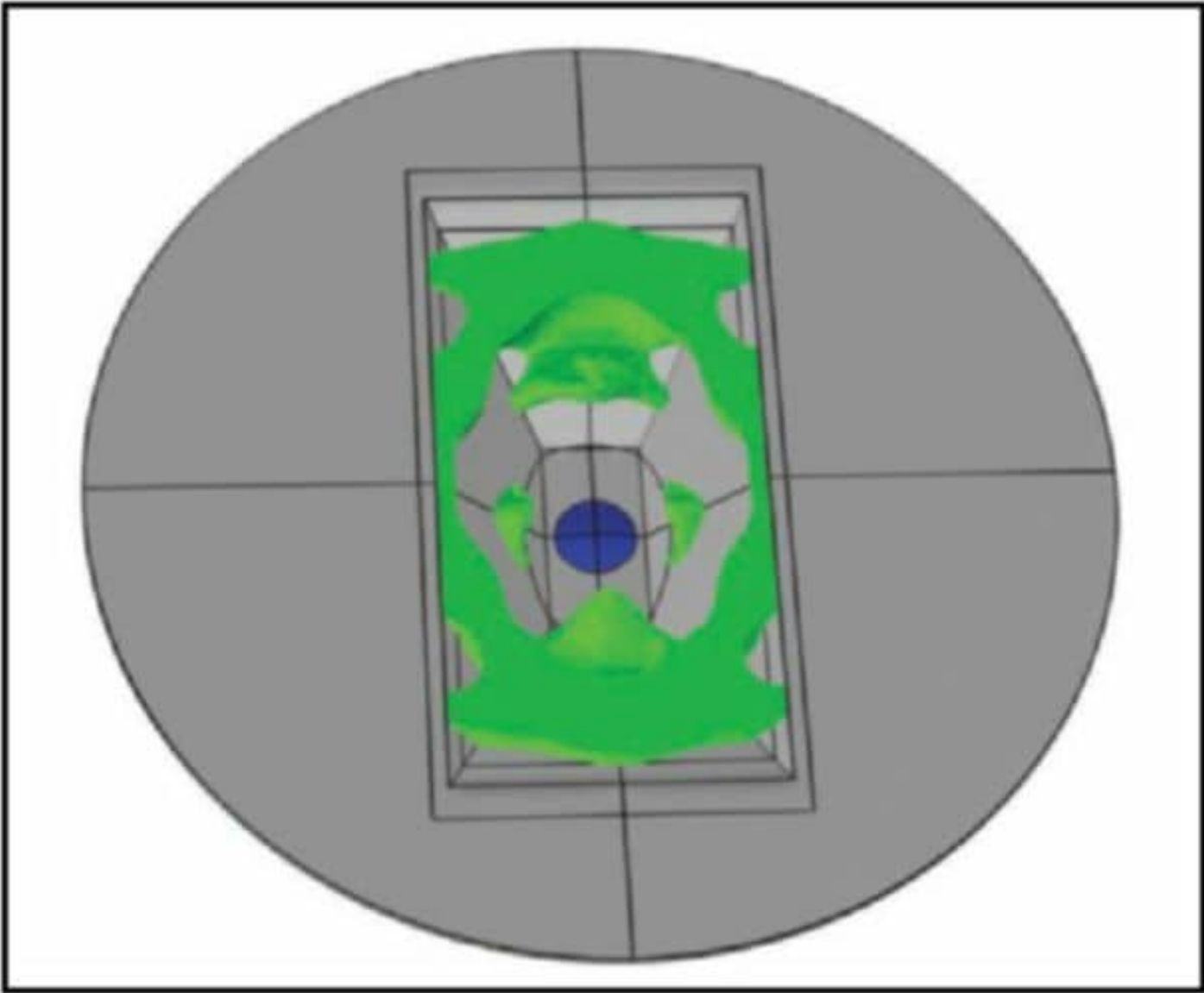


Figure 7: A topology optimized phase plug (green) for the waveguide.

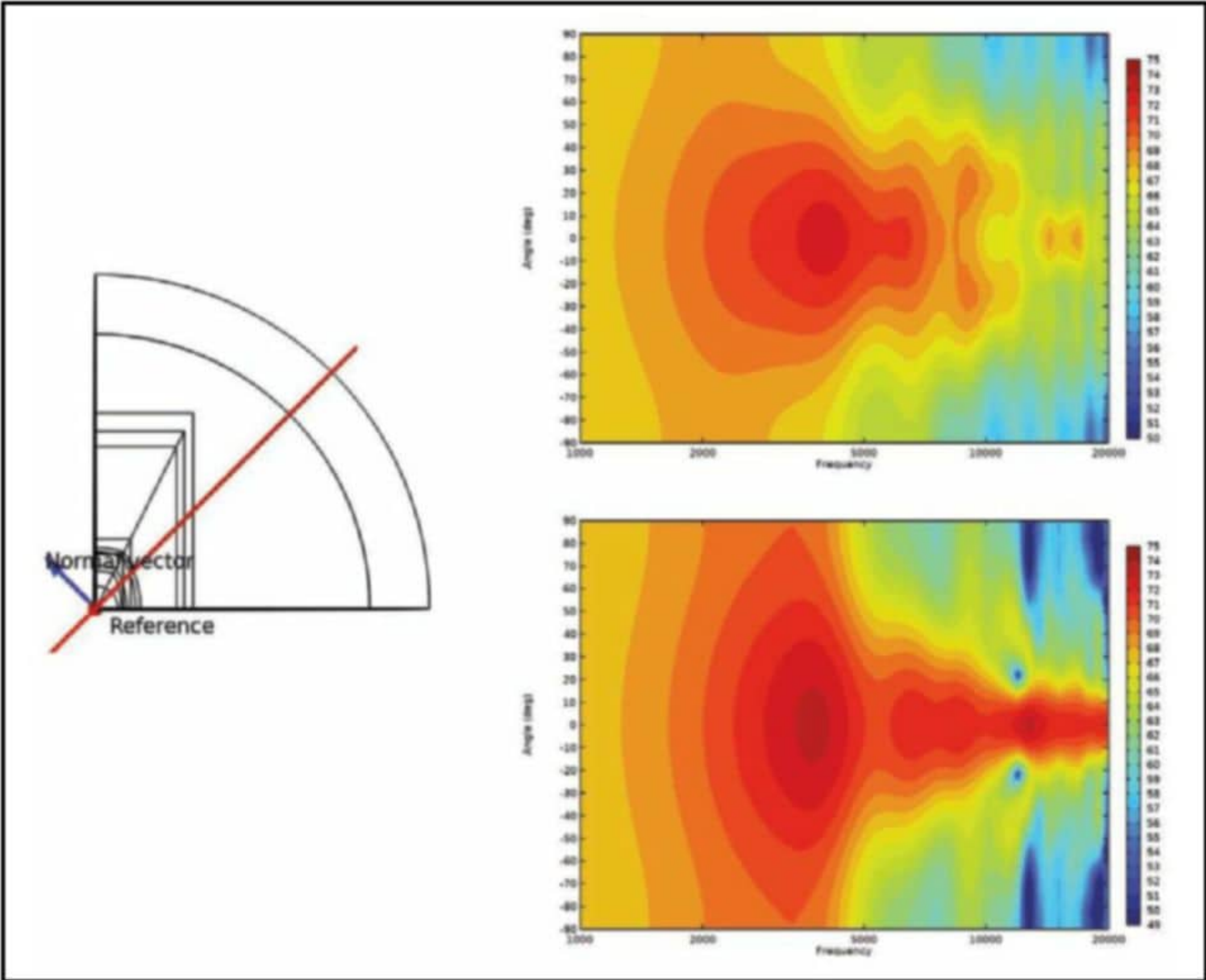


Figure 8: The directivity in plane at 45 degrees before (top right) and after (top bottom) optimization

### References

[1] J. Antony, *Design of Experiments for Engineers and Scientists*, Elsevier Science & Technology, 2014.  
 [2] M. Bendsoe and O. Sigmund, *Topology Optimization: Theory, Methods and Applications*, Springer, 2002.  
 [3] R. Christensen, “Acoustic Topology Optimization - Implementation and Examples,” in COMSOL Conference, Cambridge, MA, 2019.  
 [4] R. Christensen, “Loudspeaker Driver Displacement Decomposition,” *audioXpress*, January 2023.

# Wrong Things “Everybody Knows” About Acoustics

By  
**Richard Honeycutt**

Our acoustic expert sets the record straight about some of the more common misconceptions about acoustics that he has heard over the years.

My father had five careers. He taught motor vehicle maintenance in the US Army Antiaircraft Artillery Battalion during World War II. After the war, he taught high-school math. Subsequently, he farmed for a short time. Next, he worked as an engineer for Western Electric, working on guidance systems for missiles and rockets. Then his job title was changed to Senior Technical Writer. When I was in high school, I asked him a technical question to which he did not know the answer. Seeing my surprised look, he said, “A good engineer does not have to know all the answers; he just has to know where to look to find them.” I am convinced that this is true not just for engineering, but for all five of Dad’s careers, and in fact for most professions.

As an example, I recently received the following e-mail:

“We are a general contractor ... and I have a few questions for a client about the cost of acoustic treatment for a large sound stage type building. Call me at the number below please.” In the subsequent thread, it turned out that he was indeed just looking for cost information. I never found out who, if anyone, had decided what acoustical treatment was needed, or even if anyone had. He was not asking me to actually consult on the project.

There is a type of business called an “acoustical contractor.” In working on projects with several of these types of contractors over the years, I have noticed that many of them are mostly drywall installers who also install suspended ceilings. Some of these actually have acoustical value, but many do not, yet they are still called “acoustical.” Some of these businesses offer to determine how much and what kind of treatment a room “needs,” and where to locate it. Probably my correspondent perceived my business in that light.

In pondering this request, I knew right away that he was looking for an answer but did not know where to look for it. This realization led me to thinking of the many common misconceptions concerning acoustics. Let’s examine some of them.



## Good Acoustics

**“Acoustics” is just a person’s impression of how a space sounds.**

“Good acoustics” has the same definition for all spaces.

In fact, as discussed in the *audioXpress* November 2013 Sound Control column, the personal impression of how sounds varies from person to person in a given space, and depends upon the program: (e.g., a lecture, drama, movie, music), an in-person conference, a teleconference, none at all (think restaurant). Variations from

person to person are affected by tastes, hearing acuity, and other factors.

**A loud sound system overrides the acoustics of a space (makes its own acoustics).**

When we're talking about physical acoustics, this idea is patently false. But if we add the consideration of physiological and psycho acoustics, it's partially true. At high SPLs, the combined nonlinearities of air and of the human hearing system create significant and measurable intermodulation distortion (IMD) in the ear canal. The effects of this distortion include decreased ability to understand speech and a general sense of excess loudness.



**Everybody Knows What an Acoustical Consultant Will Do**

**Any acoustical expert will recommend the addition of acoustical absorption (fuzz) for any room.**

I, like most consultants, have worked to improve the acoustics of rooms that started out too "dry" (insufficient reverberation for a satisfying musical performance). The acoustical impression given by talking in an anechoic chamber is quite uncomfortable, and can be unsettling. Choosing the correct amount and type of absorption to impart optimal acoustics in a room is an art and a science abetted by experience. There are no "one-size-fits-all" statements about what a competent acoustician will recommend for a room.

**All acoustical problems can be treated by adding fuzz.**

This is closely related to the misconception we just addressed. Acoustical absorption reduces the Reverberation Time (RT) and can provide some reduction of internal noise (adjacent conversations, HVAC noise), but does not block noise from outside the room or from adjacent rooms.

**Acoustical consultants install "fuzz."**

A true consultant does not sell or install anything as part of his/her professional services. The consultant's work includes recommending ways to control internal and external noise, eliminate echoes, help to provide beneficial early reflections, and optimize RT. In cases in which acoustical absorption is indicated, the consultant recommends the type, amount, and location of the absorptive treatment.

**Acoustical consultants stay up-to-date on the prices of acoustical treatment materials.**

While the best consultants always try to provide recommendations that will give the client the best results for the least cost, and they maintain a general rank-ordering of the relative costs of different materials, knowing exact pricing at any given time is generally not within their scope. Instead, when a client needs to know pricing, the consultant refers him/her to a manufacturer's representative for authoritative information.



**General Inaccuracies About Room Acoustics**

**Well-designed acoustical spaces have no echoes, reverberation, or "room sound."**

Echo is sometimes used for deliberate effect. Examples include 1950s rock guitarists' "tape echo" and theatrical sound effects. (I once saw a performance of *The Phantom of the Opera* in which the echo of a gunshot was reproduced by a surround speaker just above my head—in the nosebleed section—and it was uncomfortably realistic!). But acoustical echoes are almost always to be avoided, so in that sense, a well-designed space will be echo-free.

Reverberation differs from echo in that it is made up of a series of reflected sounds that are not individually distinguishable, but form a sort of "sound tail." In the proper amount, and with the proper frequency characteristics, reverberation can enhance the sound of a venue. "Room sound" is an imprecise term often used to indicate the ways in which a program characteristically sounds in a given venue versus how it would sound in an open outdoors space. All concert halls have their characteristic room sounds, but teleconferencing rooms and audio production rooms should not. No one wants other teleconference participants to think (s) he is talking in a bathroom!

**The acoustical design targets of a room are determined only by its dimensions and category.**

In initial meetings with clients, acoustical consultants will ask the purpose of the room, and some questions about those who will use the room. "Auditorium" is not a sufficient answer for the purpose: a range of programs must also be specified—lectures, drama, cinema, choral performance unamplified instrumental performance by orchestras, or by small ensembles,

pipe organ, group singing, amplified music (Rock? Jazz? Folk music?) A “conference room” is not a sufficient answer either: In-person meetings? Meetings involving audio-visual elements? Teleconferences? “Adults” is not a sufficient answer for those who will use the room: Age range? Possible hearing challenges for a significant percentage? Seated listeners? Standing listeners? Multiple people eating, and how many? You get the idea.

## **The NRC tells you all you need to know about the performance of an acoustical absorption panel.**

In order to simplify rating the effectiveness of acoustically absorptive materials, Wallace Sabine introduced the concept of acoustical absorption coefficient. He measured these coefficients for a number of different materials and at a range of frequencies. Manufacturers of such materials pay third-party labs to test their materials’ absorption coefficients in one-third octave bands. These test reports are available to acoustical consultants, who use them in order to optimize the design of spaces for the desired purposes, programs, and occupants. Further simplification was added when the Noise Reduction Coefficient (NRC) was introduced. The NRC is the numerical average of the octave-band absorption coefficients of the 250Hz through 2kHz bands. While useful for evaluating an acoustical absorber’s effectiveness in some applications, the NRC provides insufficient information for specifying absorption materials for a concert hall, music rehearsal hall, practice room, or teleconference/telepresence room.

## **“Fuzz” blocks sound transmission from one room to another.**

Most people are familiar with the concept of thermal insulation involving fibrous materials, such as blankets, fiberglass batts, and such. Since many acoustical absorbing materials look similar, and may even exhibit the same fibrous nature, it seems logical that since fibrous heat insulation blocks heat transfer, fibrous sound materials will also block transmission of sound. Further confusion is added by the fact that sound-blocking materials are sometimes called “acoustical insulation”

In fact, acoustical absorption and acoustical insulation are very different things that depend upon very different physical mechanisms. Acoustical absorption works by absorbing acoustic energy, often at room boundaries. Acoustical insulation works by blocking sound transmission, principally by massive structures such as concrete walls. A fiberglass batt has little mass, and therefore has little sound-blocking effect. Further information on acoustical isolation can be found in the *audioXpress* January 2014 Sound Control column.

## **Anything soft (window curtains, art banners) helps tame reverberation and echoes.**

Some years ago, I was tasked with improving the acoustics of a gathering room in a nearby children’s home. When I visited the site to collect information, the interior designer was among those present. I immediately identified the boomy reverberant sound of the concrete-walled room and mentioned that we should

begin by correcting that issue. Immediately, the designer replied, “Sure, we need soft materials. Of course, she was correct. However, I soon found out that her idea of “soft materials” was thin drapes.

More recently, I was asked for advice on a sound system upgrade for a small church. This time, the problem was flutter echoes between side walls, aural hot spots caused by focused reflections from the concave ceiling, and somewhat excessive reverberation. One of the church members had a relative who was hired to design and install the improved sound system. You guessed it: The next Sunday, there were thin cloth banners on the walls. Unsurprisingly, the acoustical problems remained.

Often the amount of acoustical treatment is specified in terms of area. But the type and thickness of the material used is at least as important. One local church—built in 1954—had architect-specified 0.5”-thick painted fiberglass panels from 3’ upward on all walls. As any competent acoustician could have predicted, the room sounded boomy, with the high frequencies being perceived as blanketed.



## **Why Bother with a Consultant?**

**You can ignore your consultant’s advice when building a venue, then add treatment later if you decide it is necessary.**

Bringing in a contractor to retrofit an acoustical design often proves more expensive than doing it as part of a building or renovation project. Also, generating enthusiasm among decision-makers for additional work after a major project has been completed leads to the “never enough money to do it right, but always enough to do it later” syndrome, which should be understood as irony: Only in the imaginations of those advocating “doing it later” is there enough money later. Reality frequently dispels this wishful thinking, by which time it is too late.

The disappointment of a poor acoustical result can taint the enthusiasm of potential users/audiences, creating a bad reputation that subsequent improvements may have a hard time overcoming.

## **Acoustical contractors should be relied upon for acoustical design advice.**

A good acoustical contractor knows how to properly install acoustical treatment elements. (S)he may or may not understand how to use a computerized calculator or cell-phone app to find

the required total absorption needed in a given room. The contractor is unlikely to understand the conditions that a room must meet for the simple statistical acoustics calculations used by such calculators to be even approximately accurate. Unlike a competent consultant, who knows when and how to use a full computerized model in professional modeling software (e.g., EASE, Odeon, CATT Acoustic, etc.), the contractor is likely to depend upon inappropriate calculations.

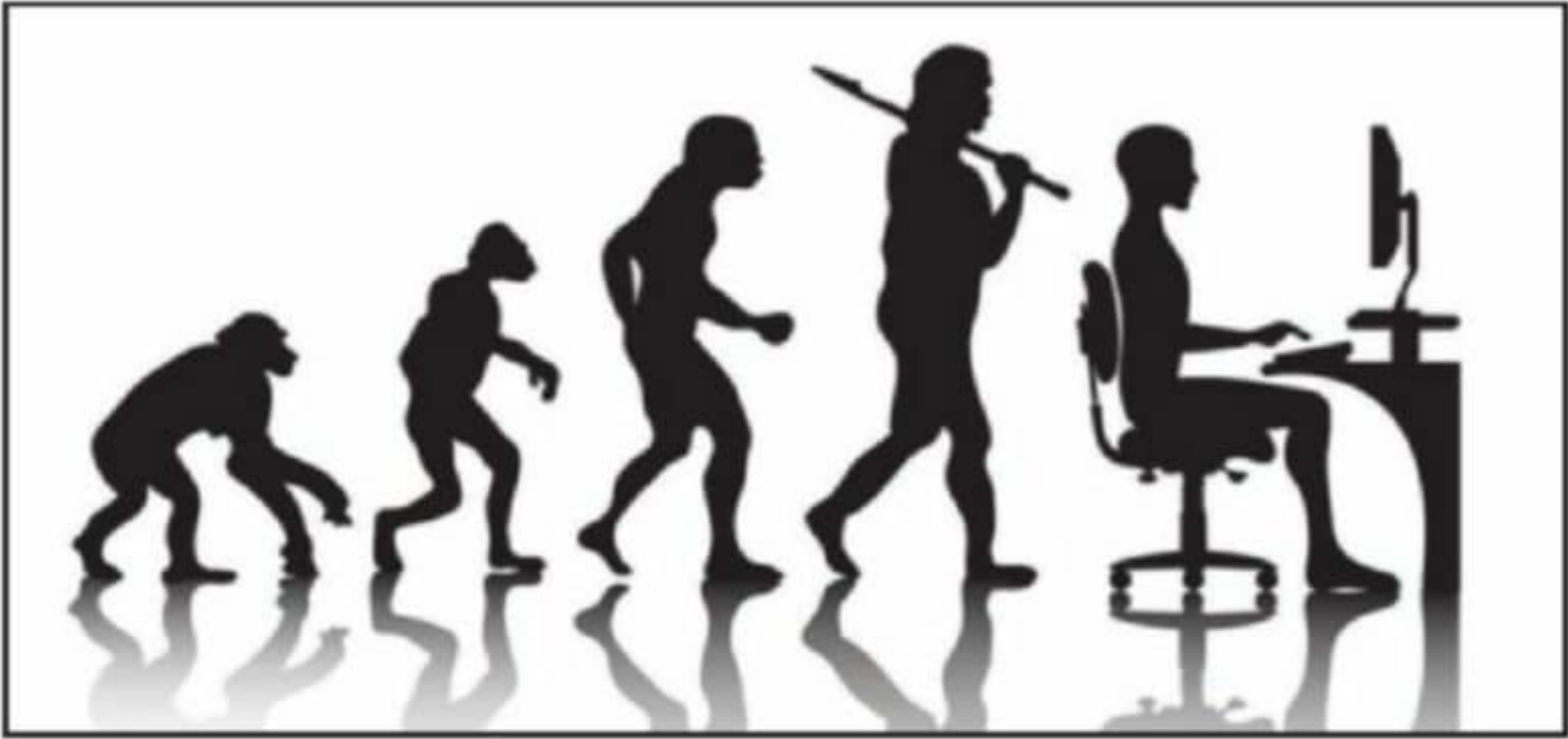
In the design of professional installed sound systems, one important step is making sure that the number of sources feeding a DSP does not exceed the number of inputs of the DSP: The “gozoutas” must not exceed the “gozintas.” Many acoustical contractors—while quite competent installers—have too few informational “gozintas” and too small a tool box to function well as a consultant.

**Anybody who has an iPhone with the right app can make accurate acoustical measurements.**

Back before iPhone apps, I received a phone call from a local landowner who was trying to get the county commissioners to pass a noise control ordinance. He asked what I thought the cost would be to supply deputy sheriffs with sound level meters (SLMs). He knew that at that time RadioShack offered one for about \$39.00. I told him that before getting into equipment cost, we should look at some issues that would affect the county’s ability to make a noise charge stick if brought to court. What training did the deputies have, when were the SLMs last calibrated, and where would the noise measurements be made (at the property line, inside the plaintiff’s house, on the plaintiff’s lawn)? Admittedly, the requirements for legal action exceed those for which most acoustical measurements are made, but the requirements for training, calibration, and proper protocol cannot be ignored if measurements are to be useful.

In the case of RT measurements, locations of the sound source and test mic are critical, as are the background noise level and spectrum. If the measurement is to inform choices of acoustical materials, the cost of relying upon inaccurate measurements can be high. Interpretation of the implications of any acoustical measurement is as crucial as the measurement accuracy, and understanding plus experience are necessary for useful interpretation.

As discussed in the Sound Control articles published in the October 2014, November 2015 and December 2015, and October 2022 and November 2022 issues of *audioXpress*, not all apps give trustworthy results. So having “the right app” is a non-trivial issue.



**The Bottom Line**

It’s important to remember that, while a little knowledge is a dangerous thing, we all progress step by step: no knowledge > a little knowledge > semi-competency > full competency > expertise. The point of this article has been to help us keep our minds open enough that more bits of knowledge can enter, but never to trust hearsay that “everybody knows.”

**About the Author**

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

GETTING THE MOST OUT OF  
**VACUUM TUBES**  
By Robert B. Tomer

**Vacuum Tubes—The Important 4/5ths of Tube Circuits**  
Over 80% of all tube equipment defects result, directly or indirectly, from tube failures. Why do tubes fail? What can be done to prevent them from failing before their time? How can you determine whether a tube is good or bad, or how well and how long it will work in a given circuit? Should tubes be replaced periodically, whether they’ve failed or not . . . or should they be tested every so often, and replaced if indications show them to be below par? This book supplies the answers to these profound questions . . . plus many, many more.

**Like Watching a Play From Backstage**  
Author “Bud” Tomer, an international authority on tubes, is well-known for his candid “Tech Tip” bulletins and his slide-film presentations to service, military and engineering groups. Within this book you’ll find a virtual encyclopedia on vacuum tubes—yet the author has not delved into mathematical details that would be of interest to only a few tube engineers. Everything is focused toward the interest of those who use tubes—either for replacement purposes or in original equipment designs.

In his refreshing, down-to-earth style, Mr. Tomer sheds light on such questions as: “Why so many tube types?” “What About Tube Testers?”, etc.

**Other Interesting and Informative Topics:**

- 10 reasons why tubes fail
- Predicting tube performance
- Selected and preferred tube types
- Lengthening tube life

Anyone who has anything at all to do with tube equipment design, or proper use—should read this book. It is equally useful for the hobbyist, the technician, and the production and manufacturing processes—or even for the repair of vintage computers and radar gear.

**You’ll also learn about:**

- Catastrophic failures—what they are, when and how they happen
- Degenerative failures—what happens to tube performance over time
- Subjective failures—why a seemingly normal tube may be defective
- Characteristic variables—differences that matter
- Special-purpose tubes—considerations for filament and thyratron types.

**Order your copy**  
**cc-webshop.com**

# Vintage Op-Amp Noise Simulation

## Part 1 — Deviation from Realistic Noise Figures in Datasheets

By  
**Burkhard Vogel**

This article is divided into two parts. The first part presents the deviations from realistic noise figures in datasheets, based on measurements and calculations for real op-amps. In the second part, the findings are expanded through the use of Mathcad calculations, culminating in an accurate Spice model for the noise sources, and conclusion about noise source correlations.

For simulation purposes of the noise production of operational amplifiers (op-amps or OPAs), the major thing to find is the right math equations that describe the traces of the equivalent input referred voltage and current noise of these devices. With these kinds of equations and with the findings described in my book *Slopes and Levels – Spice Models to Simulate Vintage Op-Amp Noise* (the book—by kind permission of Springer Nature, Switzerland), we can model the noise production of any kind of noise generating OPA.

In this article and with the data of Texas Instrument's (TI) example OPA NE5534A (or Raytheon's RM5534A [1]) we have enough input for

further handling of simulation models for any kind of old or new op-amps. All simulations are carried out by application of Linear Technology's LTSpice IV [2] or XVII (now Analog Devices), and the accompanying math software comes from Mathcad (MCD). The MCD project files can be found under *audioXpress'* website section "Supplementary Material."

Nearly 90% of this article's text comes from Chapter 14 of my *Slopes and Levels* book and Chapter 15 of the book fully presents the two MCD Worksheets. For a better understanding and because of the many links within the book's 26 chapters, the remaining 10% of this article is additionally taken from them.

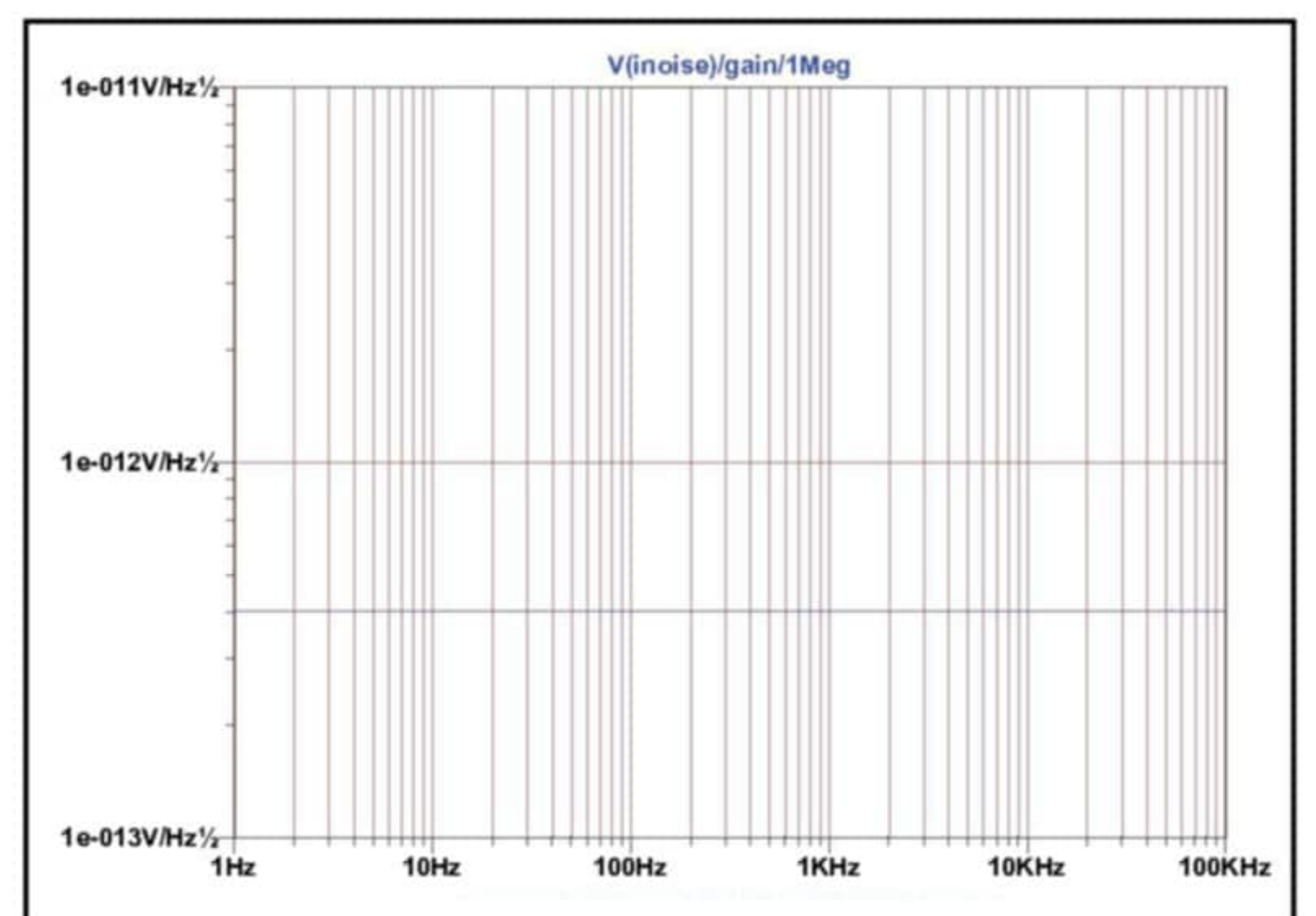
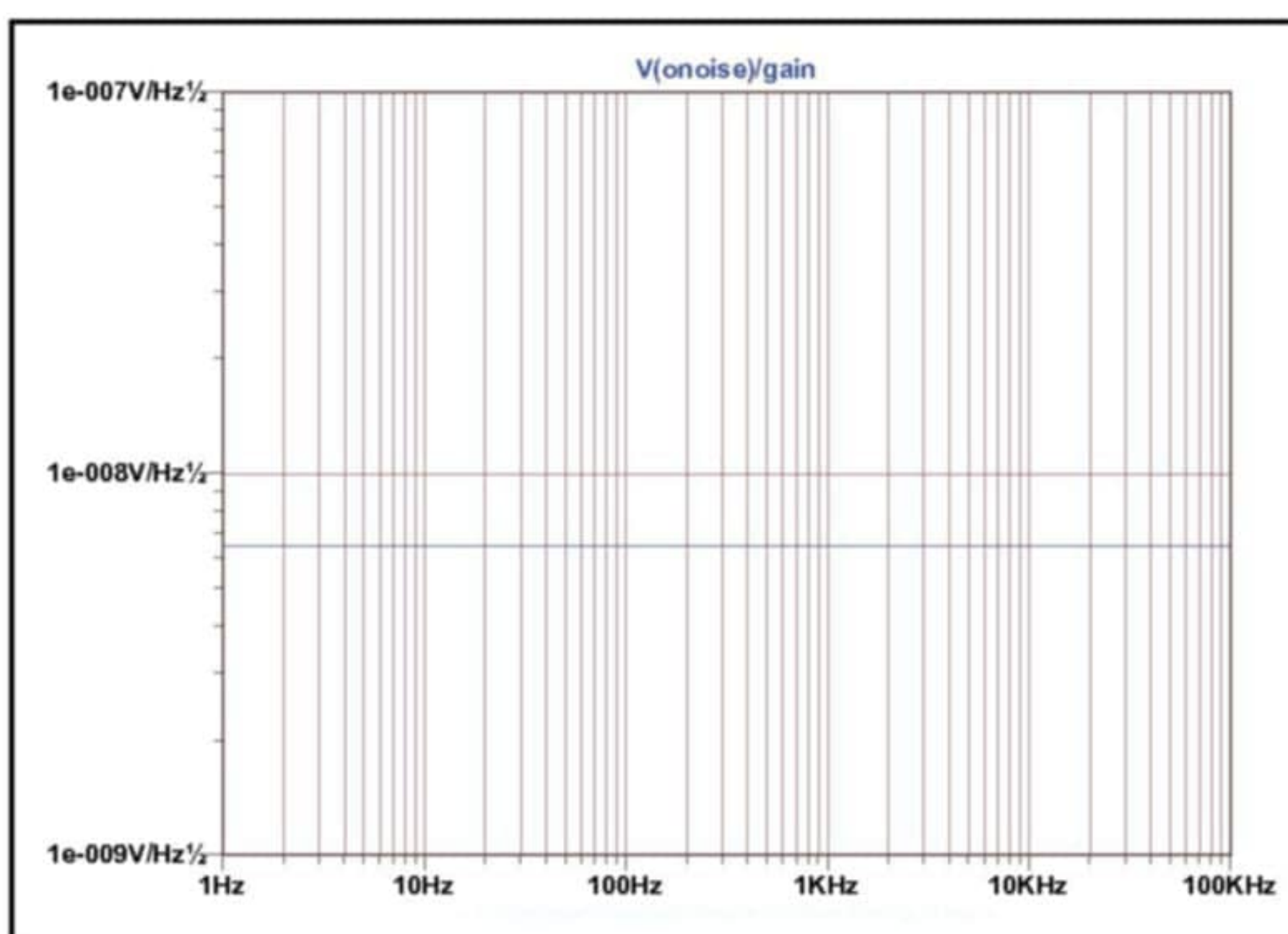


Figure 1: Voltage noise trace of TI's NE5534A simulation model

Figure 2: Current noise trace of TI's NE5534A simulation model

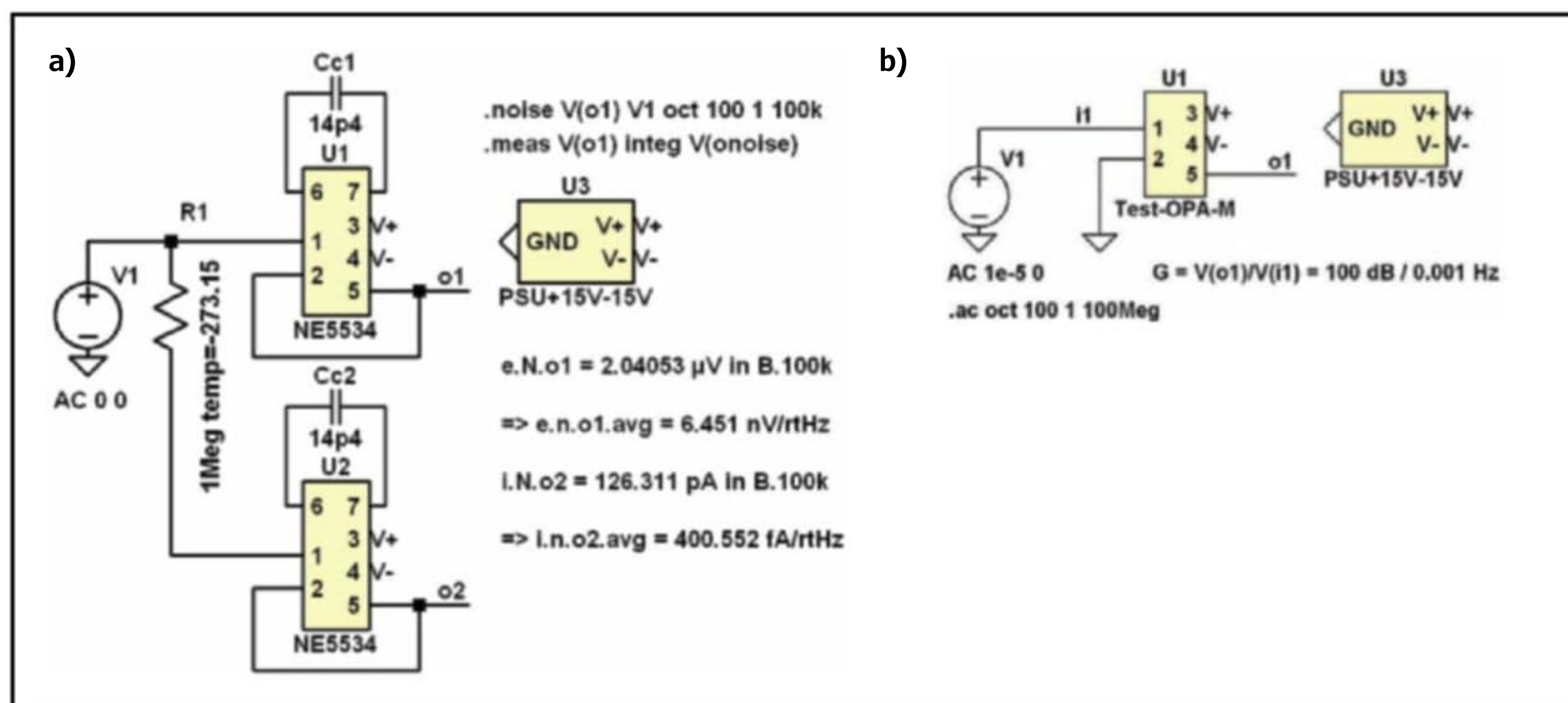


Figure 3a: Test arrangement for TI's 5534A model  
Figure 3b: Spice schematic to test the open-loop response of a DUT (here: Test-OPA-M)

As a side note, it was never my intention to write such an article for an audio magazine. It was Jan Didden, the inventor of the book's subtitle, who convinced me to try it, knowing that the amount of information presented might test a reader's patience. Nevertheless, this is the result.

However, before we jump into the matter, I want to thank Texas Instruments (TI) and Winfield Hill and Paul Horowitz, the authors of *The Art of Electronics*, for their permission to publish specific useful graphics and tables.

## The Simulation Model Curves Presented by the Manufacturer

The TI noise model of the NE5534A [3] performs as follows: **Figure 1** and **Figure 2** show the voltage and current spectral noise curves:

- Both flat in  $B_{100k}$
- White noise level voltage noise: 6.451 nV/rHz (rounded 6.5 nV/rHz)
- White noise level current noise: 400.552 fA/rHz (rounded 0.4 pA/rHz)

Both spectra were created with the test arrangement of **Figure 3a**. In this test arrangement we find  $Cc1 = Cc2 = 14p4$ , capacitance values good enough for the general findings approach presented here.

Both capacitances guarantee equal gain-bandwidth-product (GBW) or transit frequency ( $f_T$ ) of the manufacturer's simulation model NE5534 used here. Thus, application of the **Figure 3b** schematic creates **Figure 4** for TI's NE5534A.

A further test with one of my book's test OPAs (here we use the Test-OPA-M with gain parameters close to the NE5534 ones) leads to **Figure 5**. With only a tiny deviation from the datasheet graph below +10dB, it equals **Figure 4**, at least up to the open loop gain of +40dB at 150kHz. The open-loop gain curves of the Raytheon and of the very old 5534-predecessor from Valvo (TDA1034B) look the same.

The Test-OPAs I've modelled are based on the LTSpice IV UniversalOpamp2. They equal the LTSpice XVII UniversalOpamp3. Thus, Test-OPA-M's netlist for NE5534 gain purposes looks as follows:

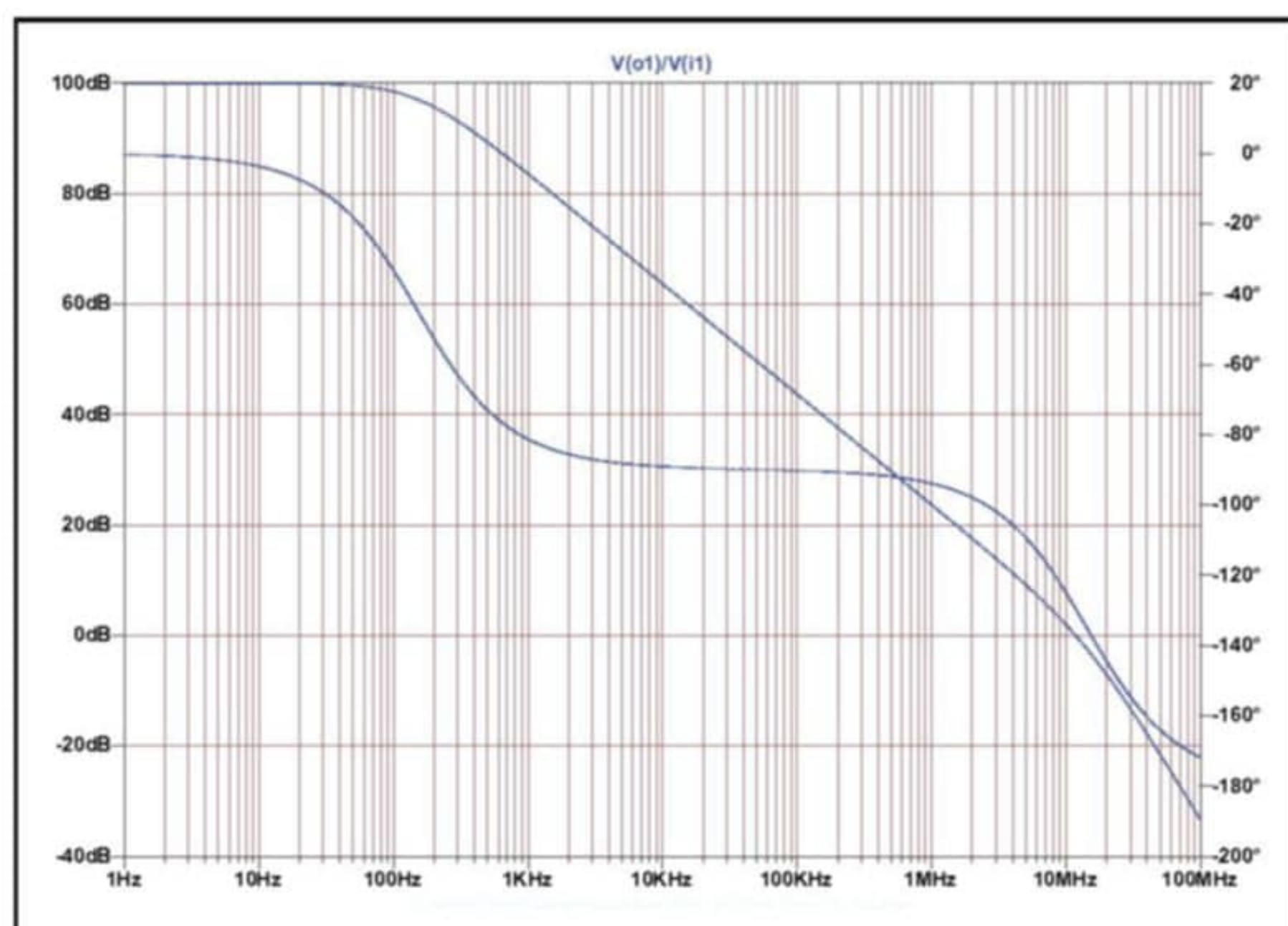


Figure 4: Open-loop gain of TI's NE5534A with  $Cc = 14p4$

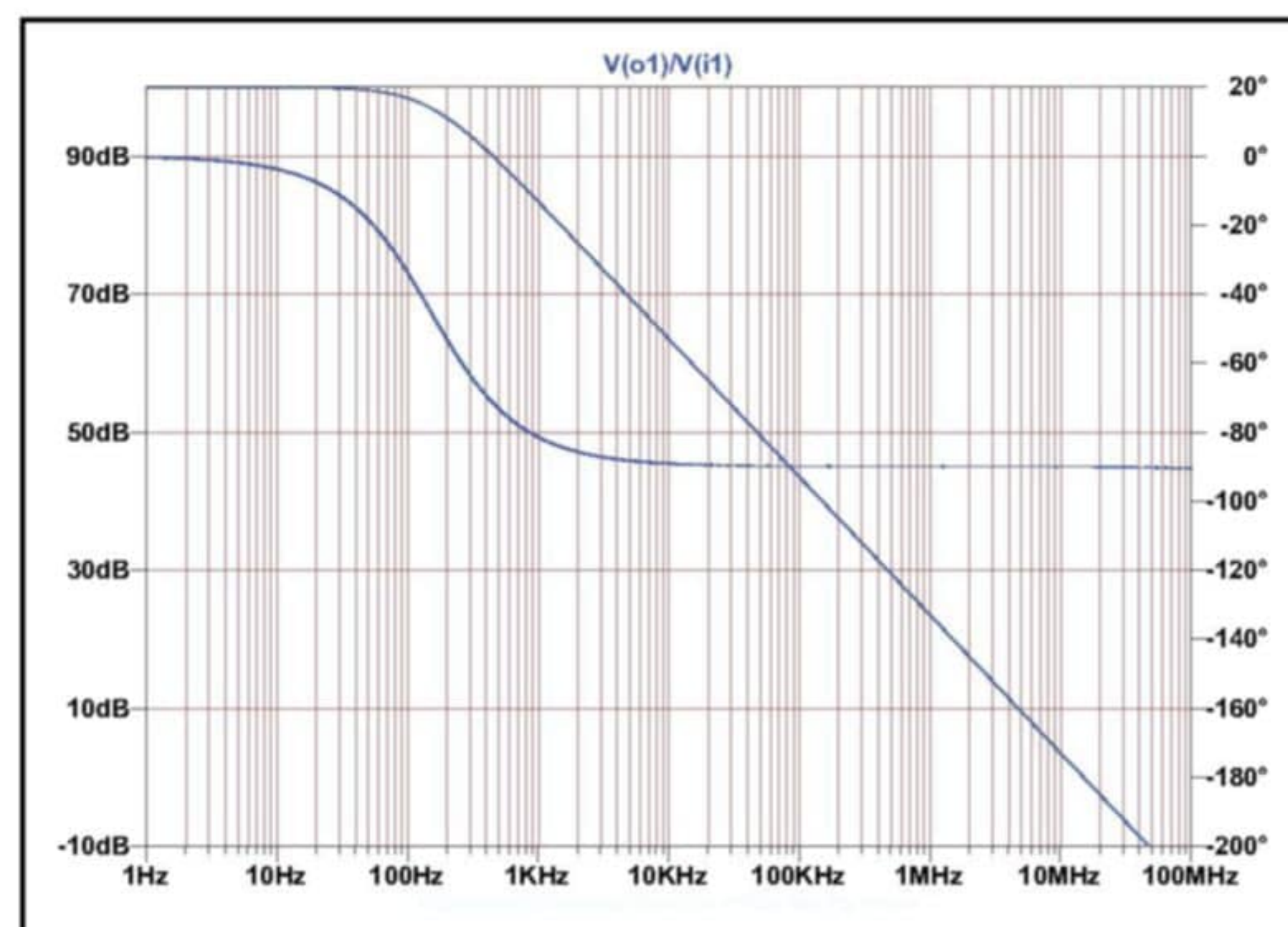


Figure 5: Open-loop gain with Test-OPA-M



```
.subckt Test-OPA-M 1 2 3 4 5
XU1 1 2 3 4 5 level.2 Avo1=0.1Meg
      GBW=15Meg Slew=13Meg
+ ilimit=25m rail=0 Vos=0 phimargin=45
+ en=0 enk=0 in=0 ink=0 Rin=500Meg
.lib UniversalOpamps2.sub
.ends Test-OPA-M
```

To get a Test OPA without any noise as described in the netlist shown here, we only have to set to zero en (input voltage noise), enk (e corner frequency), in (input current noise), and ink (i corner frequency). Additionally, there are two fundamental points to mention when working with the LTspice universal OPAs. Their input referred current noise sources produce 100% correlated noise. It won't be a big issue as long as one of the OPA's inputs is connected very close to ground. In this case the correlation effect doesn't hurt. In all other cases we have to find solutions that can overcome this correlation matter. This article will detail how.

The LTspice universal OPAs show—if so configured—1/f-noise slopes of -10dB/dec only. This creates problems to model OPAs with deviating slopes. Later in the article we will see how this

problem can also be solved by application of additional measures.

There is one thing for sure: neither the curves shown in Figure 1 or Figure 2 without any negative slopes at the lower end of the frequency spectrum, nor the white noise level of the voltage noise in nV/rHz hit the datasheet specs. With the exception of the right white noise level in pA/rHz, the same lack of negative slope applies to the current noise. Thus, we have to find the right data to get the equation that allows mirroring the real situation.

Data Collection

In Table 1 and Table 2, I've summed-up what we can find in the datasheets, and what we can get from the shown traces. Additionally, authors such as Douglas Self (notated in the tables as DS) with his second edition of *Small Signal Audio Design* and Hill and Horowitz (notated in the tables as H/H) with their third edition of *The Art of Electronics* considerably broaden the horizon of 5534 knowledge. Table 1 shows the voltage noise numbers, and Table 2 shows the ones of the current noise.

Before we jump into the tables, let's have a look at the mathematically generated and datasheet

Table 1: Datasheet and trace-based numbers of the different voltage noise sources

| Datasheet specs of equivalent input voltage noise [nV/rHz] |                  |          |            |                        |                |        | DS             | H/H |     |  |
|------------------------------------------------------------|------------------|----------|------------|------------------------|----------------|--------|----------------|-----|-----|--|
| Frequency                                                  | Raytheon RM5534A |          | TI NE5534A |                        | Valvo TDA1034B |        | 5534A          |     |     |  |
| [Hz]                                                       | Level            | Source   | Level      | Source                 | Level          | Source | Level [nV/rHz] |     |     |  |
| 1Hz                                                        | 65.0             | Figure 6 | nn         | DS table + DS Figure 7 | nn             | DS     |                |     |     |  |
| 10Hz                                                       | 8.5              |          | 7.0        |                        |                |        |                |     | 5.5 |  |
| 30Hz                                                       | 5.5              | DS       | 5.5        |                        |                |        |                |     |     |  |
| 100Hz                                                      | 4.2              | Figure 6 | 4.2        |                        | nn             |        |                |     |     |  |
| 1kHz                                                       | 3.5              | DS       | 3.5        |                        | 3.5            |        | 3.5            |     |     |  |
| 10kHz                                                      | 3.4              | Figure 6 | 3.5        |                        | nn             |        |                |     |     |  |
| 100kHz                                                     | 3.4              |          | 3.5        |                        |                |        |                |     |     |  |

Table 2: Datasheet and trace-based numbers of the different current noise sources

| Datasheet specs of equivalent input current noise [pA/rHz] |                  |          |            |                        |                |        | DS             | H/H |  |  |
|------------------------------------------------------------|------------------|----------|------------|------------------------|----------------|--------|----------------|-----|--|--|
| Frequency                                                  | Raytheon RM5534A |          | TI NE5534A |                        | Valvo TDA1034B |        | 5534A          |     |  |  |
| [Hz]                                                       | Level            | Source   | Level      | Source                 | Level          | Source | Level [pA/rHz] |     |  |  |
| 1Hz                                                        | 95.0             | Figure 7 | nn         | DS table + DS Figure 8 | nn             | DS     |                |     |  |  |
| 10Hz                                                       | 5.5              |          | 4.5        |                        | 1.5            |        |                |     |  |  |
| 30Hz                                                       | 1.5              | DS       | 1.5        |                        |                |        |                |     |  |  |
| 100Hz                                                      | 0.7              | Figure 7 | 0.7        |                        | nn             |        |                |     |  |  |
| 1kHz                                                       | 0.4              | DS       | 0.4        |                        | 0.4            |        | 0.4            | 0.4 |  |  |
| 10kHz                                                      | 0.35             | Figure 7 | 0.4        |                        | nn             |        |                |     |  |  |
| 100kHz                                                     | 0.35             |          | 0.4        |                        |                |        |                |     |  |  |

equaling traces [4], [5], [6]. In the tables I've also mentioned the traces of Valvo's 1976 TDA1034B. Despite the small difference of the DC current setting arrangement of the input circuit {FET-based (TI and Raytheon via Q8 in Part 2's Figure 17) versus BJT driven (Valvo via R4 + R5 + T5)}, this old OPA could be regarded as the predecessor device that shows absolutely the same noise characteristics. Obviously, this difference in treatment of the input DC current does not lead to a different input referred noise production.

Fortunately, there is no difference between the voltage noise traces of the three datasheets. The only exception is that the X-ordinates of Valvo's datasheet start at 10Hz. Thus, **Figure 6** and **Figure 7** exactly mirror the curves of the datasheets. The Mathcad (MCD) worksheet (available in the Supplementary Material) explains how to get them.

Additionally, I added Figure 8.60 from Hill and Horowitz's *The Art of Electronics*, the one with the datasheet traces of many low-noise OPAs [7], [8]. Our candidate is the one named "G," in the upper and the lower graph [9].

The trace in the graph of Figure 6 ( $e_n = 3.1 \text{ nV/rHz}$  at 1kHz, first 1/f-noise corner frequency  $f_{c,e1} = 50\text{Hz}$  does not exactly match the NE5534 data from Table 8.3a in Hill and Horowitz's *The Art of Electronics* book ( $e_n = 3.5 \text{ nV/rHz}$  at 1kHz,  $f_{c,e1} = 50\text{Hz}$ ).

However, and previously mentioned without data, the trace in the graph of Figure 7 ( $i_n = 0.4 \text{ pA/rHz}$  at 1kHz, first 1/f noise

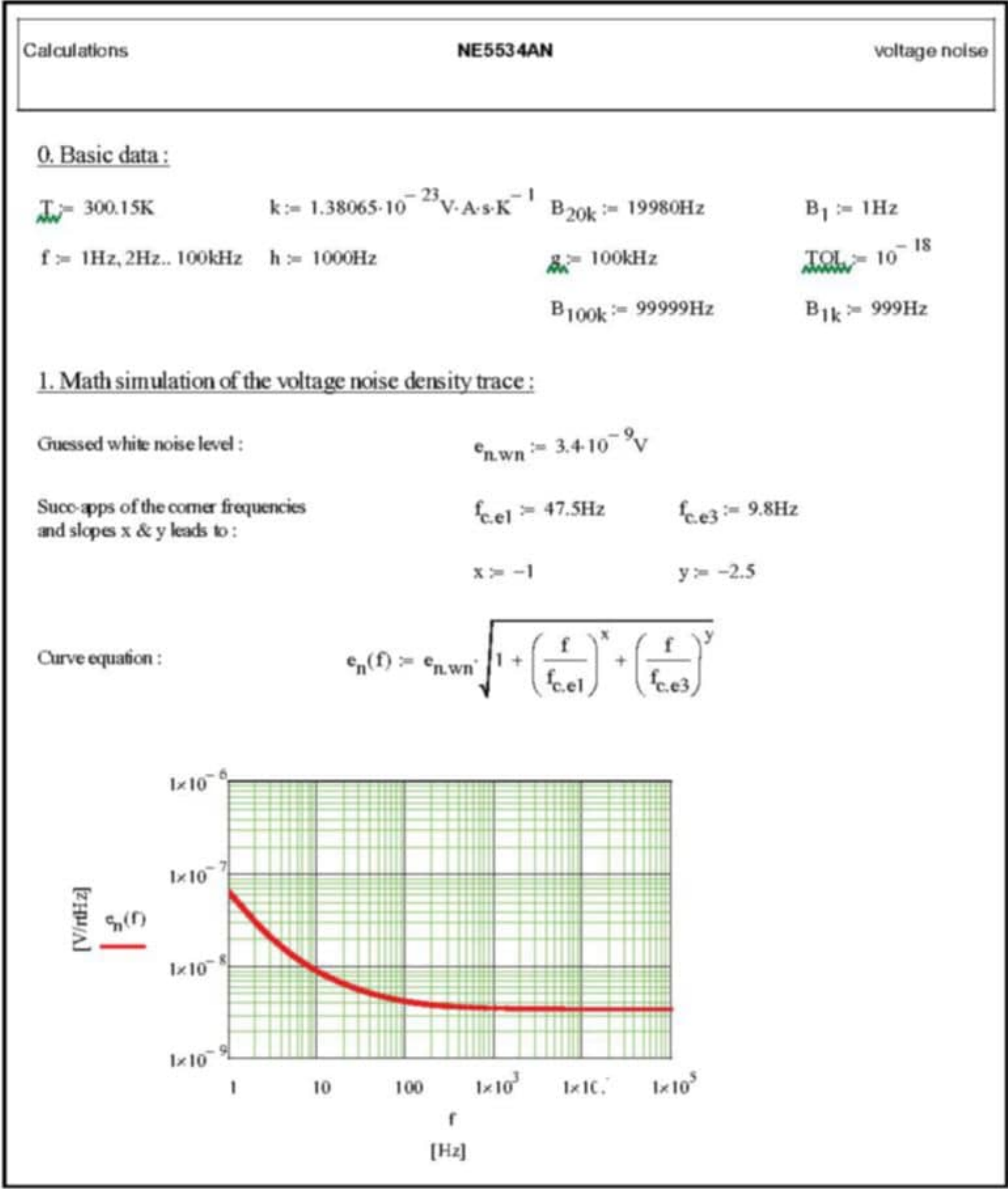


Figure 6: Mathematically constructed trace of 5534A's (and TDA1034B's) equivalent input voltage noise



corner frequency  $f_{c,i1} = 298\text{Hz}$ , the second corner frequency  $f_{c,i3} = 74\text{Hz}$  (see MCD Worksheet 2, Section 1)) does match the datasheet nearly perfect.

To round-up the knowledge about noise traces of OPAs, in Part 2's section about Further Material of Noise Traces, I present two other graphs: mentioned in reference [8] Hill and Horowitz's Figure 8.110 and Figure 8.111, which show an additional selection of measured voltage and current noise spectra (detailed in Part 2's Figure 15 and Figure 16).

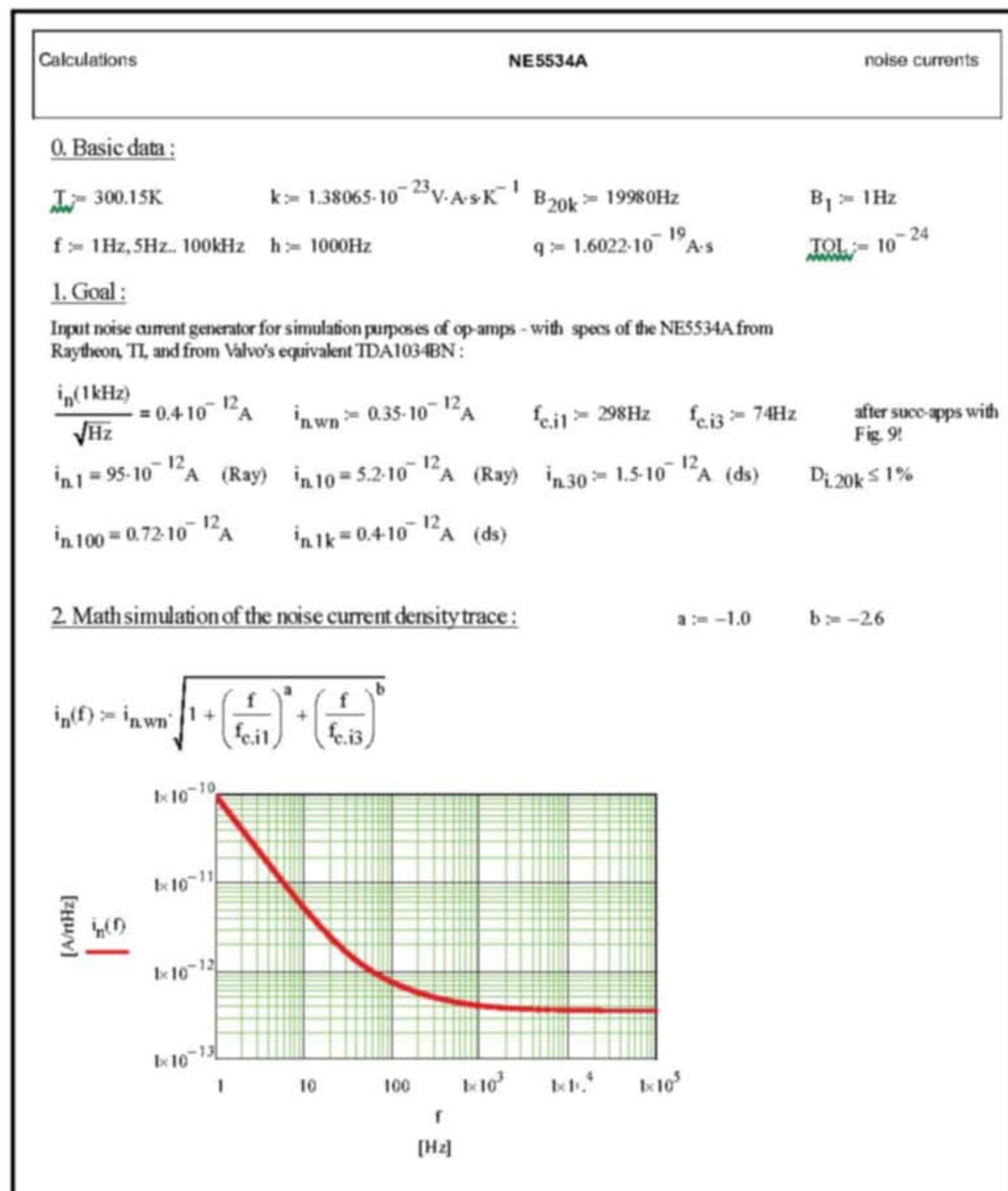


Figure 7: Mathematically constructed trace of 5534A's (and TDA1034B's) equivalent input current noise

The caption to Hill and Horowitz's Figure 8.60 [7] reads: "Voltage and current noise versus frequency for a selection of "low-noise" op-amps featured in Tables 8.3a-c on pages 522-524. All are adapted from datasheet plots, with the exception of measured in for op-amps G and J. The labels are in order of ascending en at 10Hz. See Figures 8.110 and 8.111 for measured noise spectra of selected op-amps." [8]

## Decision About the "Right" Traces

With the exception of an enlarged frequency range down to 1Hz the numbers in both tables and the noise traces in Hill and Horowitz's Figure 8.60 suggest that Raytheon's expanded noise traces could very well serve our needs to find the right equations for the new OPA simulation model NE5534AN. This can be done with the help of Mathcad (or any other math software) by successive approximation (succ-apps) of the trace's slope with the level numbers at specific frequency points à la Table 1 and Table 2, based on equations (1) and (2). Because of their increasing slopes at the lower end of the spectrum in our case here we have to expand these equations by additional 1/f noise corner frequencies  $f_{c,e3}$  and  $f_{c,i3}$  and steeper slopes  $y$  and  $b$ . Hence, we obtain the equations for the frequency dependent traces of the input referred voltage noise density  $e_{n,i}(f)$

## About the Author

**Burkhard Vogel** obtained his academic degree Dipl.-Ing. (Telecommunications) in 1973 from Darmstadt University of Technology, Germany. He then decided to follow a general management career path in the high-tech and IT industry in Germany, Switzerland, and Austria. In the recent decades fruitful discussions on tube audio and electronic noise topics with customers, employees, colleagues and friends led to a broad range of works on these issues. They were published in several magazines including *Electronics and Wireless World* (EWW), *Linear Audio* (LA), and *Design & Design* (D&D), together with many additional comments in EWW, LTEs in EWW, LA, and D&D. Finally, edited by the Springer Nature publishing house, Burkhard has seven books (currently number 8 is "under construction") that round up the gained knowledge.



## Project Files

Two Mathcad Worksheets can be found in the Supplementary Material section of the audioXpress website: MCD-WS 1 (voltage noise) and MCD-WS 2 (current noise) To download the Mathcad worksheets, visit <http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

## References

- [1] Raytheon, 1994 Data Book and 1989 datasheet, [www.rtx.com](http://www.rtx.com)
- [2] G. Brocard, *The LTSpice IV Simulator: Manual, Methods, and Applications*, Swiridoff Verlag July 2013.
- [3] NE5534 Operational Amplifier "Macromodel" Subcircuit Created Using Parts Release 4.01, Texas Instruments, April 1989.
- [4] Linear Integrated Circuits Handbook, Raytheon, 1989, pp. 4-232.
- [5] SLOS070D Datasheet, Texas Instruments, July 1979, Revised November 2014, Figure 7 and Figure 8.
- [6] Integrierte Schaltungen TDA 1034...TDA 1034 BN Datasheet, Valva, September 1976
- [7] P. Horowitz and W. Hill, *The Art of Electronics*, 3rd Edition, Cambridge University Press, 2016.
- [8] P. Horowitz and W. Hill, *The Art of Electronics*, 3rd Edition, Cambridge University Press, 2016.
- [9] Graph courtesy of Texas Instruments (TI), [www.ti.com](http://www.ti.com).

and of the input referred current noise density  $i_{n,i}(f)$  as follows:

$$e_{n,i}(f) = e_{n,wn} \sqrt{1 + \left(\frac{1}{f_{c,e1}}\right)^x + \left(\frac{1}{f_{c,e3}}\right)^y} \tag{1}$$

$$i_{n,i}(f) = i_{n,wn} \sqrt{1 + \left(\frac{1}{f_{c,i1}}\right)^a + \left(\frac{1}{f_{c,i3}}\right)^b} \tag{2}$$

In Equation (1) and Equation (2), the slopes of the noise traces are set by exponents x and y in the voltage noise case and a and b in current noise case. Hence: eg x = -1.0 would express a negative slope of -10dB/dec and y = -2.5 mirrors a negative slope of -25dB/dec. The same applies to the a and b exponents. A positive slope must then be expressed by a + sign. The subscript "wn" signals the white noise level of the voltage or current noise.

Next Month

In Part 2, the findings are expanded through the use of Mathcad calculations, culminating in an accurate Spice model for the noise sources and conclusions about noise source correlations. 

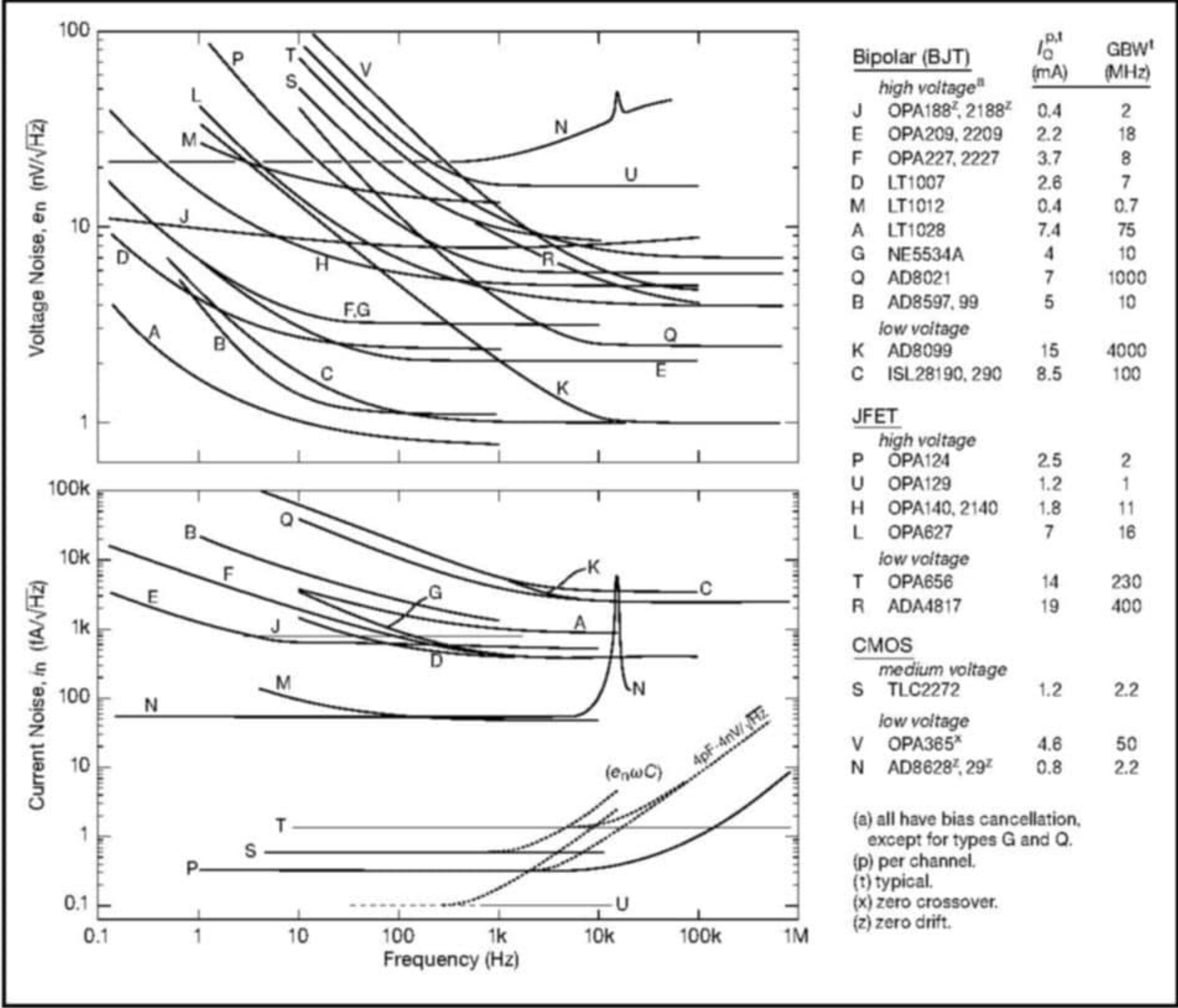


Figure 8: This is Figure 8.60 (a, b, c, d) from Winfield Hill and Paul Holowitz's book, *The Art of Electronics*.



The New QA404 Audio Analyzer...\$649

- 110 dB THD+N
- +18 dBV Output (4 ranges)
- +32 dBV Input (8 ranges)
- USB Powered
- Fast, automated testing
- REST programmable



www.QuantAsylum.com

Designed and manufactured in the United States  
Image shown is pre-production. Final image may change.



# Monoblock Power Amplifier With Precision Bias Control

## Part 1— The Design

By  
Bruce E. Gillingham

The project described here is a high-power (300W into 4Ω), bipolar, Class-AB, monoblock audio power amplifier. The design utilizes voltage feedback, a wide open-loop bandwidth (> 20kHz) approach, and an instrumentation amplifier architecture. It also features a novel precision Bias Controller that the author developed.

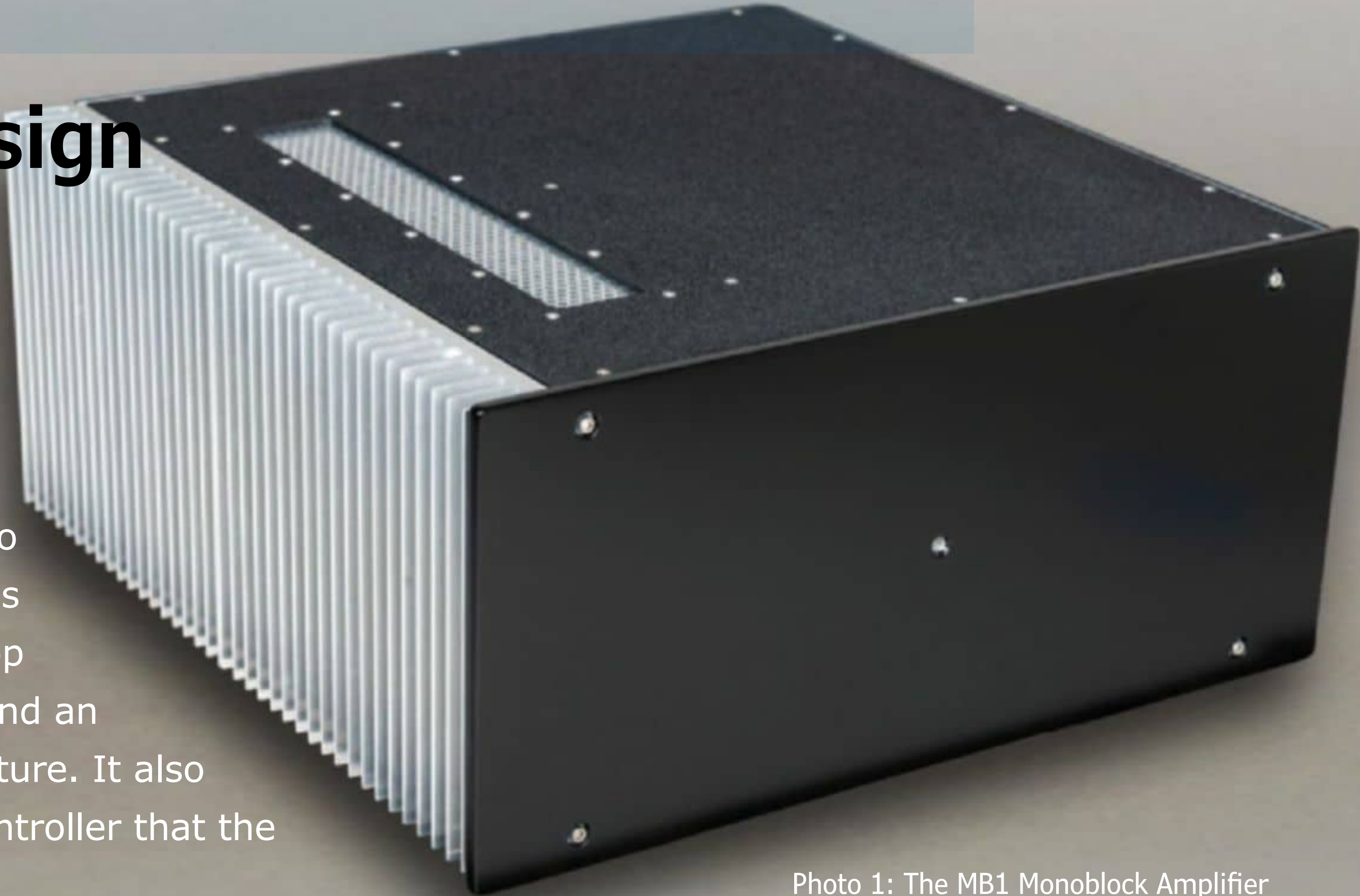


Photo 1: The MB1 Monoblock Amplifier

I refer to this amplifier as the BEG MB1, or MB1 for short, as it is my first monoblock design. The MB1 was developed gradually over an eight-year period. At first, the motivation for this project was to satisfy a longtime desire to use monoblocks in my audio system. My two previous designs, each

used for many years, were stereo. Later, the MB1 became a platform on which to develop the Bias Controller. **Photo 1** shows the MB1.

The “need” for wide open-loop bandwidth, most famously championed by Dr. Matti Ojala in the 1970s, has been debunked by Bob Cordell and others [1, 2, 3]. Still, I find some of Ojala’s tenets to be esthetically pleasing. Limiting open-loop gain to extend bandwidth sacrifices ultimate distortion performance, especially at low frequencies, but very low distortion can be still achieved by careful gain stage design and the application of a moderate level of global feedback. There are also real benefits to this approach. Well-defined gains and impedances ease design and test and improve repeatability. Limited open-loop gain improves behavior under overload conditions without adding circuit complexity. The MB1 handles full-scale square waves with aplomb, clips cleanly, and can even be run open loop without adjustment or DC servo correction. The measured open-loop output DC offset without correction is less than ±1V.

MB1’s instrumentation amplifier architecture provides high-impedance, fully balanced inputs, stable closed-loop gain, and high common mode rejection (CMR). Noise is extremely low, even when installed in a system with multiple devices connected to power safety ground.

| Parameter             | Performance               | Conditions                         |
|-----------------------|---------------------------|------------------------------------|
| Rated Output Power    | 150W RMS into 8Ω          | 20Hz to 20kHz: <0.003% THD         |
|                       | 300W RMS into 4Ω          | 20Hz to 20kHz: <0.006% THD         |
|                       | 400W RMS into 2Ω          | 20Hz to 20kHz: <0.006% THD         |
| Peak Output Current   | 30A into 1Ω               | 25μs                               |
|                       | 18A into 1Ω               | Continuous                         |
| CCIF IMD              | All spurs < -101dB        | Ref 150W peak into 8Ω              |
|                       | All spurs < -97dB         | Ref 300W peak into 4Ω              |
|                       | All spurs < -97dB         | Ref 400W peak into 2Ω              |
| Gain                  | 26dB                      | 4Ω load                            |
| Bandwidth             | 20Hz to 20kHz, ±0.1dB     | 4Ω load                            |
|                       | 0.1Hz to 134kHz, +0, -3dB | 4Ω load                            |
| Signal-to-Noise Ratio | >115dB                    | 20Hz to 20kHz BW, Ref 300W into 4Ω |
| Damping Factor        | >170                      | 1kHz into 8Ω                       |
| Input Impedance       | 200kΩ                     | 20Hz, differential                 |
| Common Mode Rejection | >80dB                     | Below 1kHz                         |

Table 1: This table shows the MB1’s key performance parameters.

Other high-level design choices include fully regulated power supplies for the amplifier’s gain stages, and extensive fault detection and protection circuitry.

Key performance parameters of the MB1 are shown in **Table 1**. **Table 2** provides a brief description of the five operating modes of the MB1 and the status indications provided to the operator via a single multi-color LED on the front panel.

Implementation Overview

The MB1 is relatively complex and its development involved many design trades. One design goal was to have very low power supply spurious (hum and buzz). The best way to accomplish this is through distance—separation of power supplies and amplifier. Fabricating two additional chassis and dealing with high current external cabling was daunting. I decided to stay with a single chassis per channel, dimension it to allow separation of sensitive circuitry from the transformers and internally partitioning it for shielding. Aluminum is not very effective with magnetic fields, but totally enclosing the transformers with aluminum is better than nothing and it is very effective with electrostatic fields (**Photo 2**).

The audio path circuitry is housed within the smaller of the two chassis partitions (**Photo 3**). I divided it into three circuit card assemblies (CCAs): Driver, Output, and Speaker Interface. Almost all of the audio circuitry is on the Driver CCA. The final output power transistors are not, as I wanted to spread them across the heatsink so each would “see” the same unshared cooling fin area. This required a separate printed wiring board (PWB) to provide connection and power supply bypassing for the 12, 5-pin TO-264 plastic packages. The Speaker Interface CCA contains the standard inductor/resistor (L-R) isolation network and a solid-state relay (SSR) for connecting/disconnecting the speaker.

The power section of the chassis contains two power transformers and five CCAs. The “main” transformer is a 2× 45V, 500VA toroid. It is turned on and off by the AC Switch CCA. The always-on “idle” transformer is a 2× 15V, 30VA toroid. It powers the Low Voltage Regulator/Protection CCA (**Photo 4**). The Capacitor Bank CCA holds 80,000μF of capacitance for filtering the unregulated ±65V used by the output stages. The High Voltage Regulator CCA has a full-wave voltage doubler powered by the main transformer from which it generates clean ±75V for the gain stages. The LED CCA is a tiny card used to mount the single tricolor LED to the front panel.

All of the MB1’s CCAs are connectorized for cabling. This provides for easy repair or replacement

| Mode             | Indicator | Description                                                                                                                                                                |
|------------------|-----------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Standby          | Red (dim) | AC power applied, amplifier off, speaker muted, low power consumption                                                                                                      |
| Power-up         | Yellow    | Initiated by Power switch or Trigger input, amplifier warming up, speaker muted                                                                                            |
| Normal           | Green     | Normal operation, speaker connected                                                                                                                                        |
| Fault            | Red       | Momentary clipping/distorion: 0.5 sec red flash per event, speaker connected<br>Sustained clipping/distortion or other fault detected: solid red indication, speaker muted |
| Thermal Shutdown | Blue      | Power or Trigger On: bright blue indication<br>Power or Trigger Standby: dim blue indication<br>Amplifier off, speaker muted, low power consumption                        |

Table 2: MB1’s various modes and status indications are detailed here.

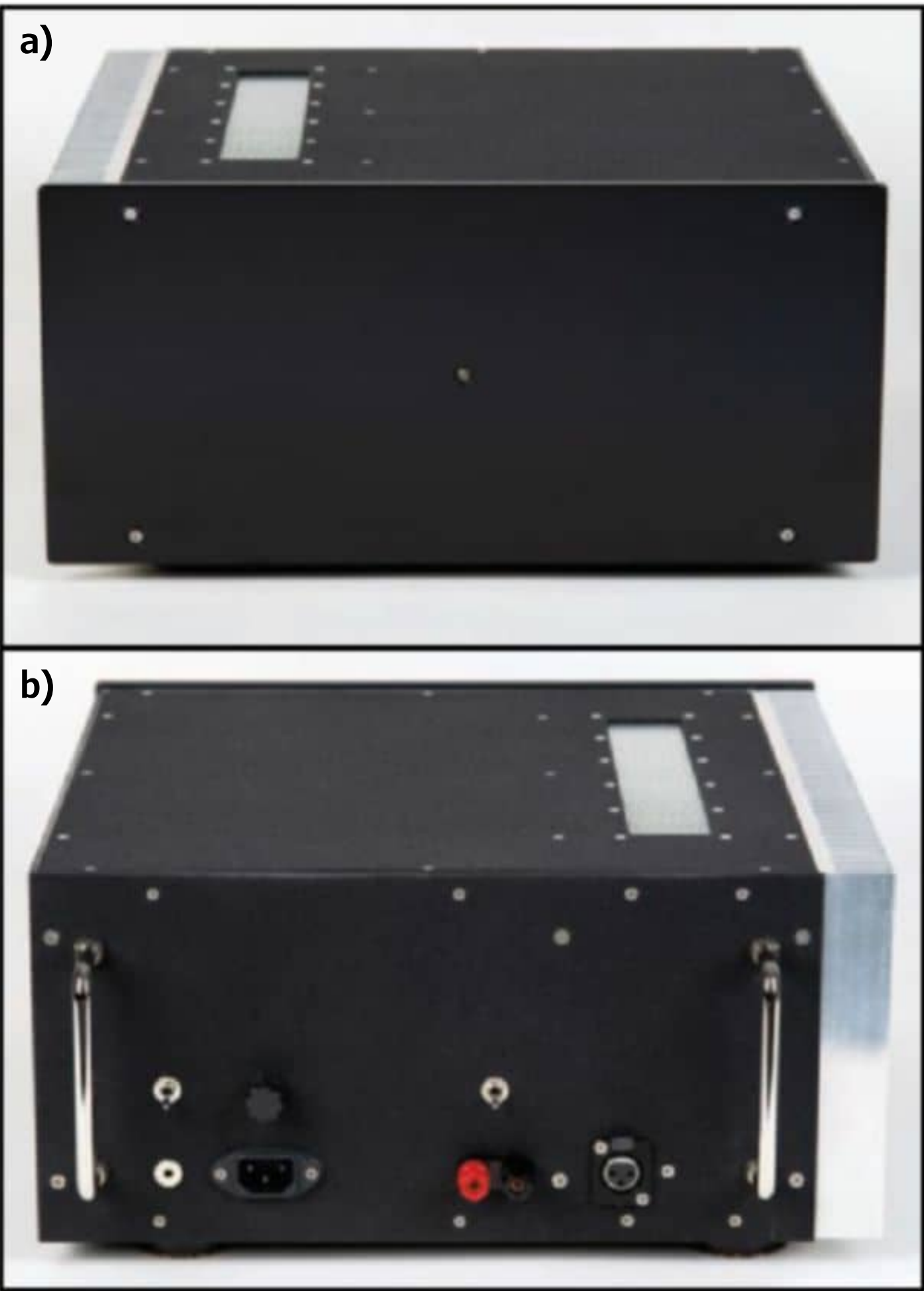


Photo 2: Here is a front view (a) and back view (b) of the completed MB1 Monoblock Amplifier.

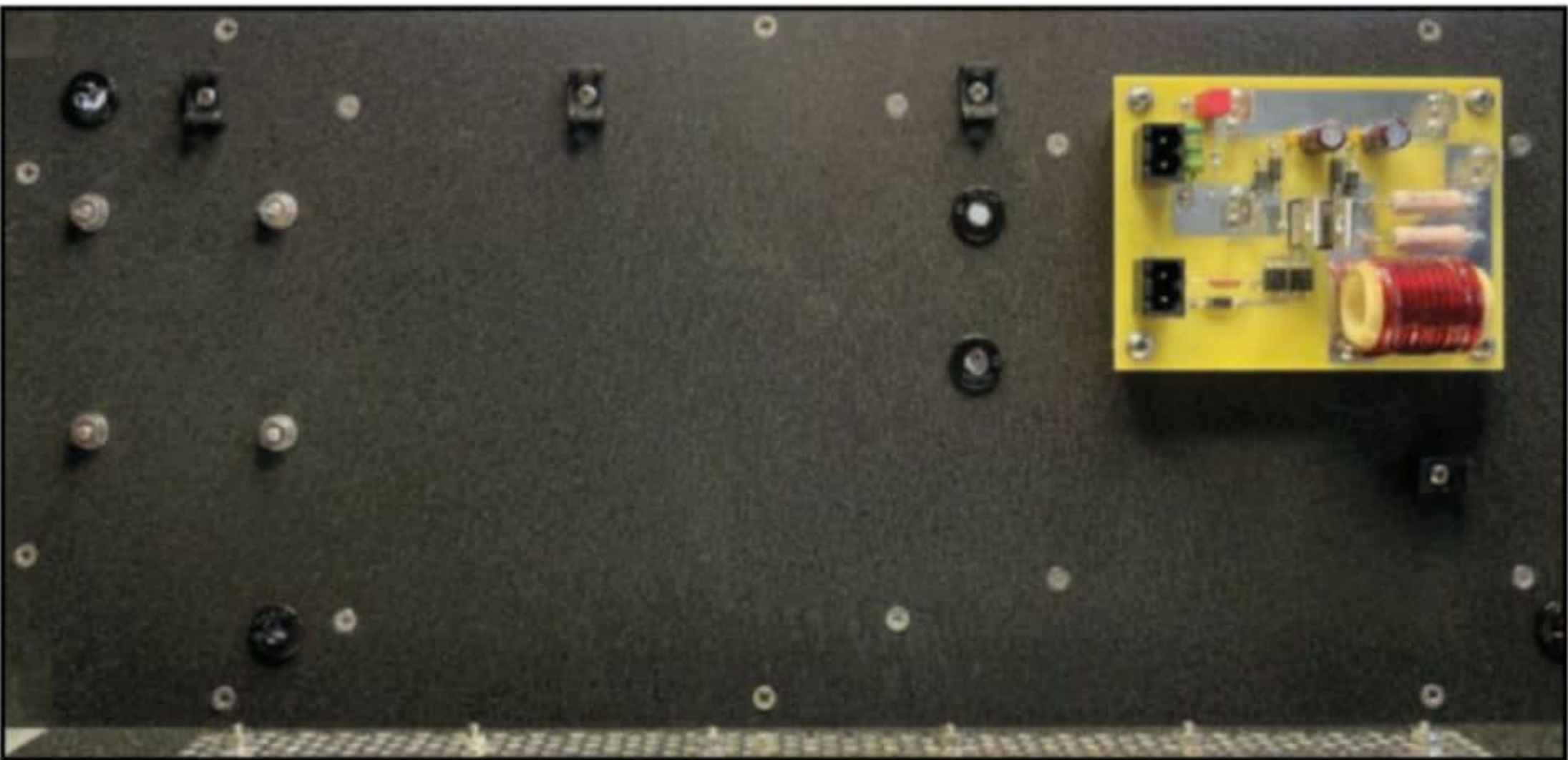


Photo 3: This is the audio path chassis partition with only the Speaker Interface CCA installed.

for upgrading. Next, I will describe the MB1's circuitry in detail by function.

## Audio Path

**Figure 1** shows the schematic diagram of the MB1's audio path from input to voltage amplifier stage (VAS) output. Functions are separated in the diagram by the font color of the reference designations and part values.

MB1's architecture is that of an instrumentation amplifier. Both differential inputs (+IN, -IN) are filtered by two-pole passive low-pass filters with a cutoff frequency of 150kHz (orange font). The design of the filters assumes a 600Ω balanced source (300Ω to ground on each input). The use of two separate filters, instead of a single balanced (floating) filter, ensures that both differential and common mode ultrasonic frequencies are suppressed. Although any mismatch between the filters can degrade the amplifier's CMR, this only occurs at very high frequencies. The MB1's CMR is greater than 80dB at 1kHz and is still 55dB at 20kHz. This is more than adequate for home audio usage.

I selected an OPA2132 dual high-speed FET-input op-amp for the the buffer amplifiers. Although this is an old design, it has excellent AC and DC performance and is available in a through-hole PDIP package. My goal was to use all through-hole parts to ease construction with aging eyes

and hands. I was mostly successful; there are only two surface-mount components in the MB1. The OPA2132 permits a high input impedance (200 kΩ differential at DC).

The main amplifier's input stage (IPS) and VAS are shown in blue font. These stages are powered from fully regulated  $\pm 75V$  supplies for low noise and predictable behavior. The IPS uses a single differential transistor pair, U1a and U1b. An Analog Devices SSM2212RZ was selected here for its ultra-low noise and excellent matching. With only a 40V rating and fairly high collector-to-base and collector-to-collector capacitance, cascoding is required (Q9, Q10). Q13 and Q15 form a high-performance current sink for the differential pair tails. The use of current source collector loads with a common mode feedback loop for stabilization was "borrowed" from Bob Cordell [4].

I chose to take the feedback from the emitters of the second differential pair per Samuel Groner rather than from the emitter follower buffers (Q5, Q6) [5]. This was done as simulations showed it to reduce distortion in the emitter followers. It also provides an additional 0.65V of headroom for the IPS, further reducing distortion. The loop forces the sum of the IPS load currents to equal the tail current. It also stabilizes the mean IPS collector voltage, and, hence the tail voltage and current of the differential VAS. Resistor R13 is a differential load for the IPS, which defines the gain and extends the bandwidth of this stage. This loading is fairly light given the 2mA per side IPS bias.

Emitter follower buffers Q5 and Q6 prevent the VAS from loading the IPS. The VAS is a cascode differential pair (Q3, Q4, Q7, and Q8) with a tail current of approximately 15mA per side. Common base stage Q11 reduces the power dissipation in Q7 and matches it better to Q8. Current mirror transistors Q16 and Q17 "reflect" the current from one side of differential pair to provide a symmetrical drive for the output stage. "Helper transistor" Q12 is an emitter follower buffer that provides the base bias for Q16 and Q17, improving the accuracy of the current mirror. Q14 makes a cascode with Q17. The collectors of Q8 and Q14 are connected together via D2-D5 and the bias generator. Diodes D1-D6 form simple Baker clamps to prevent saturation of the VAS, Driver, and Output Stage (OS) transistors. Resistors R23 and R30 lightly load the VAS output to define its gain and extend its bandwidth.

The amplifier's Miller compensation capacitor is C7 (33pF). Although it encloses three transistors in the feedback loop, emitter follower Q6 and the Q4/Q8 cascode, no evidence of instability was found in simulations or on the bench. The inclusion and



Photo 4: The High Voltage Regulator CCA (left) and Low Voltage Regulator/Protection CCA (right) are mounted on the chassis partition.

## About the Author

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



The “Must Have” reference for  
loudspeaker engineering professionals.

# Home, Car, or Home Theater!

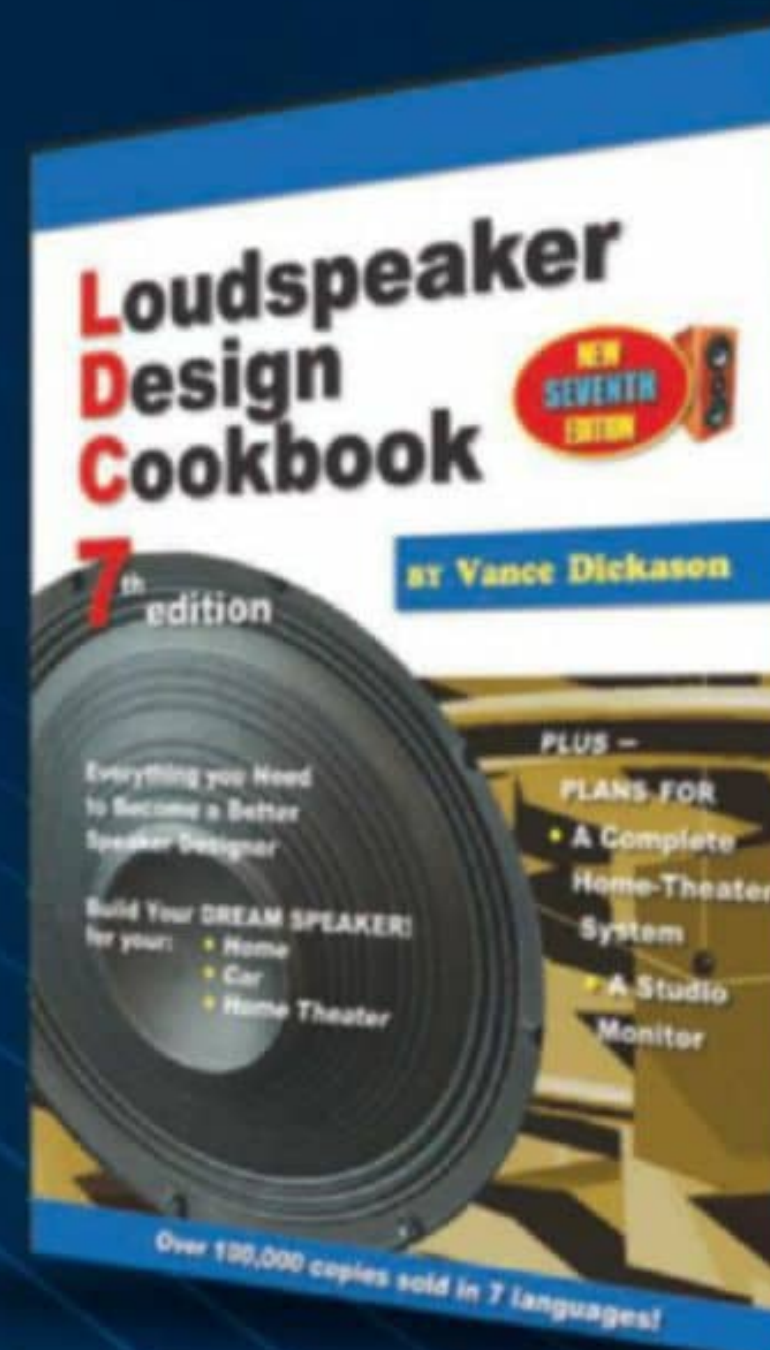
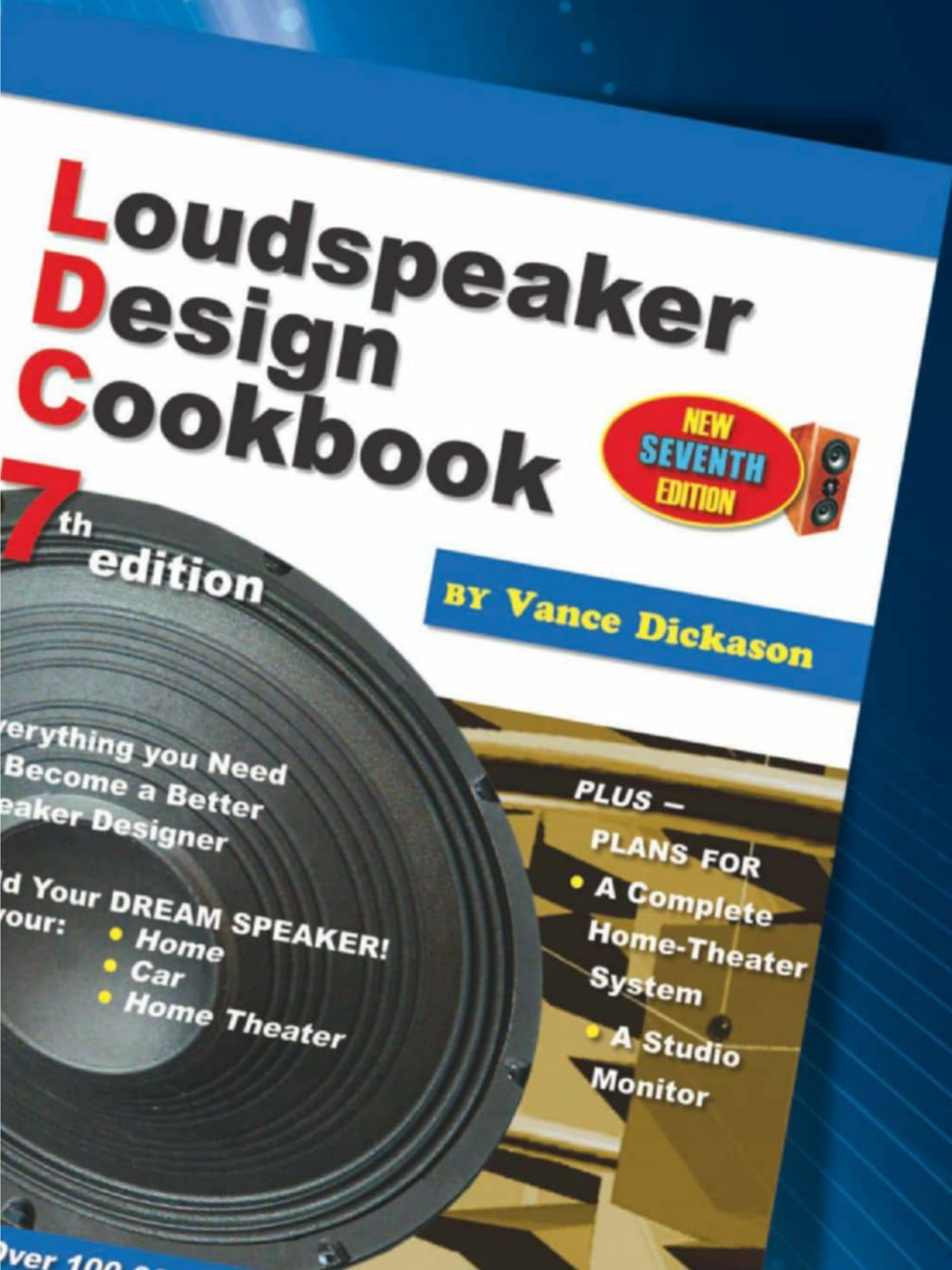
Back and better than ever, this 7th edition provides everything you need to become a better speaker designer. If you still have a 3rd, 4th, 5th or even the 6th edition of the *Loudspeaker Design Cookbook*, you are missing out on a tremendous amount of new and important information!

**Now including: Klippel analysis of drivers, a chapter on loudspeaker voicing, advice on testing and crossover changes, and so much more!** Ships complete with bonus CD containing over 100 additional figures and a full set of loudspeaker design tools.

**A \$99 value!**

**Yours today for just \$39.95.**

Shop for this book, and many other audio products, at  
**[www.cc-webshop.com](http://www.cc-webshop.com).**



location of C6 (33p) is per Samuel Groner [5]. Zobel network R22+C9, suggested by Bob Cordell, provides a slight improvement in phase margin and helps maintain a low VAS output impedance at high frequencies, even during clipping [6]. This helps ensure stability of the OS.

The bias generator (green font) utilizes a compound/Sziklai pair (Q1, Q2). This provides lower impedance than the standard single transistor VBE multiplier. The value of C1 was chosen to flatten the impedance curve in simulation. A very large value is not optimum as it would introduce an additional low-frequency pole in the bias control loop. U1b, the phototransistor in an optocoupler, enables the bias control loop to automatically adjust the bias generator setting. R2 is a manual bias adjustment

for checkout and calibration purposes and to set the maximum bias current if the loop should fail. Zener diode Z1 was included as a secondary fail-safe in the case of bias generator failure.

**Figure 2** shows the schematic diagram of the MB1's path from VAS output to the isolation network output. The safe operating area (SOA) protection circuit on the Driver CCA is shown in the magenta font. There are two types of output current limiting implemented. The first is "natural current limiting" as per Bob Cordell to limit peak current [7]. This simple technique is very fast and clean. The second is dual-slope VI limiting. Here, the natural current limiter protects the output stage until the more sophisticated, but slower, VI limiter can react. Natural current limiting is implemented

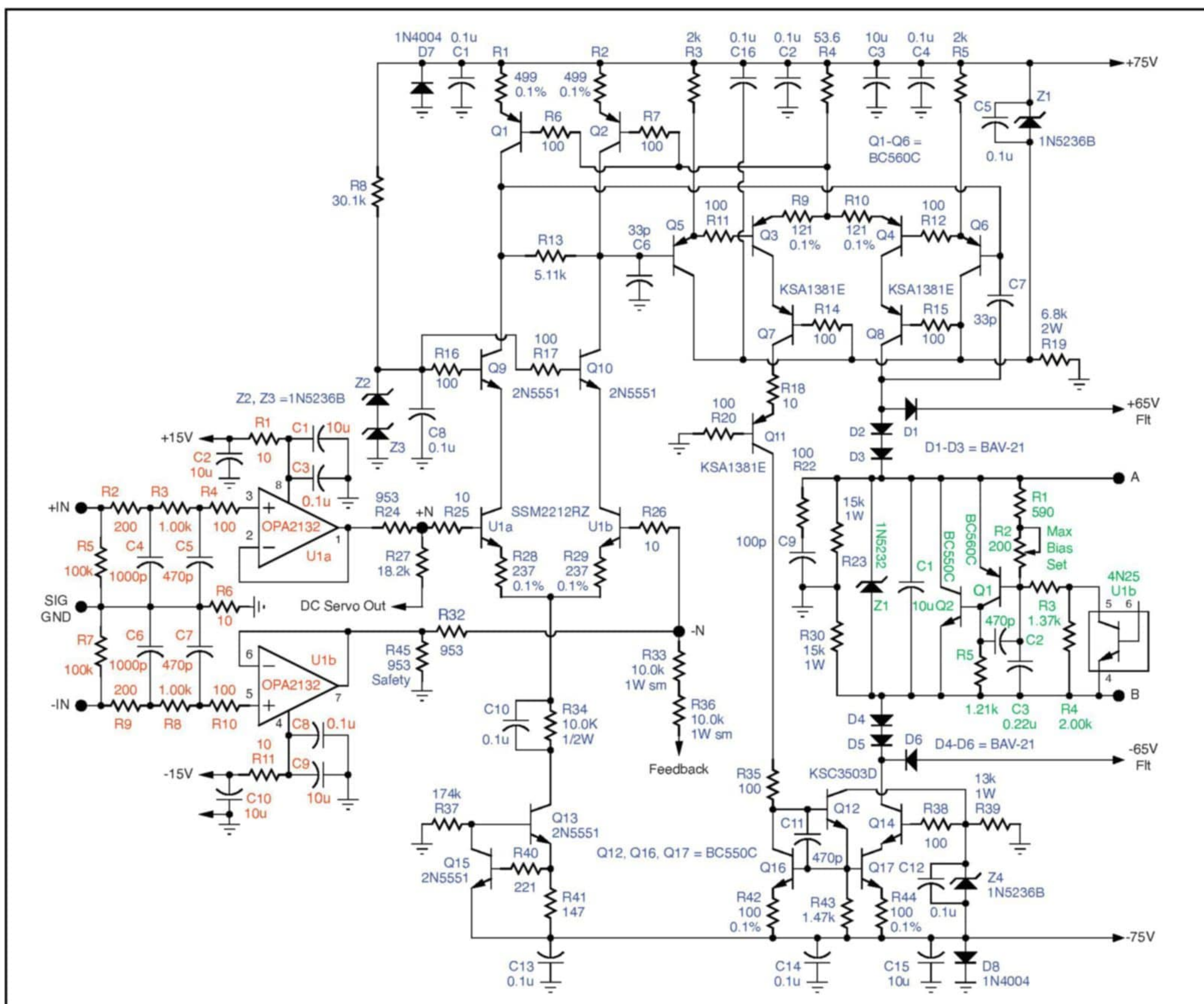


Figure 1: This schematic shows the Audio Path—input to VAS output.

by diodes (D4, D6). They limit amplifier output current on transients to  $\pm 33\text{A}$  maximum. Compound/Sziklai pairs (Q1-Q2 and Q3-Q4) are used in the VI limiter for their high current gain to sharpen the clamping action and guarantee that the 30mA VAS current can be fully shunted to the output bus. The short circuit current is set by the value of the final stage emitter resistors ( $0.33\Omega$ ) and the VBE of Q2 and Q3. With six pairs of output transistors, this results in a total steady-state short circuit current of approximately  $\pm 10.3\text{A}$  at room temperature.

The output stage is a Class-AB Locanthi Triple. The pre-driver (Q1, Q4) and driver (Q2, Q3) emitter follower stages are shown in teal font. These power transistors are mounted to heatsinks on the Driver CCA. The final stage of the triple (red font) is on a different CCA (Output). For clarity, only two of the six transistor pairs are shown. Each stage has  $\pm 65V$  power supply decoupling to improve stability.

The approach to achieving unconditional stability in the triple was shamelessly “borrowed” from “The YAP Output Stage” where Ovidiu Popa did some very nice simulation work [9]. Key are the poles introduced in the driver stage by R4, R6, and C5 on the positive side and R13, R16, and C7 on the negative side. Resistors R8 and R11 are included so the Bias Controller, also on this card (not shown in Figure 1 or Figure 2), can measure the driver output currents.

The Output Stage CCA provides connection to the six pairs of heatsink-mounted power transistors and hosts their associated passive components. The

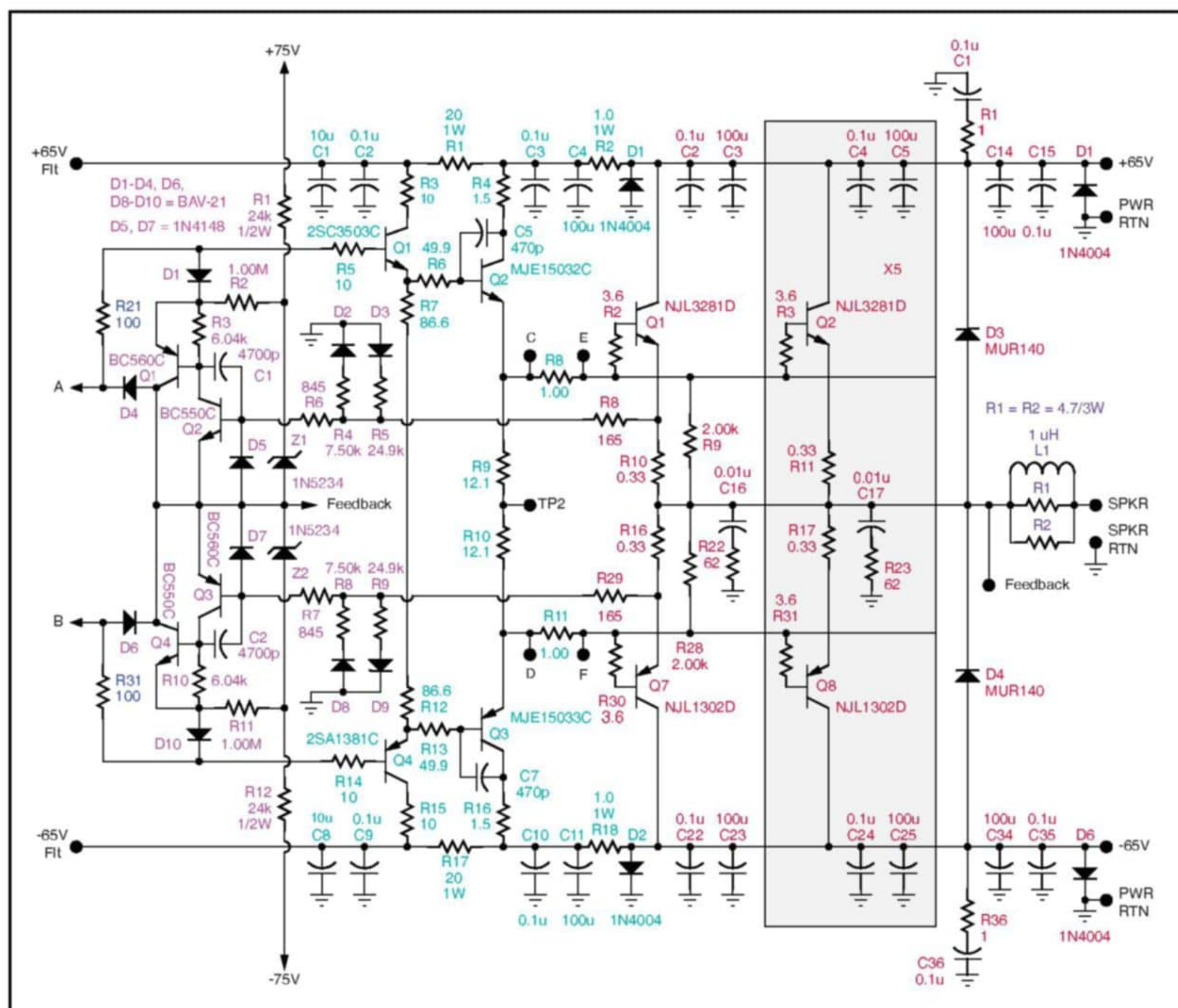


Figure 2: This schematic shows the Audio Path—VAS output to the isolation network output.

transistors are ThermalTrak devices in 5-pin TO-264 plastic packages. Only one pair of transistors (Q1, Q7) is used for current and temperature sensing. These transistors are mounted top row center of the heatsink, which is the hottest location. Using only one pair for sensing simplifies the layout and allows initial amplifier integration to be performed with a single transistor pair and still have full functionality.

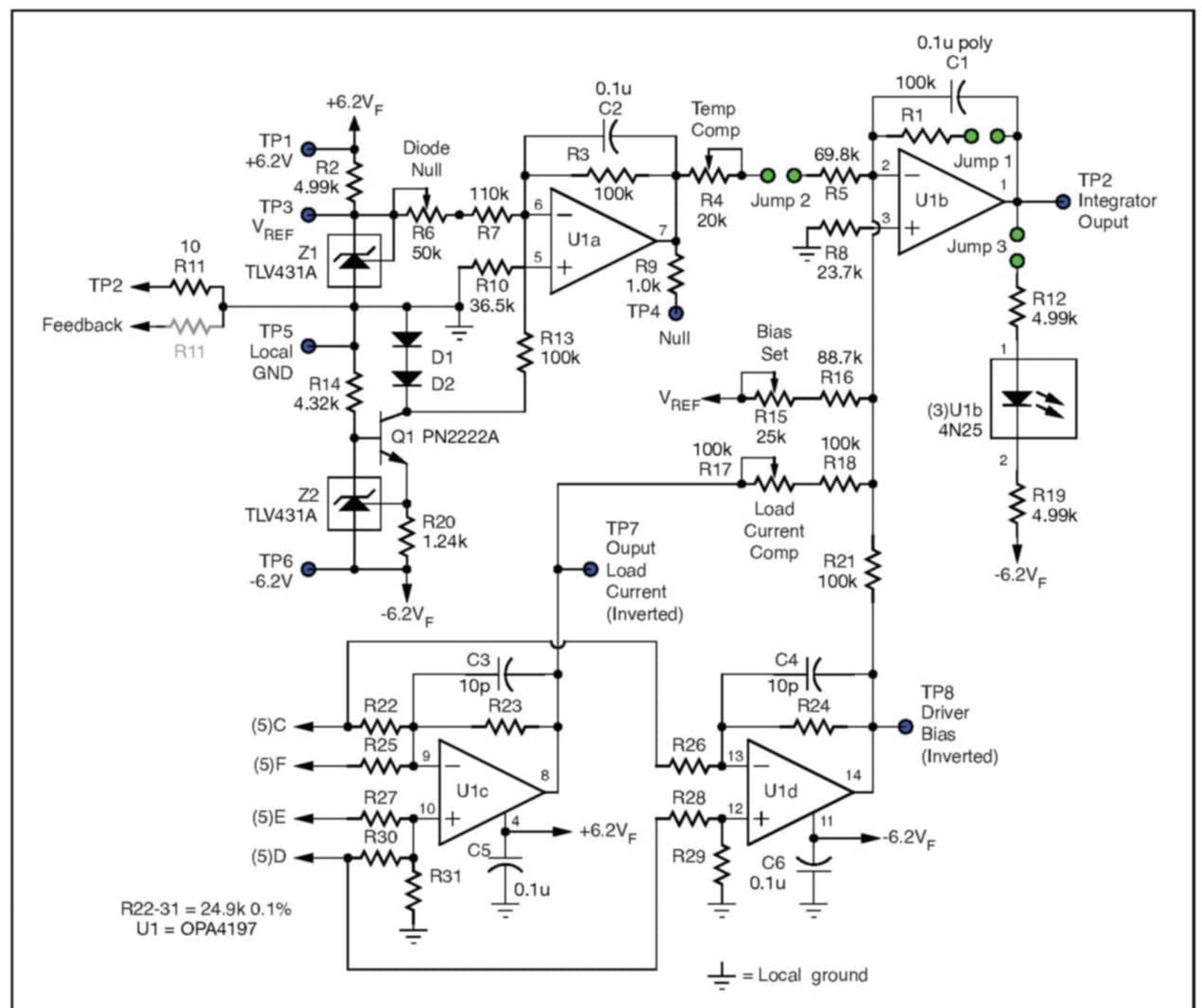
The temperature sensing diodes internal to Q1 and Q7 are put in series for use by the Bias Controller. R8 and R29 are actually part of the VI current limiting circuit on the Driver CCA. They are on this CCA to minimize capacitive loading of the emitters.

Each transistor pair has its own set of bypass capacitors (100 $\mu$ F, 0.1 $\mu$ F) and Zobel network (0.01 $\mu$ F, 62 $\Omega$ ). Emitter resistors are 0.33 $\Omega$  for good current sharing and simplification of the “natural” and VI current limiting circuits. Catch diodes D3 and D4 protect the output stage from load-induced voltage spikes and diodes D1 and D6 protect the output stage from voltage reversal. Resistors R9 and R28 ensure that the power transistors turn off if the input drive is removed.

Great care was taken in the layout of the main heatsink and the Output CCA to minimize system noise, ensure stability of the output stage, and optimize thermal performance. Suggestions provided by Bob Cordell in his book and online at the diyAudio forums were followed closely [10, 11, 12]. Input power ( $\pm 65$ V) and power return are tightly twisted together in the cabling to the card. A secondary star ground was created on the Output card for the Driver CCA and the speaker return.

The complementary pairs of output transistors are mounted next to one another on the heatsink, rather than having all of the NPNs on one side and the PNP on the other. The positive supply is run on one side of the card and the negative on the other, directly underneath, minimizing the loop area between them and allowing a tight layout of each transistor pair, their bypass capacitors, and the Zobel network.

The 100 $\mu$ F capacitors on each rail at each transistor pair have their return sides connected together with a heavy trace. The center of the trace is then connected to power return. This is to resolve rail-to-rail noise currents before entering the ground bus. The 0.1 $\mu$ F bypass caps go directly to power



return to minimize inductance and maintain a low impedance at very high frequencies. The output L-R isolation network (purple font shown in Figure 2) protects the amplifier from capacitive loads. It is mounted on the Speaker Interface CCA.

**Auxiliary Audio Circuitry**  
**Bias Controller**

The MB1 employs a new precision Bias Controller that I designed for this project [13]. It is located on the Driver CCA with its own isolated ground plane. The Bias Controller functions by stabilizing the voltage between the emitters of the driver transistors with a closed control loop, leaving only the output transistors to compensate open loop. The latter is accomplished by using the internal sense diodes in the ThermalTrak transistors to adjust the set point of the pre-driver/driver control loop. The gain in this path can be adjusted to match the temperature coefficient of the compensation to that of the transistors, providing a high level of precision.

The bias control circuitry, shown in **Figure 3**, is built around a single quad op-amp (OPA4197). This device has excellent DC precision, low input bias current, good slew rate, and consumes little

power. The Bias Controller is powered by  $\pm 6.2V$ . The power return “rides” on either the output bus or, alternatively, the center tap between the driver transistors. The selection is made by the installation position of R11. I chose the connection to the drivers rather than the output. This minimizes the common mode voltages that must be rejected, and maintains a reference for the bias control circuitry in the event the connections between Driver and Output CCAs are disrupted.

U1d is configured as a standard unity-gain, single op-amp, differential amplifier. It measures the voltage between the emitters of the two driver transistors ( $V_{C-D}$ ). This voltage appears, in inverted form, at TP8. A proportional current is drawn from the control loop filter (an integrator) through R21. The integrator (U1b) compares this negative current to a positive one injected from precision voltage source Z1 through R15 and R16. If the two currents are not equal, the output voltage of U1b will move toward one power supply or the other. This changes the optocoupler LED current in the bias generator (Figure 2). This, in turn, changes the current in the optocoupler’s transistor, the voltage across the bias generator, and  $V_{C-D}$ . This establishes a closed

**CELEBRATING OVER 40 YEARS!**

**ACO**USTICS BEGINS WITH **ACO**™



**MEASUREMENT MICROPHONES**  
Polarized & Electret

Windscreens  
3 to 18 inch & Custom  
Weather/UV Resistant



Community/Industrial  
Noise Monitors/Alarms  
Indoors and Out

24/7 Schedule



Microphone Systems  
Standalone  
IEPE Powered  
Phantom Power  
Custom

**ACO Pacific, Inc** Tel: +1-650-595-8588  
www.acopacific.com sales@acopacific.com  
**Serving End-Users and OEMs Worldwide**

**DO IT YOUR WAY**

**Panels/Boxes/Cables**  
WITH REDCO AUDIO'S  
“Build Your Own” web pages

Full control for design, editing and more.  
**WE’LL BUILD IT JUST THE WAY YOU WANT!**



**REDCO AUDIO**

www.redco.com 203-502-7600

control loop that tightly regulates  $V_{C-D}$ .  $V_{C-D}$ , which sets the bias current in the final output stage, can be adjusted via R15.

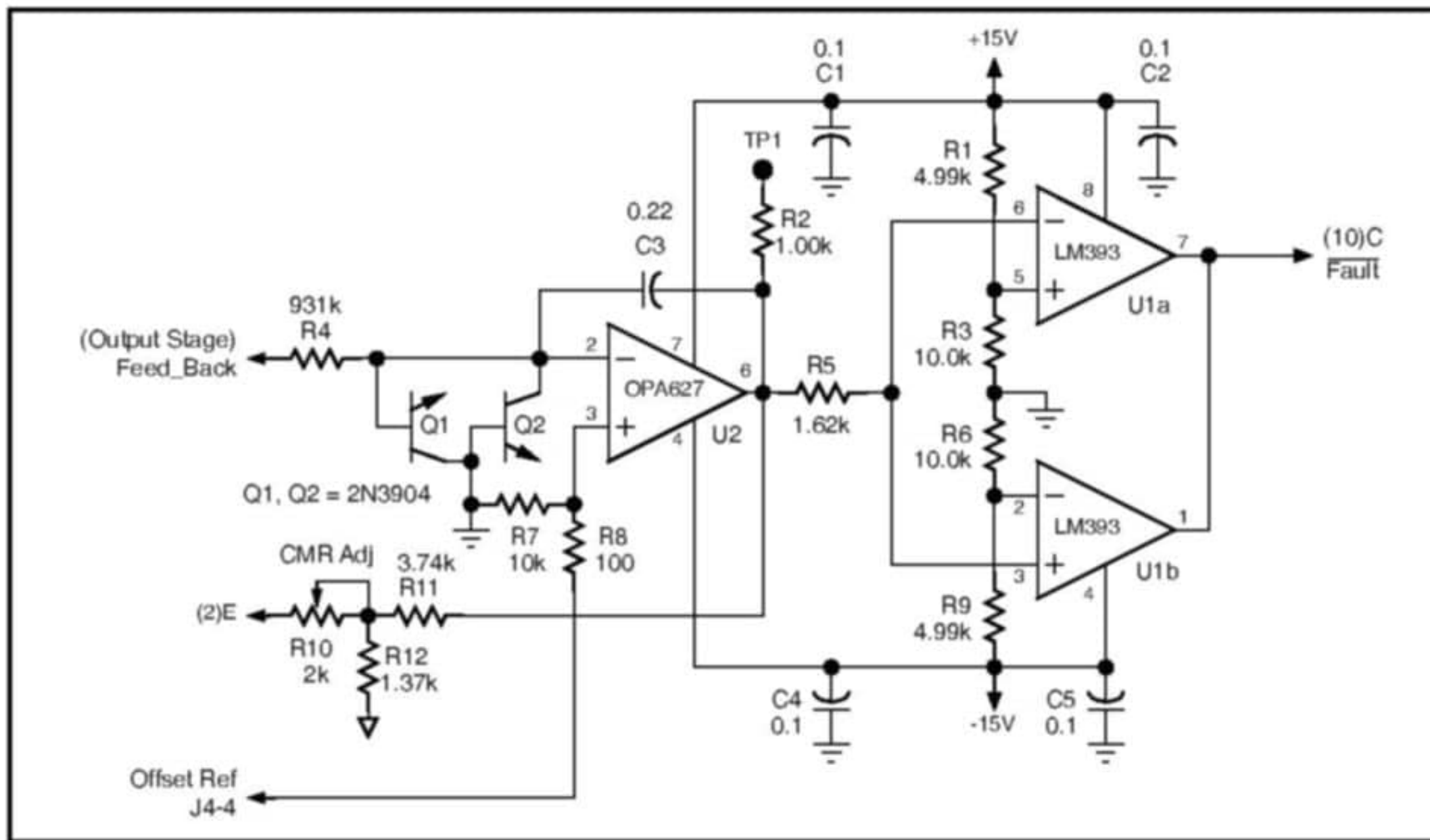


Figure 4: Here is the DC Servo circuitry located on the Driver CCA.

## References

- [1] M. Otala, "Transient Distortion in Transistorized Audio power Amplifiers," *IEEE Transactions on Audio and Electroacoustics*, Volume AU-18, pp. 234-239, September, 1970.
- [2] B. Cordell, "Another View of TIM [Part I]," *Audio*, pp. 38-49, February 1980.
- [3] B. Cordell, "Another View of TIM [Part II]," *Audio*, pp. 39-42, March 1980.
- [4] B. Cordell, "A MOSFET Power Amplifier with Error Correction," *Journal of the Audio Engineering Society (JAES)*, Volume 32, No. 1/2, January/February 1984.
- [5] S. Groner, "A new audio amplifier topology with push-pull transimpedance stage," *Linear Audio*, Volume 2, September 2011.
- [6] B. Cordell, *Designing Audio Power Amplifiers* (Second Edition), p.89, Routledge, New York, 2019.
- [7] B. Cordell, *Designing Audio Power Amplifiers* (Second Edition), p. 432, Routledge, New York, 2019.
- [8] M. Kiwanuka, "The Safe Operating Area (SOA) Protection of Linear Audio Power Amplifiers," *Electronics World*, Volume 118, pp. 20-35, May 2012.
- [9] O. Popa, "The YAP Output Stage," Yet Another Poweramp, [www.synaesthesia.ca/YAPOPstage.html](http://www.synaesthesia.ca/YAPOPstage.html).
- [10] B. Cordell, "Bob Cordell's Power Amplifier Book," #9210, diyAudio, [www.diyaudio.com/community/threads/bob-cordells-power-amplifier-book.171159/post-5760308](http://www.diyaudio.com/community/threads/bob-cordells-power-amplifier-book.171159/post-5760308).
- [11] B. Cordell, "Hafler DH-200/220, Mods," #1864, diyAudio, [www.diyaudio.com/community/threads/hafler-dh-200-220-mods.31131/post-6229529](http://www.diyaudio.com/community/threads/hafler-dh-200-220-mods.31131/post-6229529).
- [12] B. Cordell, "Bob Cordell's Power amplifier book," #10519, diyAudio, [www.diyaudio.com/community/threads/bob-cordells-power-amplifier-book.171159/post-6732562](http://www.diyaudio.com/community/threads/bob-cordells-power-amplifier-book.171159/post-6732562).
- [13] B. Gillingham, "Novel Precision Bias Controller For Class AB Power Amplifiers," #1. diyAudio, [www.diyaudio.com/community/threads/novel-precision-bias-control-system-for-class-ab-power-amplifiers.388015/post-7066526](http://www.diyaudio.com/community/threads/novel-precision-bias-control-system-for-class-ab-power-amplifiers.388015/post-7066526)
- [14] B. M. Oliver, "Distortion in Complementary Pair Class B Amplifiers," *Hewlett-Packard Journal*, pp. 11-16, February 1971.
- [15] LTspice, developed and distributed by Linear Technology Corp./Analog Devices, Inc., [www.analog.com](http://www.analog.com).

In order to compensate  $V_{C-D}$  for temperature-induced changes in the base-to-emitter voltages of the output transistors, U1a measures the voltage across the internal sense diodes (D1, D2) in one pair of ThermalTrak output transistors. D1 and D2 are biased on by a precision 1mA current source (Q1, R14, R20, Z2). R6 and R7 allow the output of U1a to be nulled at room ambient.

As the output transistors and their internal diodes warm, the voltage across the diodes decreases, driving the output of U1a negative. This pulls current from the integrator through R4 and R5, lowering the set point for  $V_{C-D}$ . R4 adjusts the temperature coefficient of the compensation to match that of the output transistors.

The remaining op-amp, U1c, is configured as a four-input summer/subtractor. It has connections to both sides of the two driver load current sense resistors (Figure 3). It outputs a voltage with an absolute value proportional to the sum of the driver's load currents. This voltage is always  $< 0$ . As the load current increases, U1c pulls more current from the integrator through R17 and R18, compensating for the natural decrease in  $V_{C-D}$  under load. R17 provides calibration for this function.

Three jumpers are included to ease initial calibration. Jumper 2 and Jumper 3 are normally installed; Jumper 1 is not. Output transistor bias was adjusted to 45mA per transistor pair, 270mA total. This value was the optimum level for THD-1k into 4Ω over a 30dB range of output level. This is pretty far from the Oliver point, but in general agreement with LTspice simulation results [14, 15].

## DC Servo Control

The MB1 has very good DC stability even without a servo or the typical series capacitor to ground in the feedback. Adding a DC servo, however, reduces the offset to below 1 mV without adjustment even with offset on the input, eliminating the need for an input capacitor. The DC Servo circuitry on the Driver CCA is shown in **Figure 4**. An OPA627 is used for the loop integrator. Although this part is an old design, it has excellent AC and DC performance. Q1 and Q2 are connected as low leakage diodes to protect U2.

Feedback is to the non-inverting side of the differential audio amplifier through an attenuator (R11, R12). This eliminates the need for an extra inversion, or a "positive integrator" using two capacitors. Since the input buffer provides a fixed low impedance source, an input coupling capacitor is not required with this connection. R7 provides a ground reference for the integrator when the CCA is in isolation. When integrated into the amplifier, R8 establishes the ground reference at the speaker

return. The LM393 dual comparator is set up as a “window” comparator. If the integrator’s output exceeds  $\pm 10\text{V}$ , a fault is declared. R10 allows optimizing the amplifier’s CMR.

Speaker Interface

Figure 5 shows the Speaker Interface CCA schematic. It includes the L-R isolation network described in Figure 2 and a solid-state relay (SSR) to connect/disconnect the speaker load. The SSR is implemented with two high-current back-to-back N-channel MOSFETs (Q1, Q2). They are turned on and off by a pair of paralleled photovoltaic MOSFET drivers. These are optocouplers with a floating output voltage. The drivers have an embedded fast turn-off circuit.

Two devices are paralleled to speed up turn-on. Diodes D1-D4 clamp any load-inductance-induced voltage spikes to the  $\pm 65\text{ V}$  rails to protect the amplifier and SSR. The small Zobel network (C1, R3) helps terminate the speaker line at RF frequencies. R4 provides a path for leakage currents to ground when the speaker is disconnected.

Summary/Looking Ahead

In this article, I have provided an overview of the MB1 and its performance, and a detailed description of all of the circuitry directly in the audio path. In Part 2, I will provide a detailed description of the remaining circuitry. I will also describe the mechanical design, and show time and frequency domain test data. 

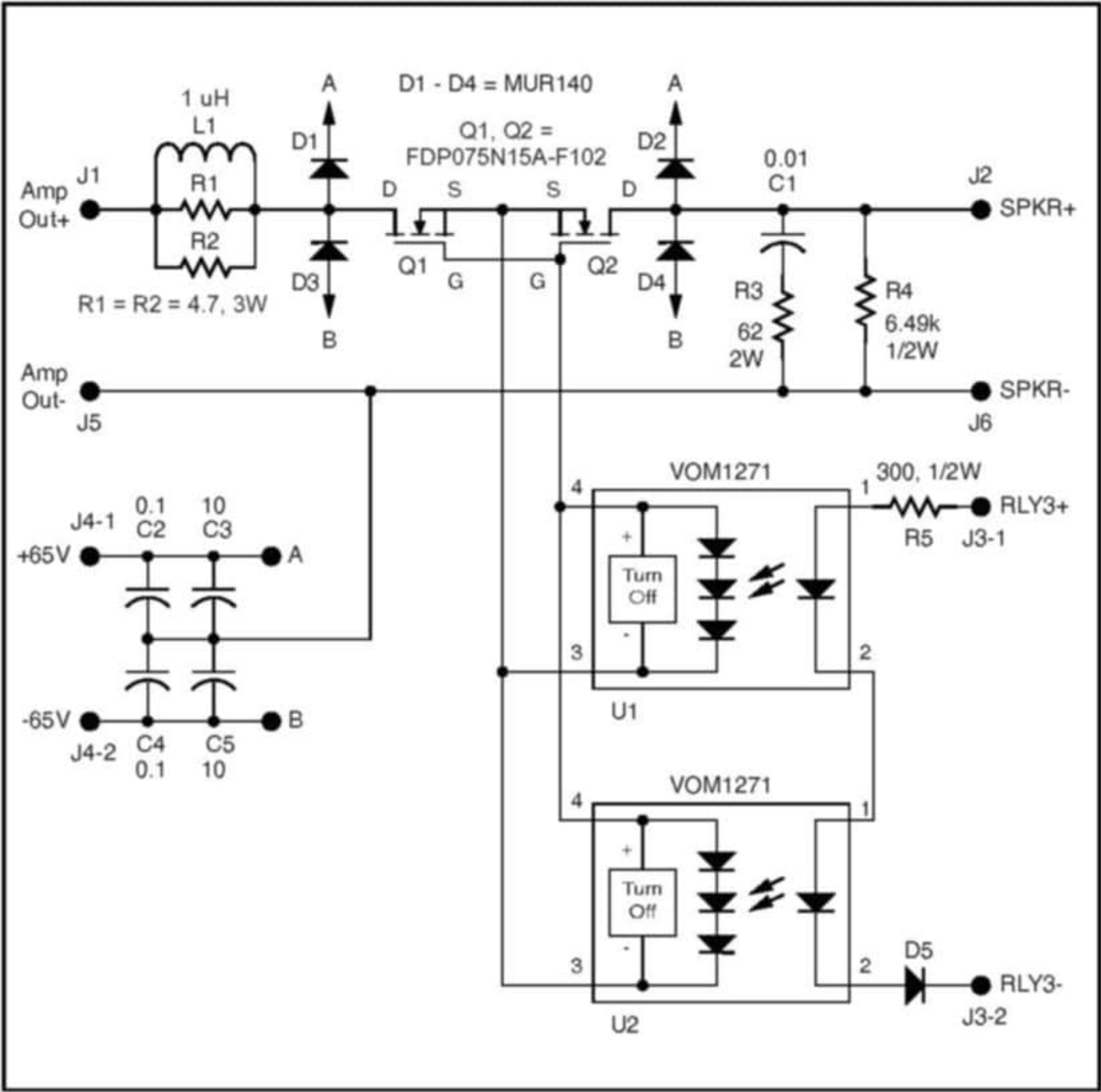


Figure 5: The Speaker Interface CCA schematic includes the L-R isolation network described in Figure 2 and a solid-state relay (SSR) to connect/disconnect the speaker load.

EXCITING NEW AUDIO TEST & MEASUREMENT SOLUTIONS FOR ELECTRONICS, SPEAKERS & HEADPHONES

A recent customer said: “.. amazing test equipment at an affordable price”!

Audio Analyzer



Custom Test Chambers

Microphone & Speaker Test Chambers



Visit: [www.spectralmeasurement.com](http://www.spectralmeasurement.com) or [www.prismsound.com/test](http://www.prismsound.com/test)  
Video Testamonal: <https://youtu.be/T93d1awPQ4o>  
Online Webinar recordings: <http://www.spectralmeasurement.com/webinars>

Contact us with your testing question: <https://www.spectralmeasurement.com/testquestion> or [sales@spectralmeasurement.com](mailto:sales@spectralmeasurement.com)



# Building a Soundbase

One of the most successful speaker categories today, soundbars were originally intended to compensate for the poor sound quality of thin-display TVs. This project takes the concept up a notch in sound quality and fidelity for any living room application.



By  
**Ken Bird**

Photo 1: The completed Soundbase shows the woofer placement, which allows up to a 65" TV to be placed on it without interference.

The Soundbase might be considered a cousin to the soundbar, which is now providing audio to millions of TV owners who don't want a fully functioning surround system with all the wires and speakers cluttering up their viewing areas. Soundbars do provide better sound than the built-in speakers and many incorporate some sophisticated digital signal processing to enhance the audio but they are not cheap and not everyone wants to pay the price for a higher degree of sound quality.

## The System

The Soundbase is a project for those people who want somewhat better audio at a lower price but not a full-blown surround system or expensive soundbar, and who might also have the need to boost the height of a TV without a wall mounting (**Photo 1**). The Soundbase has a built-in two-channel amplifier and subwoofer channel. It can handle analog audio, Bluetooth audio, and, with an optional adapter, optical audio signals from a

TV (**Photo 2**). It can also double as a music system using a CD player or other analog signal source.

The system uses eight 2" full-range speakers in a two-channel arrangement and two 4" subwoofers. The speakers have separate enclosures with the 2" speakers configured in a sealed horizontal line array and the woofers in separate ported enclosures.

The system is powered by a separate 12V 5A DC supply that plugs into the rear accessory panel. The Soundbase is 36" wide, 18" deep and 4.75" high. The front and the rear panels are painted black while the sides, the top, and the bottom are covered in a grey indoor-outdoor carpet. This provides a nice-looking finish and makes the enclosure mar proof to any furniture base it is placed on. However the builder can choose to apply any finish that is desired.

All the speakers are housed in separate enclosures within the cabinet and there are no crossover components as the amplifier includes a switchable 6dB per octave filter operating at 150Hz.

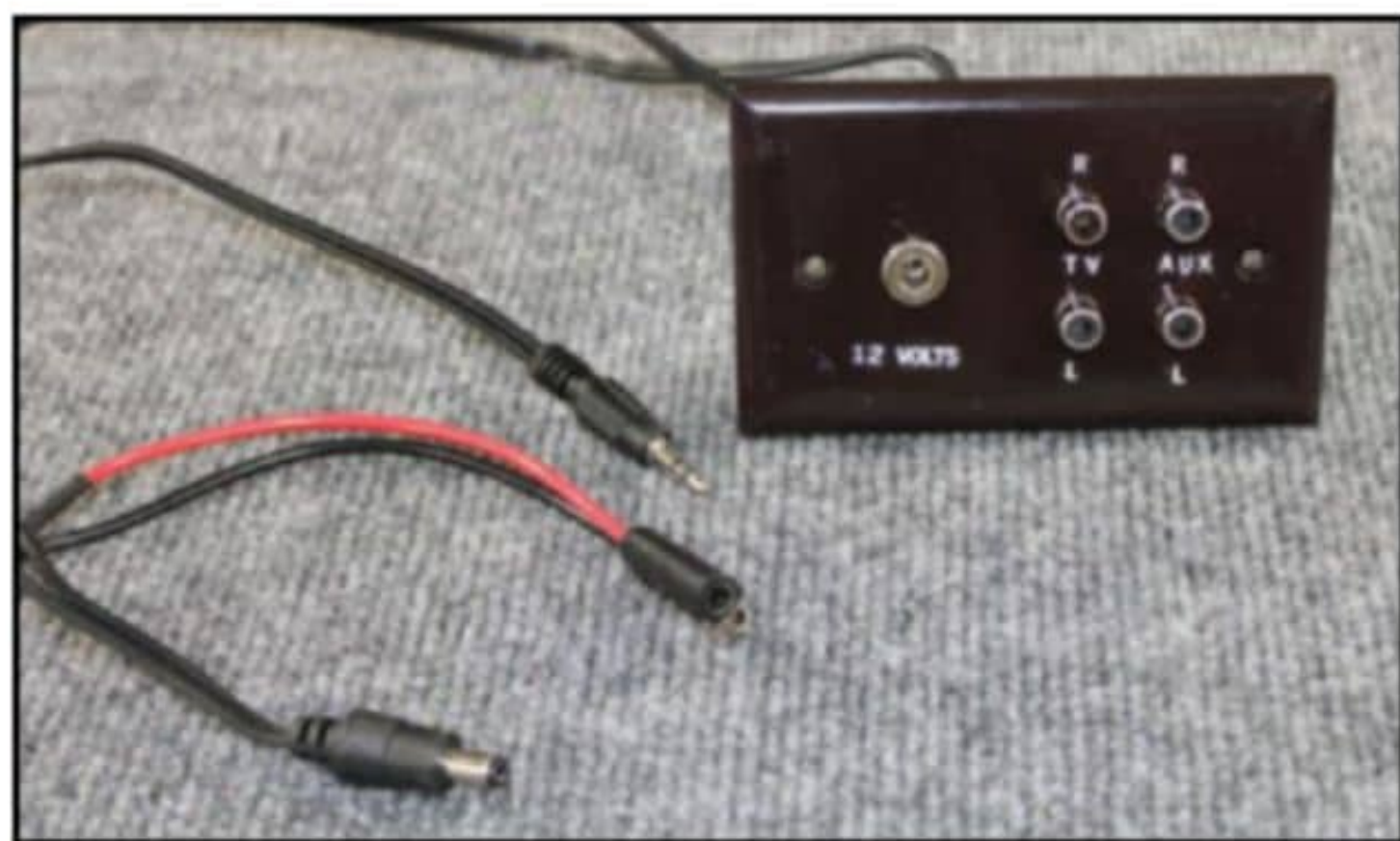


Photo 2: Detail of the rear panel interface board inputs. The amplifier only accepts analog audio but inexpensive optical-to-analog adapters are available online that will work with the newer model flat screen TVs.



Photo 3: The Peerless 2" drivers and one of two Dayton 4" subwoofers used in the Soundbase are shown here.

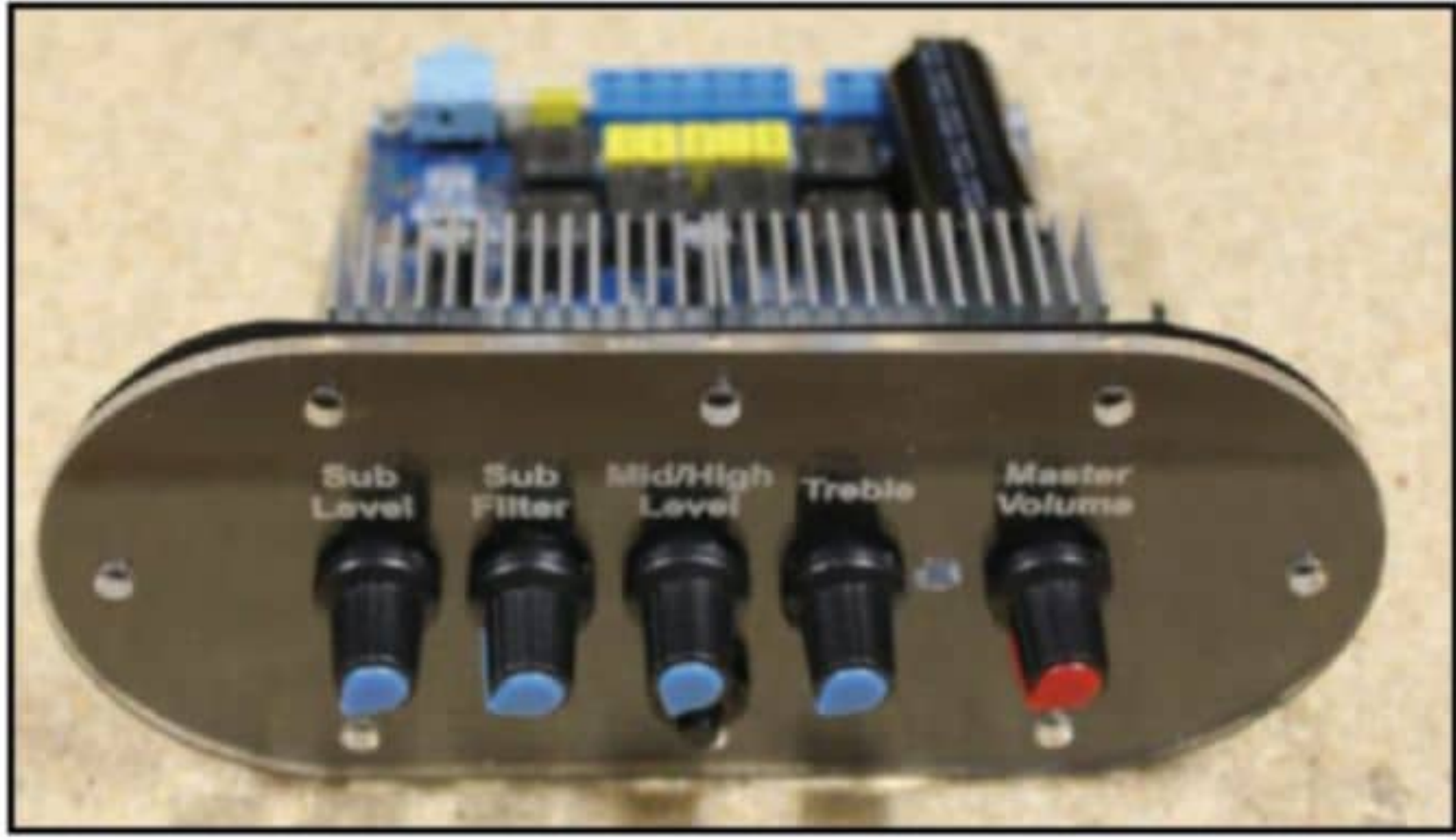


Photo 4: The Class-D 2.1 Bluetooth 5.0 Amplifier Board 2x 50W + 100W is shown with the modified mounting plate.

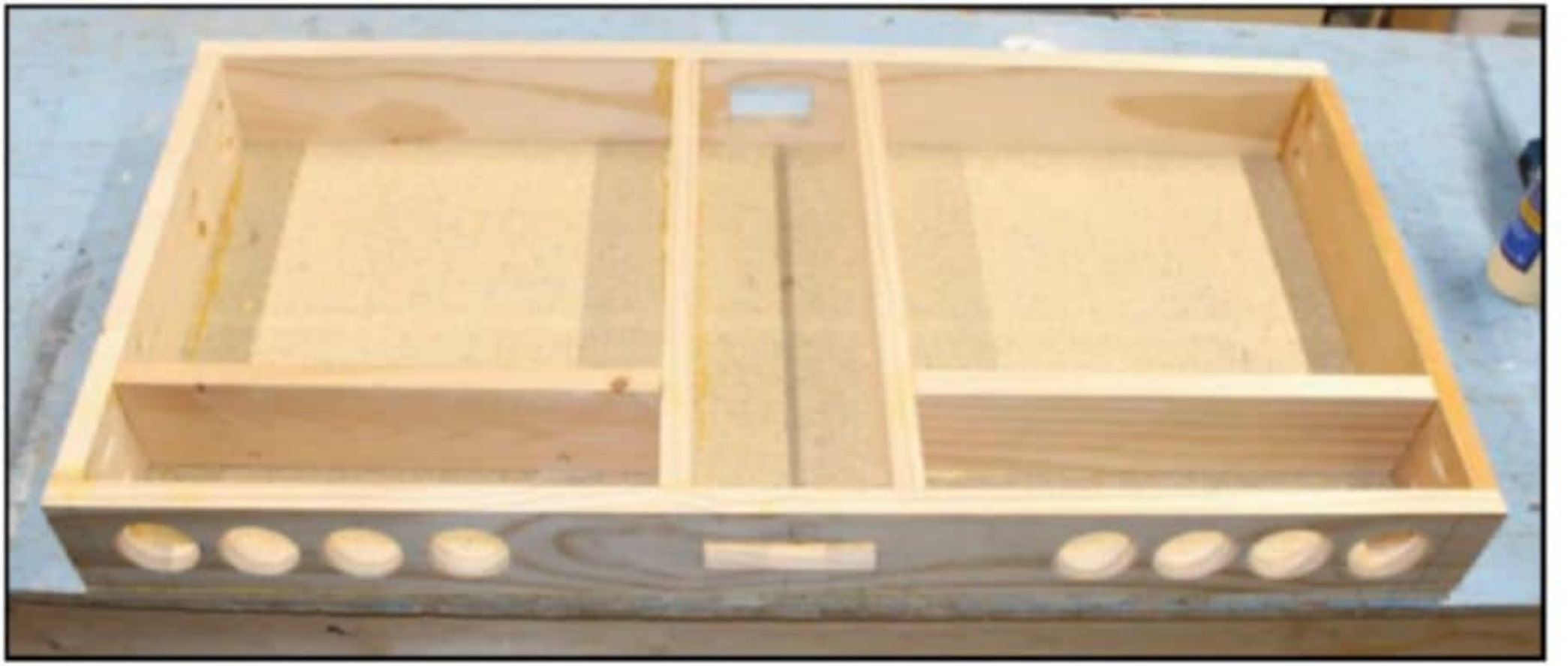


Photo 5: The Soundbase internal construction. Only the two enclosure areas are to be filled with polyester stuffing while the central amplifier area is to be left open.

## The Drivers

The Soundbase uses eight Peerless 2" drivers and two Dayton 4" subwoofers (**Photo 3**), both of which I purchased from Parts Express. The 2" models are 4Ω and have a smooth response from 100Hz to 17000Hz with good high-frequency off axis response up 4000Hz. The 4" 8Ω woofer specifications list a 55Hz to 5000Hz response.

## The Amplifier

The amplifier is a TPS3116D2 Class-D 2.1 Bluetooth 5.0 Amplifier Board 2x 50W + 100W with Filter and Volume Controls (**Photo 4**), which I also purchased from Parts Express. It produces some fine audio for the price and can be used with a 12V or 24V power adapter with a minimum of 5A. A 12V 5A unit was used with this version and provided more than enough usable audio for the project. The amp is mounted on a plastic panel, which must be purchased separately from the amplifier. See the Parts List for details.

## Soundbase Assembly

The system's enclosure is easy to assemble as it uses 1 x 4 diminishable lumber for the exterior and interior walls and 0.5" MDF or plywood for the top and bottom (**Figure 1**). Begin by cutting the 1 x 4 materials into two 36" for the front and back and four 16.5" pieces for the sides and interior tunnel. Then cut two 14" pieces for the front speaker enclosure backs (**Photo 5**).

Cut the 0.5" material for the tops making two 18" x 36" pieces. Once the pieces are cut, make the cutouts in the front and rear panels. The center cutout for the front panel is 2.75" wide and 1.25" high with a center "notch" of 0.5" x 1" to clear the on-off switch (**Photo 6**). The rear panel center cutout is 2.5" wide x 1.75" high. Also on the rear panel make two 1" port holes in each woofer enclosure section and glue in a 2.75" section of 1" PVC pipe or cardboard tube for the tuning ports.

The cutouts for the small drivers were made with a 2" hole saw (**Photo 7**). You will also need to cut out a 2" hole in the

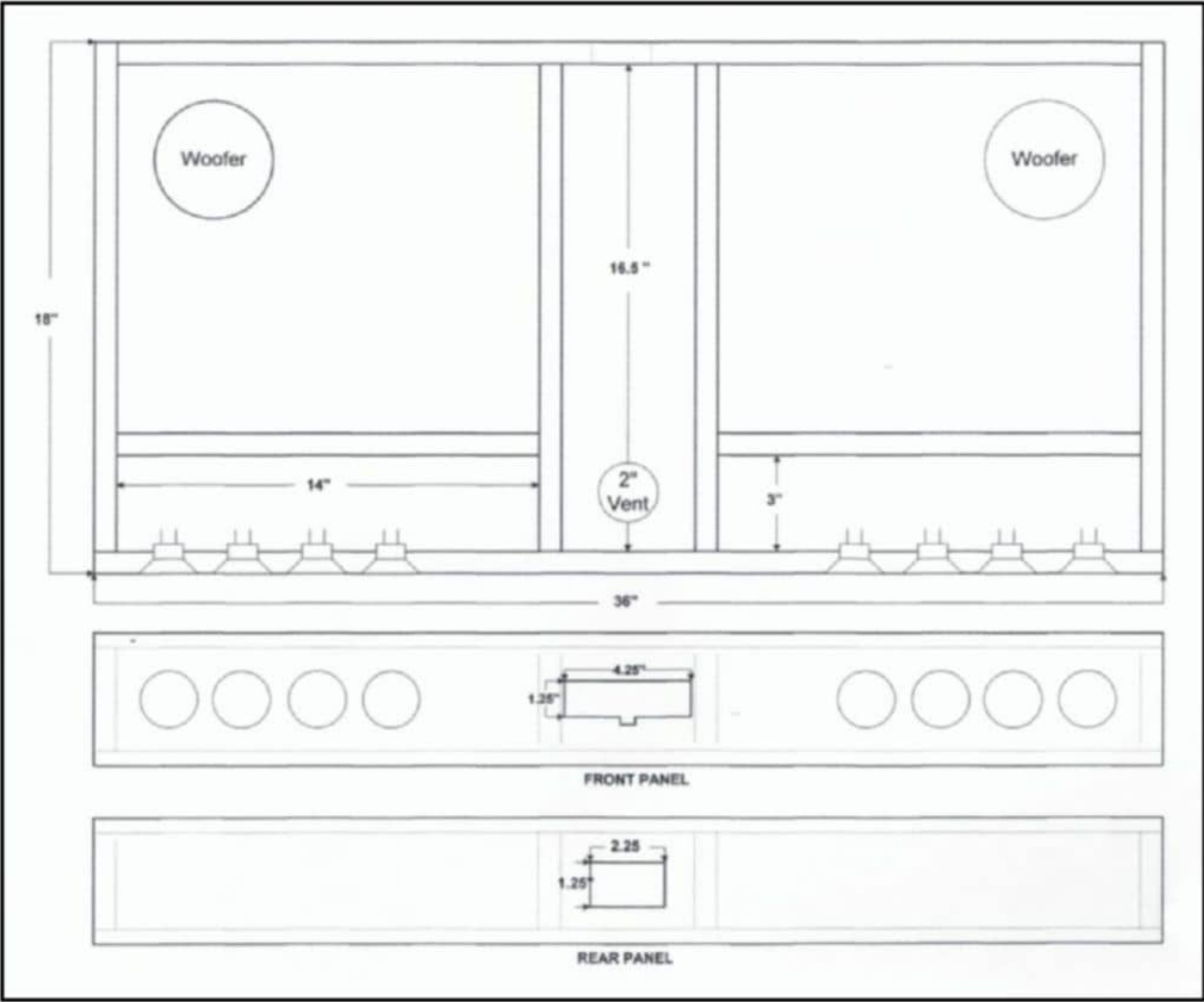


Figure 1: The system's enclosure is easy to assemble as it uses 1 x 4 diminishable lumber for the exterior and interior walls and 0.5" inch MDF or plywood for the top and bottom.



Photo 6: Detail of the amplifier mounting opening in the front panel



Photo 7: Use a 2" hole saw to cut the front panel speaker mounting holes.

center front of the top 1" from the edge for the ventilation vent. I made the woofer cutouts with a Jasper router jig. The woofer cutouts are 3.7" (9.4cm). The woofers holes should be made in the top corners, 1" from the rear and sides. The parts can be assembled with ordinary butt joints and screws or nails with a liberal amount of wood glue. I joined this system using "pocket screw joints" and a Kreg 720 pocket hole jig. It makes solid joints with no exterior screw heads showing (**Photo 8**).

Once the frame is assembled and the cutouts made, install the bottom panel first and then the interior baffle pieces. Before adding the top, pre-wire each of the enclosures with 16 or 18 gauge speaker wire, drilling 0.25" holes between the interior enclosures and the center tunnel. Make sure the wire ends are accessible near the speaker openings and leave enough wire to feed through the holes. Caulk all the enclosure section joints. Make a pad for the amplifier to sit on from a piece of the lumber scrap (**Photo 9**).



Photo 8: Corner joints in the Soundbase were glued and fastened using a "Kreg" drilling jig that leaves no exposed screw heads in the structure.

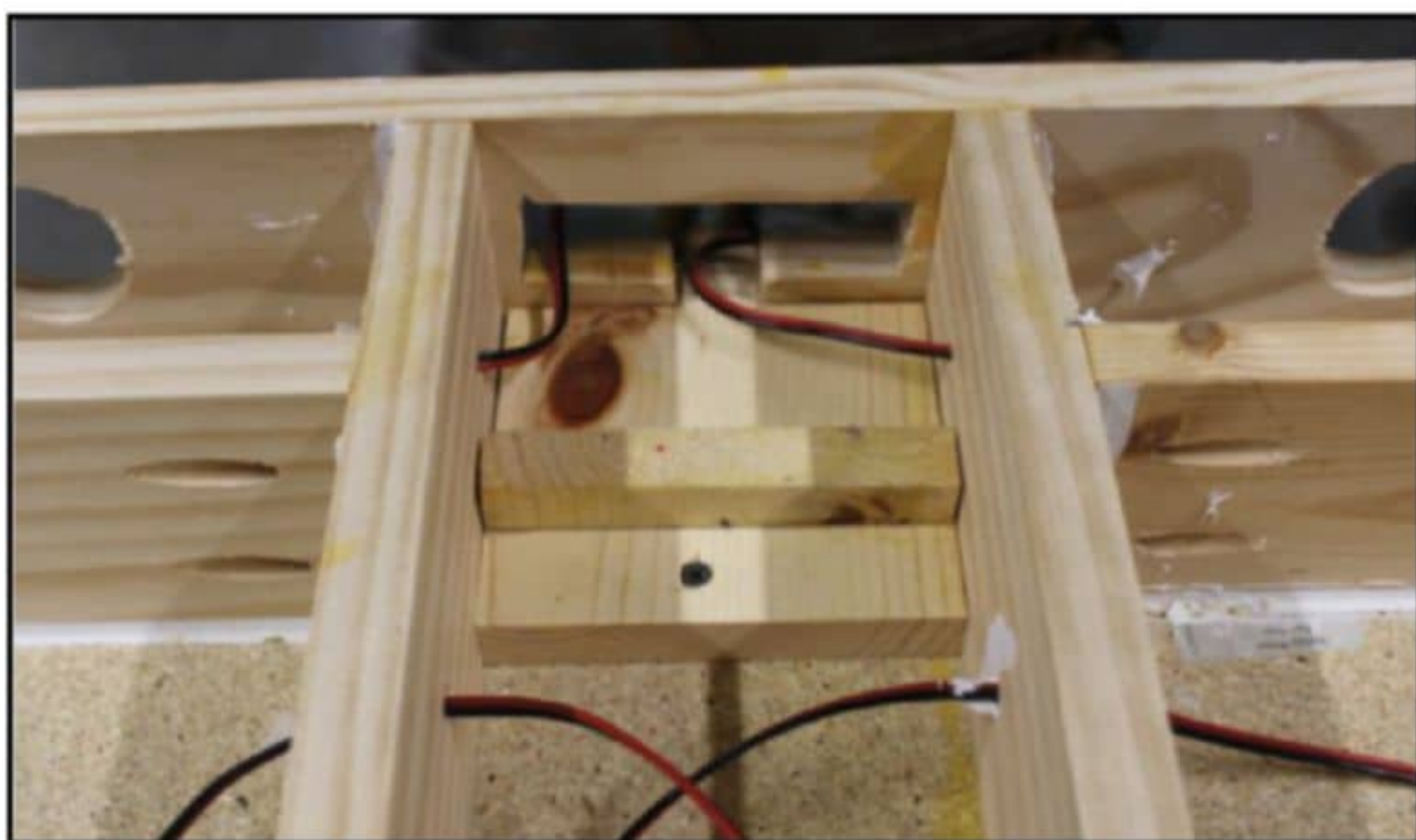


Photo 9: The amplifier base pad was made from scrap pine wood. No screws are used to mount the amp to the base. Only the front panel mounting holds the amplifier to the Soundbase frame.

Do not assemble the top of the enclosure until all the wiring is installed from the central tunnel to each enclosure and the amplifier section. Stuff the woofer enclosures with 4oz of polyester pillow stuffing, taking care not to block the ports. The tweeter-mid sections are stuffed with 3oz of the material. Once the wiring and stuffing are secured nail the top to the enclosure using a heavy bead of Liquid Nails to assure a good seal. I pre-wired the rear panel and installed after the exterior finish was applied to the enclosure the wires being taped to a stiff wire and "fished" through the central tunnel.

## Finishing the Base Enclosure

I painted the front and the rear of the enclosure with satin black paint, with only the front trimmed along the top and bottom with a 0.75" piece of decorative molding. The bottom, the sides, and the top were covered with a layer of inexpensive indoor-outdoor carpet that is carried by most of the major home stores. (See



Photo 10: The amplifier soffit vent is mounted in the center of the top over the amplifier pad.



Photo 11: The carpet covering the Soundbase was cut for the woofers after being attached to the top.

## About the Author

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



the Parts List for details.) The carpet was stapled completely around the enclosure with the matching seam centered on the bottom. Once the carpet is stapled, carefully locate the woofer and front vent hole location and using a carton knife trim out the carpet around the holes. The vent is wedged into the hole without screws (**Photo 10** and **Photo 11**).

Wiring the Speakers

Wire the speakers as shown in the diagram (**Figure 2**). Wire the 4Ω woofers in series parallel to produce a 4Ω impedance to each channel of the amplifier. After each set is wired, use a 1.5V battery to test the proper polarity, placing the positive battery pole on the positive lead of the speaker groups.

Use an ohmmeter to measure DC resistance of the left and the right speaker cables. They should measure about 3.9Ω. The woofers are wired in parallel to achieve 4Ω and are installed after the carpet covering is placed on the enclosure. Heat shrink tubing was used on the wired group speaker connections (**Photo 12**).

Installing the Amplifier

The amplifier is mounted to the front panel using the plastic panel attached to the front center

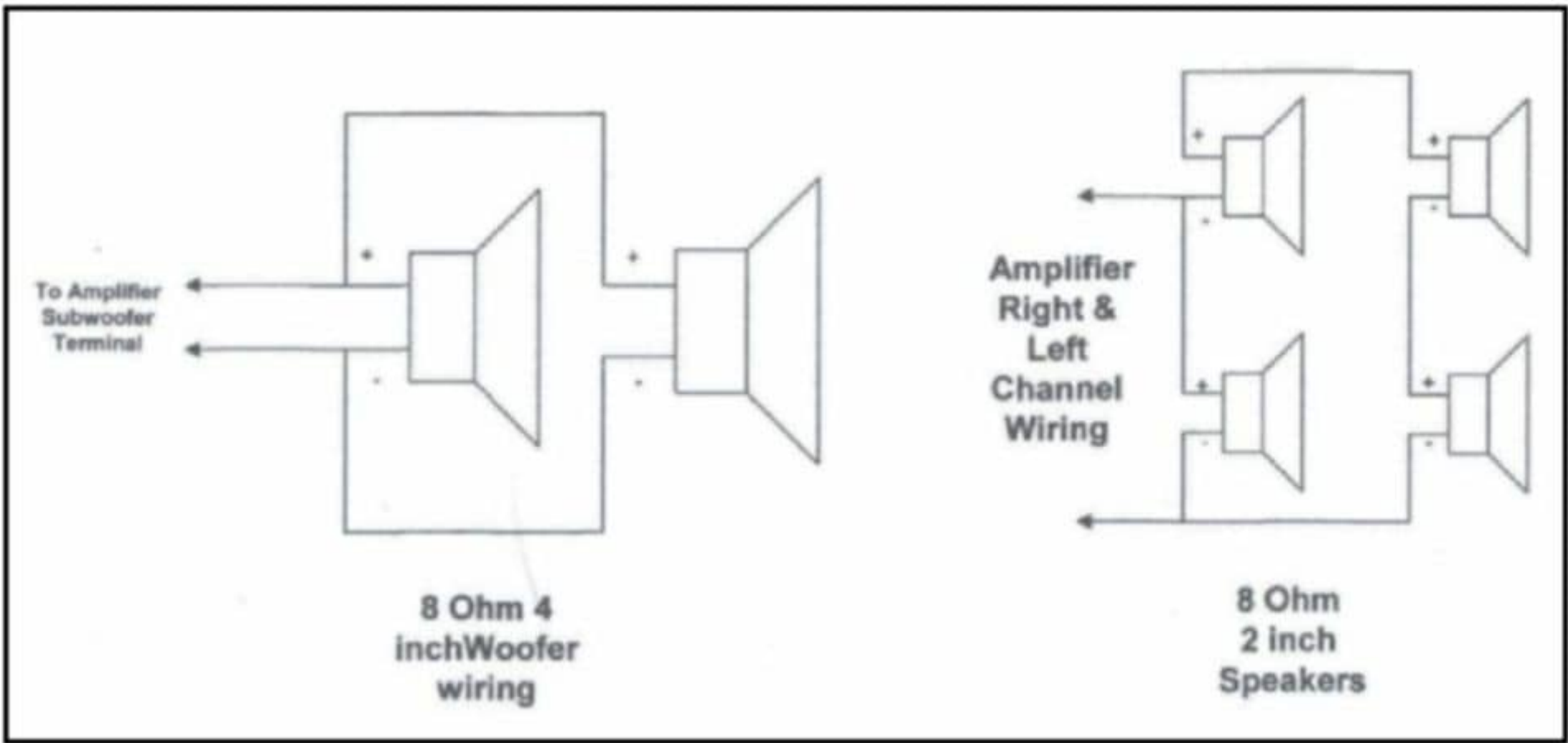


Figure 2: The diagram shows how to correctly wire the speakers.



Photo 12: The wiring of the front panel speakers. Heat shrink tubing was used on the joined speakers.

Part Lists for the Soundbase

Enclosures

- 2 each . . . . . 36" inch long 1 x 4 lumber for front and rear
- 4 each . . . . . 1 x 4 lumber 16.5" long for interior baffles
- 2 each . . . . . 1 x 4 lumber 14" long for interior baffles
- 2 each . . . . . 18" x 36" 1/2" plywood or MDF panel
- 2 pieces . . . . . 1" inside diameter PVC or cardboard pipe cut 2.75" long
- 1 each . . . . . 2" soffit vent
- One yard x 12' . Indoor-Outdoor carpet (Menards CP40N310144M Menards SKU: 7674646) or equivalent

Drivers

- 8 each . . . . . 2" speakers (Parts Express #264-1610)
- 2 each . . . . . 4" speakers (Parts Express #295-416)

Amplifier and Rear Panel Wiring Items

- 1 each . . . . . Class-D 2.1 Bluetooth 5.0 Amplifier Board 2x 50W + 100W (Parts Express #320-635)
- 1 each . . . . . Clear Acrylic Mounting plate for amplifier (Parts Express #320-634)
- 1 each . . . . . Push Button on/off switch (All Electronics Catalog #PD-13)
- 1 each . . . . . DC Power supply 12V at 5A (All Electronics Catalog #PS 12005)
- 4 each . . . . . RCA Phono Jacks Panel Mount (All Electronics Catalog #RCMJ)
- 1 each . . . . . 3.5mm Audio cable (All Electronics Catalog #CB-453)
- 1 each . . . . . 2.5mm plug and cable (All Electronics Catalog #CB243)
- 1 each . . . . . Blank electrical wall plate for rear panel interface
- 1 each . . . . . 2.5MM Coax plug with cable (All Electronics Catalog #CB -243)
- 1 each . . . . . 2.5 MM Panel Mount Coax socket (All Electronics Catalog #DCJ 25)

# Ribbon Loudspeakers

what you need to know.

Order your copy  
**cc-webshop.com**



Photo 13: Detail of the amplifier front panel shows the added on/off switch.

of the enclosure. The panel needs some modification in order to use it to hold the amplifier. The hole through which the controls pass must be enlarged to allow the threads of the controls to pass through. This can be easily done with a tapered reamer. This will allow enough of the threads on the controls so the panel can be attached using mounting holes in the panel.

The amplifier does not have an on-off switch so I installed one at the bottom of the panel. This required the drilling of a mounting hole for the switch in the center bottom area. Start the hole with a 1/8" bit and then a 1/4" bit, then use a tapered reamer to enlarge the hole to slightly over 3/8" (**Photo 13**).

The rear of the amplifier rests on a pad made of scrap pine this allows the amplifier to be removed if necessary. A piece of black construction paper was cut in the shape of the panel and mounted behind the panel and holes for the control shafts were made along with a hole for the blue pilot light to be seen. The black paper gives better contrast to the printed control labels on the panel and hides the front cutout. The amplifier is mounted into the front panel with screws in the plastic mounting panel.

## Rear Panel Wiring

The rear interface panel was made from a blank AC outlet cover drilled to hold the 12V input jack and four RCA phono jacks, which are wired in parallel (**Figure 3**). The cables for the audio input jacks were made from a mini phone plug cable cut to 25" in

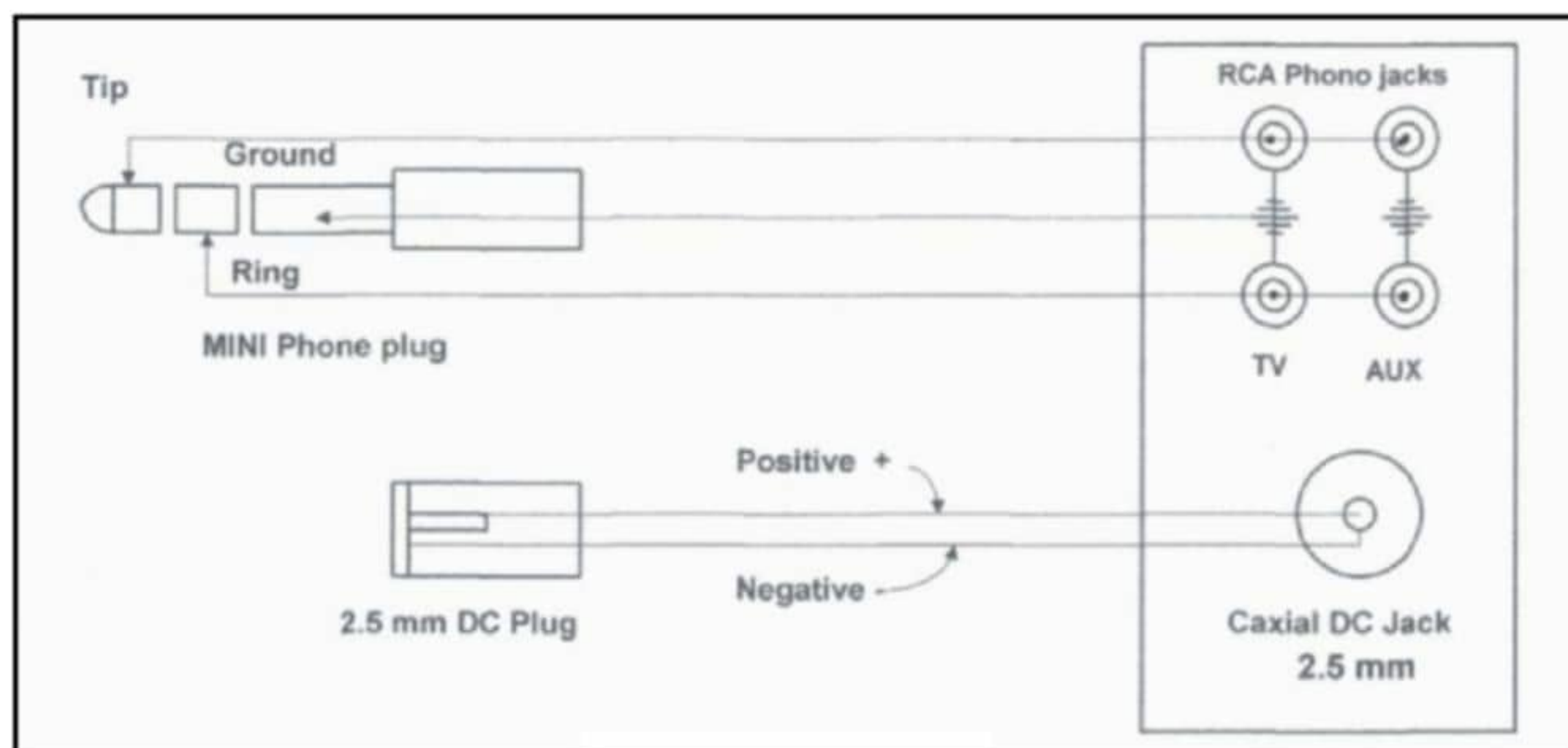


Figure 3: The rear interface panel was made from a blank AC outlet cover drilled to hold the 12V input jack and four RCA phono jacks, which are wired in parallel.

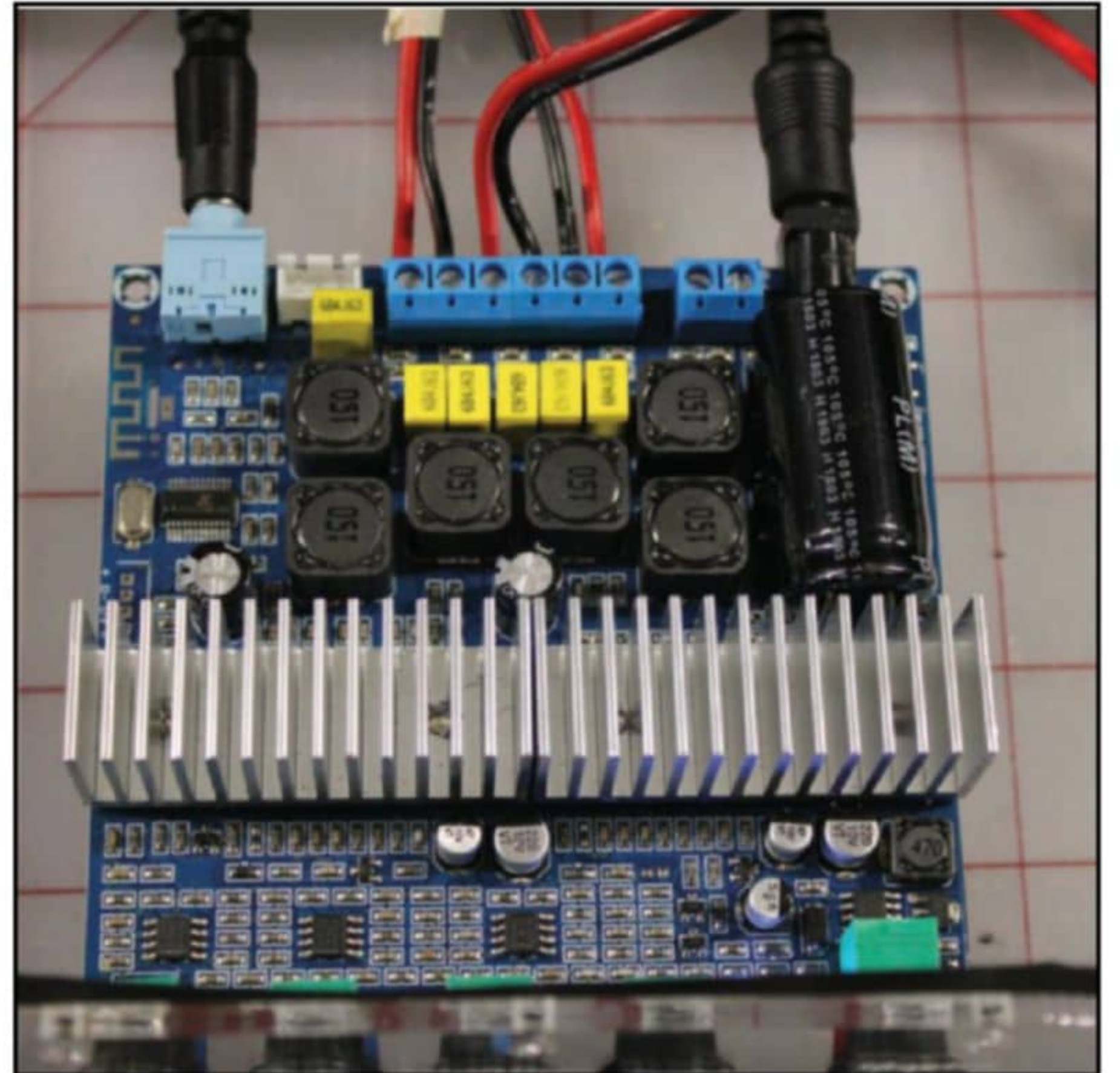


Photo 14: This is an example of the optical-to-analog audio converter that was used with a newer 65" TV.

length and soldered to the RCA jacks. The 12V power input cable was made from a 2.5mm cable with a molded 2.5 mm plug on one end. The wiring is fed through the center space of the enclosure and connected to the amplifier (**Photo 14**).

## Testing the System

Turning on the amplifier you should hear a voice say "Line in" indicating it is live and ready for a signal. Turn on a connected audio device, TV, or CD player and check that the sound is coming from the right and the left front speakers and the two woofers. If using a TV, turn down the built-in speakers and adjust the controls as required. You will find that speech from the TV is much more defined than from the built-in speakers and can be equalized better using the midrange control of the amplifier.

If you have a smartphone with music on it, sync its Bluetooth channel to the amplifier, once connected the voice will say "Bluetooth" and automatically cut out the TV audio. Listen to your favorite test track, adjusting the volume and tone controls as needed. You won't get chest thumping bass from this system, but it will definitely be well balanced and with at least two more octaves lower than the TV speakers can supply.

Overall, we found the Soundbase to be a welcome addition to our TV viewing. We have also found it to be great to use for casual music listening.

# The *Loudspeaker Industry Sourcebook 2023* Is Taking Shape...

## Act Now!

Don't miss an opportunity to be a part of this comprehensive collection of listings on loudspeaker materials! With 16 sections, 200 separate categories, and more than 600 companies from around the globe, the *Loudspeaker Industry Sourcebook* is an indispensable resource for the audio industry OEMs, specifiers, and buyers.

## Boost Your Business!!

The *Loudspeaker Industry Sourcebook* is designed to help engineers, designers, purchasing managers, and key executives locate the resources that are essential to loudspeaker manufacturing operations.

Print edition • Online PDF edition • Online database

Secure your premium advertising and **expanded vendor listing** for 2023.

Also, be sure to fill out a **FREE text-only listing** at [www.loudspeakerindustrysourcebook.com](http://www.loudspeakerindustrysourcebook.com)

For Advertising Opportunities contact:

• Strategic Media Marketing, Inc.

• P: 978.281.7708 | Email: [LIS@smmarketing.us](mailto:LIS@smmarketing.us)



# Power Transformer Parameters, Selection, and Testing

## Part 6 — Quality, Reliability, and Design

The initial group of articles detailed the history of transformer cores, their construction methods, materials, testing methods, and their losses. Part 6 focuses on commercial AC power quality, electrical component reliability, and the design of current transformers.

By  
**Chuck Hansen**

Commercial airline electric power and EMI limits, as well as environmental requirements, are specified in the American Radio Technical Commission for Aeronautics (RTCA) Spec DO-160G.

Conversely, anyone can plug a nonlinear electrical/electronic device into their 120V AC or 240V AC household power receptacles, and cause a harmonic disturbance or electromagnetic interference (EMI) back at the point of common contact (PCC). Many of us have heard the annoying buzz in an AM radio when an AC-to-DC charging device is plugged into a nearby outlet. Even some DC-to-AC converters for solar panels can contribute to excessive EMI.

The North American steady-state frequency limit for 60Hz grid-connected generators is  $\pm 2\%$ , or 58.8Hz to 61.2Hz. The system frequency is normally held to much a tighter tolerance.

Commission Regulation (EU) 2016/631 established a network code of requirements for grid connection of generators. The European National Grid is compelled to maintain the frequency of the supply to  $\pm 1\%$  of the nominal frequency of 50Hz, or 49.5Hz to 50.5Hz.

International Electrotechnical Commission standard IEC 34-1 requires industrial generators to be designed to operate within a minimum ratio between voltage and frequency (called the volts-

per-hertz ratio). Our AC backup generator follows this standard during startup and overload recovery. Otherwise, the generator could be motorized by the utility electric power.

In order to maintain the tight frequency control required for utility system stability, it is not unusual in times of high demand to temporarily reduce the grid voltage by as much as 25% (a “brown-out”) to prevent excessive load, such as during unusually hot weather conditions or the sudden loss of a large generator, from dragging down the frequency.

The grid undervoltage protection limits are designed to accommodate such a reduction. If excess load causes the frequency to get too low, the system may become unstable since, when there is not a lot of spinning reserve power available, the electrical grid inertia is reduced. In that case blocks of load would be shed (“black-out”) to maintain stability and continue to supply power to critical customers such as hospitals and other vital industries. These rolling blackouts have become increasingly common on the West Coast.

### Mechanical Losses

Ferromagnetic materials have mechanical losses associated with small dimensional changes in the core material with the applied AC voltage. This causes an audible hum or buzz at twice

the line frequency, 120Hz for a 60Hz power line, or something similar. This hum is due to the magnetostriction of the core material. It absorbs energy and results in heat that adds to the core loss and increases temperature, which then increases winding resistance. The silicon in 3% silicon steel helps to reduce this magnetostriction.

Magnetic fields generated between each turn of the windings produce side forces on the copper wire, which are especially high during secondary winding overload or short-circuit conditions.

## Reliability

Reliability is inversely related to component temperature. MIL-HDBK-217, Reliability Prediction of Electronic Equipment, is the military standard that provides failure rate data for many military-grade and some commercial electronic components. The accepted rule established by MIL-217 is that for every 10°C increase in temperature above 25°C, electronic component life is reduced by half (and the failure rate doubles).

There is a similar spec that AT&T/Bellcore developed for telephone system equipment called Telcordia TR-332, Reliability Prediction Procedure for Electronic Equipment. The Reliability Information Analysis Center (RIAC) and Quanterion Solutions, Inc. compiled actual failure rate data from government and non-government technical reports and from material supplied by various electrical component manufacturers. This was published in a report called *Electronic Parts Reliability Data 2014*.

Transformers get warmer during operation. That heat can affect the temperature of any electronic components mounted nearby. For that reason, sensitive electronic components should not be mounted near power transformers.

Aluminum electrolytic capacitors are especially sensitive to heat, and they have the lowest mean time between failure (MTBF) of all passive parts by a factor of 10 to begin with. So, it is really important to keep them as far as possible from power transformers, heatsinks, and other heat-producing areas of an enclosure. Higher ripple current also increases internal heating.

Aluminum capacitors also have a limited shelf life caused by the slow dissolution of the aluminum oxide  $\text{Al}_2\text{O}_3$  anode film over time. The oxide film is created during manufacturing by applying a “forming voltage” above the rated voltage that determines the film thickness. The film can be restored by reforming the capacitor in the same way it was formed, by applying a low DC voltage and slowly increasing it up to rated voltage while monitoring the charge current. Each manufacturer

defines its own capacitor storage life based on storage temperature and rated capacitor voltage.

Aluminum capacitors rated for 40°C to +85°C have a different electrolyte than that of 40°C to +105° capacitors, and I avoid them and use the higher temperature parts to maximize operating life. The equivalent series resistance (ESR), and internal power dissipation with ripple current, is directly a function of the type of electrolyte. I also apply a peak voltage derating factor of at least 0.85, and 0.85 derating for the maximum ripple voltage.

## Transformer Load Ratings

Ed Simon, in his July 2020 *audioXpress* article, suggested a rule of thumb for transformer VA rating of 1.2 times secondary full-load VA for 2-diode center-tap rectification and 1.8 times for a full-wave bridge. There is also excellent practical information available in the *Reference Data for Radio Engineers* in Chapter 14 for calculating average DC output current and voltage, average and RMS secondary output current and DC watts, and line power factor [5]. They are readily available on eBay and Amazon, and online used book stores. The rectifier configurations range from single-phase half-wave to six-phase star/three-phase diametric. I also recommend *The AARL Handbook for Radio Amateurs* for useful information.

The secondary output voltage will drop for resistive and inductive loads, but could increase for capacitive loads, so the load power factor influences the voltage regulation factor ( $\epsilon$ ), as mentioned earlier.

Aerospace transformer primary VA ratings depend on the design rated secondary VA load regulation factor, as well as maximum operating temperature environment, maximum altitude (less cooling air is available at higher altitudes), the number of windings, and the normal voltage and frequency tolerance of the AC power source. The regulation factor for aerospace transformers is that which best meets SWaP-c requirements. The winding temperature rise above ambient is determined by a MIL-PRF-27 winding resistance measurement procedure.

For 400Hz applications the continuous steady-state AC voltage can vary from 112.5 to 117.5 at the load center point of regulation (POR, analogous to your home utility service entrance breaker panel). In addition, there are 200% load-on, load-off voltage transients you will rarely see from your home utility power, from 84V AC (sag) for 3 to 5 seconds, to 170V AC (spike or surge) for up to 80msec.

The frequency can vary from 380Hz to 420Hz, with a special low limit of 300Hz for helicopter

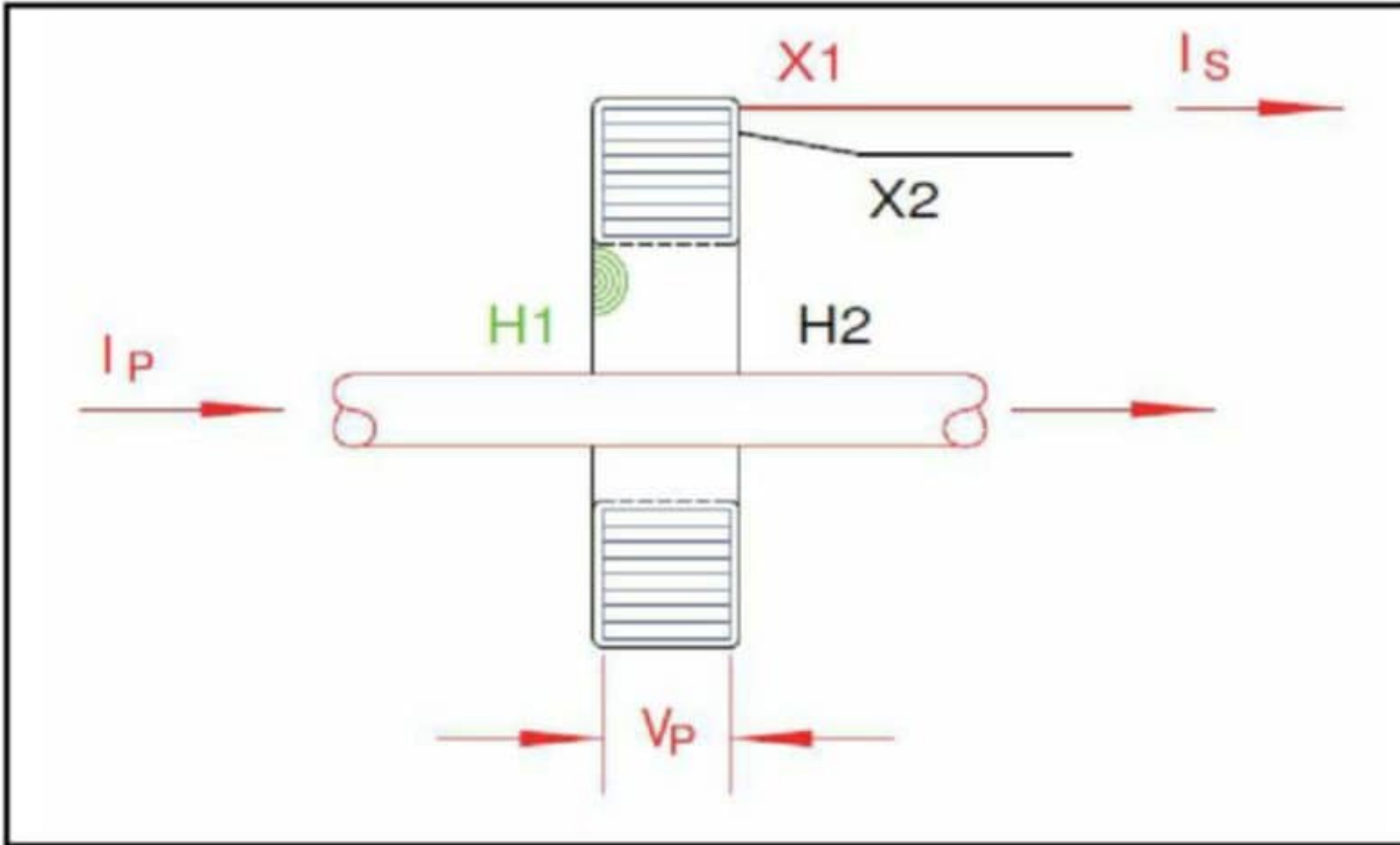


Figure 15: Current transformer (CT) primary conductor cutaway view.

autorotation mode. This mode requires an increase in magnetic core area of 33%, as compared with the constant 400Hz delivered by turboprops, auxiliary power unit (APU) gas turbines, and generators mounted on hydraulic-mechanical constant-speed drives (CSD).

Generators attached to modern high-bypass turbofan engine gearboxes can have input speed range variations up to 250% from ground idle up to maximum takeoff thrust. They require a CSD or some type of AC-DC-AC (DC link) power converter to produce the constant 400Hz frequency.

The latest AC power requirements are for frequency operation up to 800Hz. The primary DC power requirement includes the traditional 28V DC as well as 270V DC (configured as  $\pm 135V$  DC around system neutral) for some newer programs.

For high-power three-phase AC transformers and generators, the device must be designed to produce 150% of full load current for 2 to 5 minutes (depending on whether it is self-cooled, forced air-cooled, oil conduction-cooled or oil spray-cooled) and 200% for 5 seconds. It also has to withstand an AC output short-circuit current of 225% to 300% of rated current for 5 seconds minimum, for any combination of single phase, two-phase or three-phase line-to-neutral or line-to-line short circuits. Then there is a motor start requirement of 60% of rated load at 0.4 lagging power factor

while carrying full load at 0.75 power factor. AC waveform total distortion must not exceed 2.75%.

The wide voltage, frequency, and load current variations seen in the normal and abnormal modes of aerospace electric power operation require the generator and transformer designers to use a low flux density for full load, maybe around 1.0T. This has the advantage of working in the flux density range most of the time where permeability is higher, as mentioned above, but it also makes the magnetic core heavier than it could be for a design with lower peak power limits.

Smaller transformers are protected with thermal or electronic circuit breakers and must be designed based on the inverse time-current curve for the protective device. Circuit breakers for critical flight avionics and equipment are always located near the flight crew, for easy resetting. There is always sufficient redundancy for this equipment that operational rules usually dictate that a tripped circuit breaker can only be reset one time, to avoid overheating the aircraft wiring by repeated resets. This is a good practice for the homeowner as well.

## Current Transformers

Current transformers (CTs) are used for control, instrumentation and protection circuits in electric power systems. Miniature versions provide fast tripping for ground fault current interrupter (GFCI) and arc fault current interrupter (AFCI) outlets in your home.

The primary of a conventional voltage transformer is a multi-turn coil wound on the core of the transformer. By contrast the primary of a CT consists of the bus bar or power source feeder passing through the center of the CT. It does not have to be wound around the core. The primary current,  $I_P$ , enters on the H1 side of the CT (**Figure 15**) and leaves on the H2 side. We add a green dot to identify the H1 side of our CTs after a polarity verification test.

The primary voltage drop ( $V_P$ ) is limited to the section of the primary conductor directly adjacent to the CT magnetic core. You don't have to make a full turn around the core, the straight wire is the first turn. The primary impedance is only a few milliohms and microhenries, and is considered negligible. The primary current,  $I_P$  generates a current in the secondary  $I_S$  that is inversely proportional to the CT turns ratio. If the primary-to-secondary

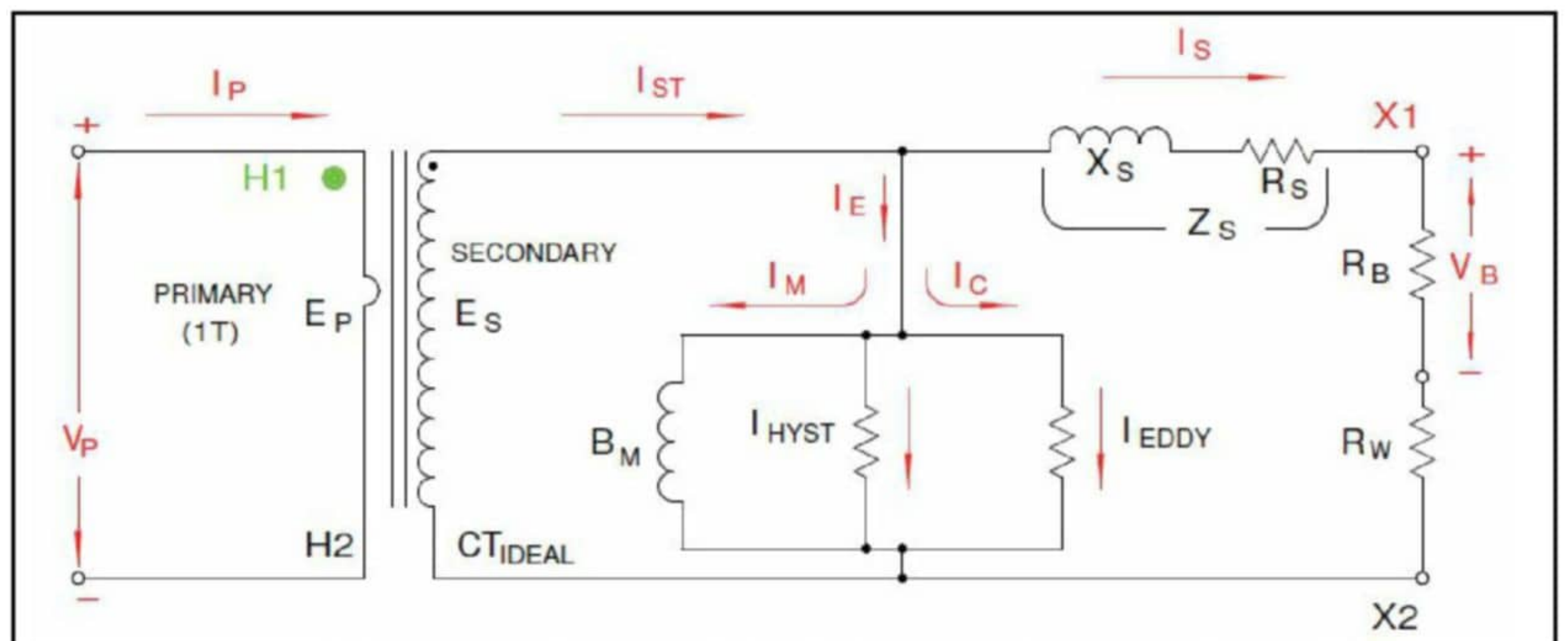


Figure 16: Equivalent circuit for a current transformer.

current ratio is 1000:1, 1000A through the primary aperture will produce 1A at the secondary winding, 250A would produce 250mA at the secondary, etc. There is a lower limit to this, once the CT becomes nonlinear at very low primary currents.

However, like any real transformer, there are exciting current,  $I_E$ , and secondary winding impedance,  $Z_S$ , losses involved (**Figure 16**).

The total secondary current,  $I_{ST}$ , is split into exciting current  $I_E$ , and the secondary current  $I_S$  that exits the CT at the X1 side of the winding (we use a color-coded Teflon lead wire for X1), and returns to the CT through X2, (a black Teflon wire). A burden resistor,  $R_B$ , is the load across the secondary, along with the series resistance of the external interconnecting wiring,  $R_W$ . The current through and/or the voltage drop across the burden,  $V_B$ , is the signal of interest in the application in which the CT is used.

This is the simplified example. For sufficiently long runs of secondary wiring,  $R_B$  and  $R_W$  may have reactance components as well. In aerospace applications, the range of burden impedances specified in MIL procedures and customer specs varies from  $1\Omega$ , to  $1.48 + j2.09$  ohms, to  $13.5 + j7.33$  ohms. The latter two impedances are part of a CT secondary winding Ratio Test and Trim Stand (**Figure 17**), and may represent a protective relay, an analog ammeter, or some other complex impedance that represents the actual equipment connected to the CT's secondary circuit. Our test stand current source is variable up to 250A, at low voltage

## About the Author

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

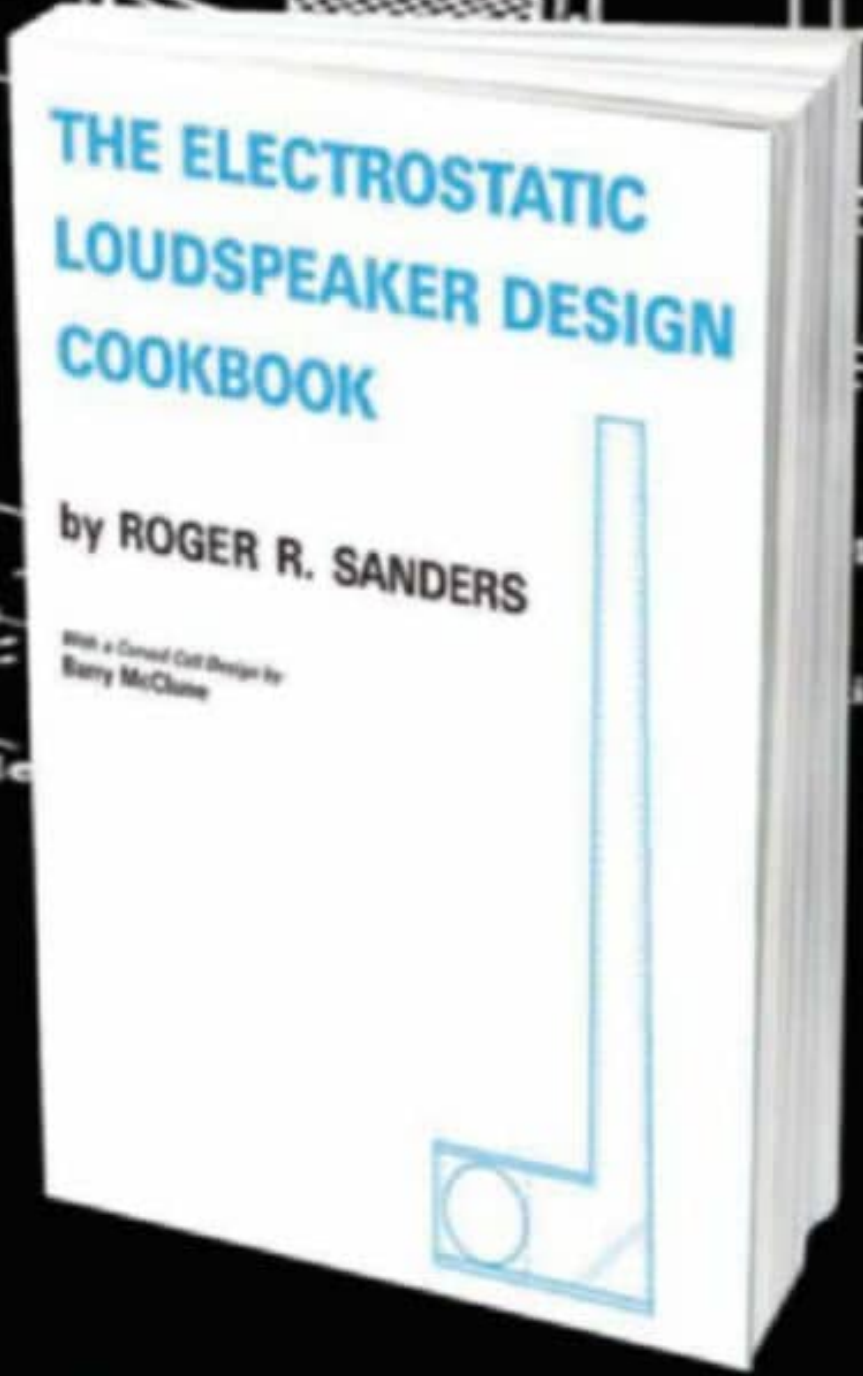


so that a large power transformer is not required to generate the primary current.

We make CTs with standard ratios of 250:2 amps (125:1 turns ratio), 500:1 and 1000:1. The 2A in 250:2 refers to the full-load secondary current. Utility CT ratios are referred to a secondary current of 5A. All current transformers have to be trimmed to their desired turns ratio, with fewer turns producing enough additional secondary current to support the exciting current, which ensures the burden current is equal to the ratio. For instance, for toroidal cores we wind 499 secondary turns for a 500:1 ratio and 998 turns for a 1000:1 ratio.

Then the turns need to be verified by a null test at rated primary load current, against calibration CTs of the same ratio, and trimmed by  $\pm 1$  turn for minimum error, if necessary. This is done by measuring the millivolt error across the  $0.180\Omega \pm 0.01\%$

**The Electrostatic Speaker Design Cookbook**  
is a complete guide for the novice and the experienced builder to successfully build this unusual sound reproduction device.



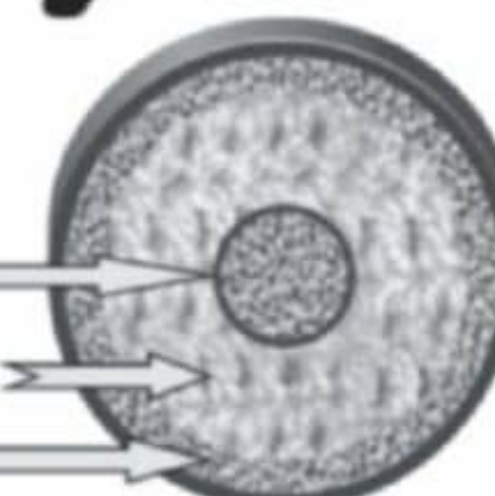
**THE ELECTROSTATIC LOUDSPEAKER DESIGN COOKBOOK**  
by ROGER R. SANDERS  
With a General Cell Design by Barry McClure

**cc-webshop.com**

**Spiral Air**<sup>tm</sup>  
Interconnects

**Coax  
Is Cool  
Again!**

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



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

**KAB**

Preserving The Sounds  
Of A Lifetime

**www.kabusa.com**

precision non-inductive null resistor, and trimming to the minimum millivolt reading. Our CTs are designed for a ratio error of 0.5% maximum, and a phase angle error of 30 minutes (half a degree) maximum. With careful winding, and an even distribution of turns on the core OD, the phase angle error can be reduced to just tens of angular minutes (60 minutes per degree).

It is very important to design the CT for the maximum fault current it will see. It needs to remain in the linear range (i.e., well below the knee) of the B-H curve, otherwise false feeder fault trips might occur if the two (or more) protection CTs don't track each other.

The turns ratio cannot be too low. Our rule is that, unless otherwise specified, protection CTs have to be at least 500:1 for sufficient accuracy. Instrumentation CTs can be as low as 100:1 if the reduced accuracy can be tolerated.

Another thing that is extremely important is to never open the CT secondary circuit. An excessively high burden impedance

will cause the CT, which is a current source, to try to generate a voltage corresponding to  $I_s \times R_B$ . This voltage will ultimately be limited by core saturation, but it could reach a high enough transient level to break down the secondary winding insulation. It will also cause a corresponding increase in the primary inductance, causing  $E_p$  to increase accordingly. Large utility CTs can generate hundreds of volts if the secondary is opened.

As non-intuitive as it may seem, it is okay to short-circuit the CT secondary winding. The secondary wire gauge is sized to support the current. Utility CTs located in switchgear have secondary shorting switches that are engaged when the CT cabinet is opened for servicing.

When I design a CT, I ask the generator or power converter engineer for a number of parameters that influence the core selection for a standard MIL-PRF-21480 AC power system. The parameters include:

- Maximum CT temperature (the highest temperature, up to 220°C, is usually for CTs installed in spray oil-cooled generators)
- Required turns ratio
- Full-load primary current
- 150% overload current and time duration
- 200% overload current and time duration (usually 5 seconds minimum)
- Maximum short-circuit current and time duration (usually 5 seconds minimum), which determines the wire gauge
- Feeder-fault protection minimum trip current and time delay
- Minimum frequency (determines the core cross-sectional area)
- Maximum frequency (determines core lamination tape thickness)
- Burden (current sensing) resistance, and any rectifier diodes in series and their type
- Any bipolar protection metal oxide varistors (MOVs) or transient voltage suppressors (TVSSs) across the secondary
- System interconnect wiring resistance between the system and the CT
- Minimum core ID, and maximum core OD and height (HT) limits in the installation

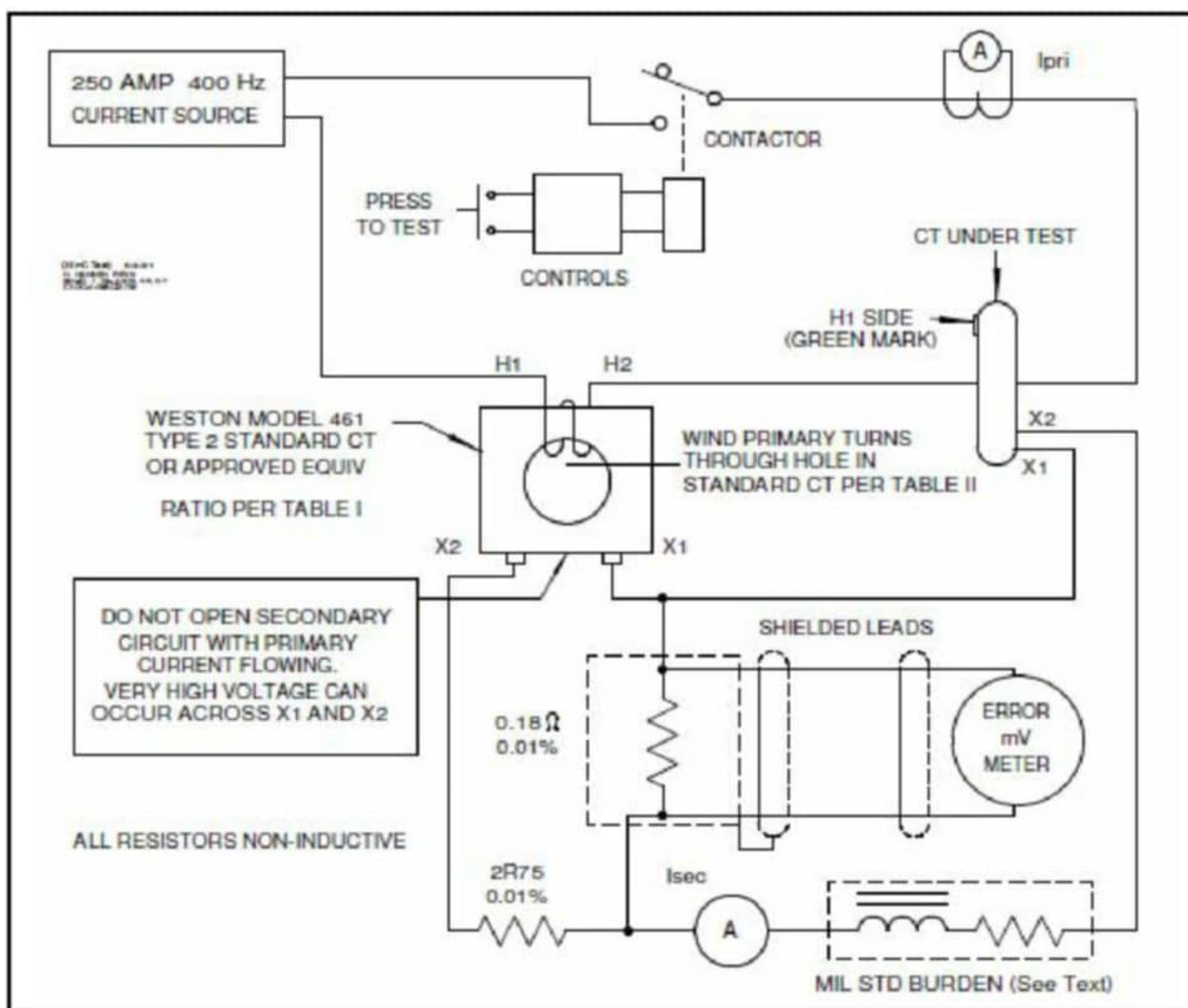


Figure 17: CT secondary winding ratio test and trim stand.

## References

[5] *Reference Data for Radio Engineers*, ITT/Howard W. Sams, Sixth Edition, 1981, Chapter 14, "Rectifier Circuit Chart."

## Resources

*The AARL Handbook for Radio Amateurs*, originally written by F. E. Handy and now in its 100th edition, <http://arrrl.org/arrrl-handbook-2023>

E. Simon, "Powering Your Circuits (Part 6): Power Transformer Rating, Selection, Fusing, and Safety," *audioXpress*, July 2020.

M. E. Van Valkenburg, *Reference Data for Radio Engineers: Radio, Electronics, Computers and Communications*, Chapter 14, 9th Edition, Elsevier, September 2001.

The CT design program will determine the optimum wire gauge, account for protective tape and epoxy encapsulation required around the coil, and generate the required core ID, OD and HT for one or more available standard cores.

## Next Month

Next month we will look at the various transformer core types, their advantages and disadvantages, and the coil winding machines used to make them into transformers.

**J SERIES**

# THE LEGENDARY J SERIES IS BACK!



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

SHIELDED AUDIO IMPEDANCE MATCHING  
DESIGN FOR A WARM SIGNAL.

SINGLE-HOLE MOUNTING ALLOWS ROTATION  
FOR MAXIMUM HUM REDUCTION.

VINTAGE CIRCUITRY, PREMIUM  
MATERIALS, HAND-CRAFTED QUALITY.

VISIT [ONLINECOMPONENTS.COM](http://ONLINECOMPONENTS.COM) TO  
ORDER OR LEARN MORE.

MADE IN THE U.S.A.



**onlinecomponents.com**

THE ONLINE DISTRIBUTOR OF ELECTRONIC COMPONENTS

# Passive and Aggressive

EPIQUE  
by Dayton Audio



## Patented MMAG Motor System

Capable of unheard-of excursion for their size, Epique drivers are engineered to produce exceedingly low distortion with controlled response. Allowing more versatility than ever, you now have the opportunity to create systems that will deliver game changing performance.

## Carbon Fiber Is All About Performance

Dayton Audio's new ground-breaking Epique E150HE-PR 5-1/2" and E180HE-PR 7" passive radiators with carbon fiber cones are the perfect companion for the Epique Series MMAG by retaining the same family look. Engineered to replace the excessive port lengths required for low-tuned enclosures, these passives give you the opportunity to create systems that provide extraordinary accuracy and fidelity without the extraordinary size.

## Epique low-distortion drivers and matching passive radiators

### Epique DVC MMAG Extended Range Subwoofers

- Patented dual gap motor design allows exceptional linearity
- Over 14 mm of excursion (at 70% BL), 28 mm peak to peak
- Damped woven carbon fiber cone delivers remarkably smooth response
- Inductance control for extremely low distortion

### Epique Carbon Fiber Cone Passive Radiators

- Carbon fiber cone is strong enough to handle the pressure of high output drivers
- Incredibly linear 19mm xmech enhances output at low tunings
- Extremely versatile, perfect compliment for any compact subwoofer

### Shop the Entire Epique Series



DAYTONAUDIO  
daytonaudio.com

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

parts-express.com/brand/  
EPIQUE-by-Dayton-Audio