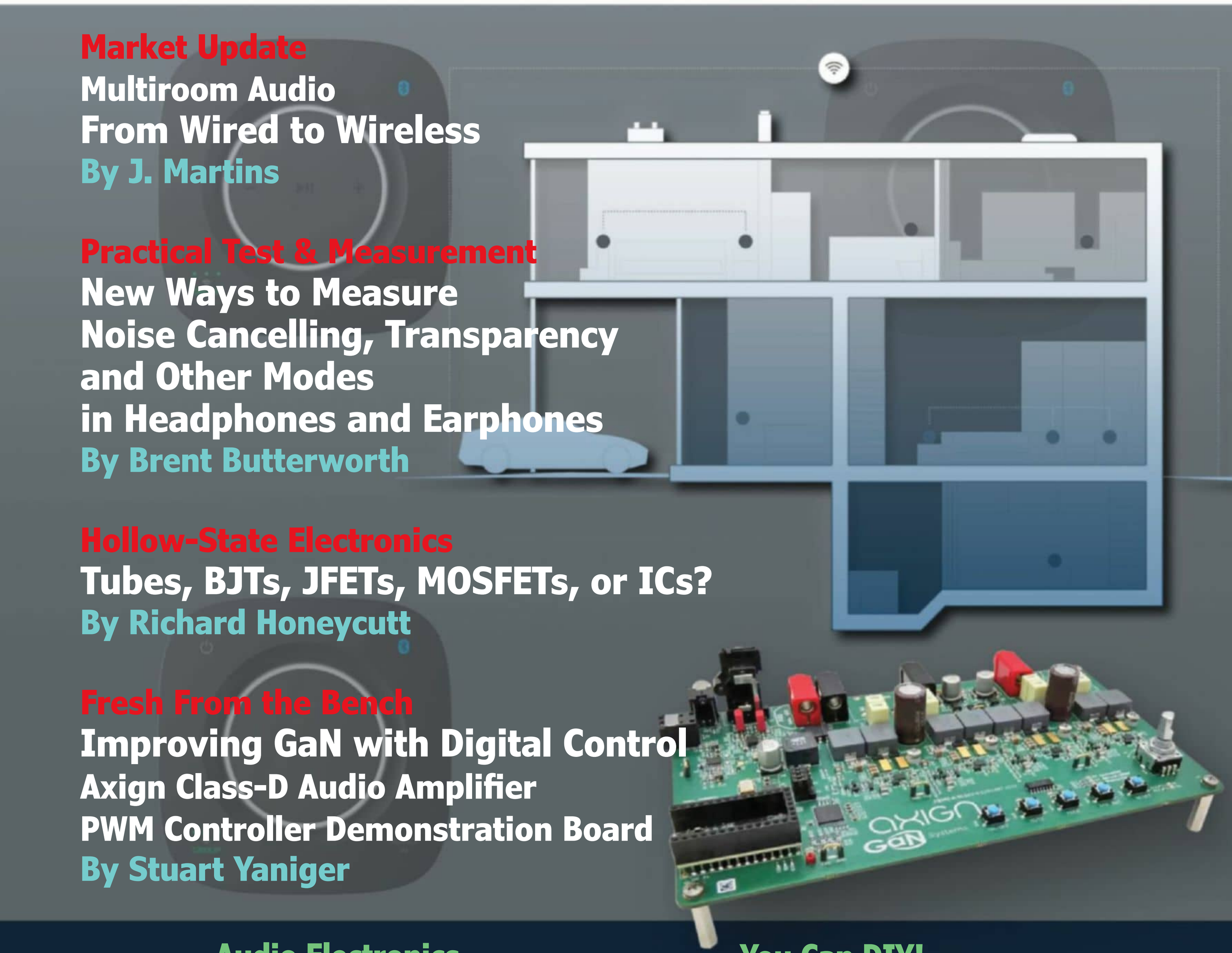
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Ten Years More of Everything

Even though my involvement with this publication started earlier, it was in November 2013 that *audioXpress* received a facelift, signaling a new life for this publication. We expanded the content and increased our focus on audio research and development (R&D), product design, and acoustic science while maintaining our roots in DIY audio and audio electronics. Over the years I have constantly re-examined those directions and more importantly worked tirelessly to expand our authors' network in line with the technology evolution and constantly evolving audio industry trends. As is always recommended on editorial projects, every year we have re-examined our mission and what was achieved, reflecting both our audience's interests and readership expansion across product and market segments.

I confess that I was surprised with the positive response we received in certain technology fields (e.g., voice, automotive audio, and DSP), while feeling frustration from the lack of contributions and response in others that we see as strategic (e.g., acoustics). As always, not everyone was positively excited with all the initial changes. Ten years later, I'm glad we stayed the course to see our original mission recognized and validated by readers, authors, and advertisers.

audioXpress continues to be committed to serving the enthusiasts and the global audio engineering community, while aiming to remain a reference for developers and designers. And its audience has expanded significantly since November of 2013, when the magazine embraced its audio development focus. Now reaching 52,000+ readers all over the world (print and digital circulation combined), while our website receives 12,000 visitors daily, on average, for the past 12 months.

As I wrote in my original editorial 10 years back: "We believe *audioXpress* will blossom into a fascinating publication that follows the latest audio innovation trends, independent of the application field, and share a common audience of engineers, consultants, and enthusiasts in the electronics and audio fields, most of whom are involved in R&D." Today, the magazine's renovated audience are engineers working in manufacturers' R&D departments, engineers involved in product design and development, product management, production, systems, integrators, and consultants.

Once again I would like to evoke how Edward T. Dell, Jr. (1923–2013) devoted his life to people with a passion for audio electronics and founded *Audio Amateur* (rebranded as *Audio Electronics* in 1996), *Glass Audio*, *Speaker Builder*, and finally *audioXpress*, which he edited until 2011. As Dell wrote in his first *audioXpress* editorial, explaining why readers, authors, and advertisers are the key ingredients for a magazine: "Readers are, collectively, a kind of community. In truth, authors are only an advanced form of reader. And I believe no avocational magazine can be healthy or successful without advertising."

The fourth "ingredient," Ed added was Passion. "My passion for good sound carried me along despite long hours of endless work, very slow growth, and even resorting to co-opting members of my young family. The task of producing a magazine is easier today as far as mechanics are concerned, but I think passion is still fundamental. I still look forward to coming to work each morning. Magazines stir the imagination, they surprise, they stimulate, they amuse. I am happy to report, somewhat immodestly, that I still enjoy that prospect myself."

Ten years past, *audioXpress* has done well in incorporating all four ingredients. Then and now, we just need more of everything we have already committed ourselves to do.

And of course I would like to recognize the essential role and support of the team that worked closely with me during these 10 years: Shannon Becker, Associate Editor; Jan Didden, Technical Editor; and KC Prescott, our publisher, and founder of KCK Media Corp. Of course, I would also like to thank Cristina Costa, our social media manager, who works hard to always be by my side, Mike Klasco and Nora Wong (always there when I need them, anywhere in the world), Vance Dickason, and all the many authors and contributors that make this publication.

Our work for 2024 has already started and we are confident that *audioXpress* is increasingly playing a role in inspiring all product developers, and Advancing the Evolution of Audio Technology.

As always audioXpress is open to contributed articles. If you feel up to the task, send an email to editor@audioxpress.com.

J. Martins
Editor-in-Chief

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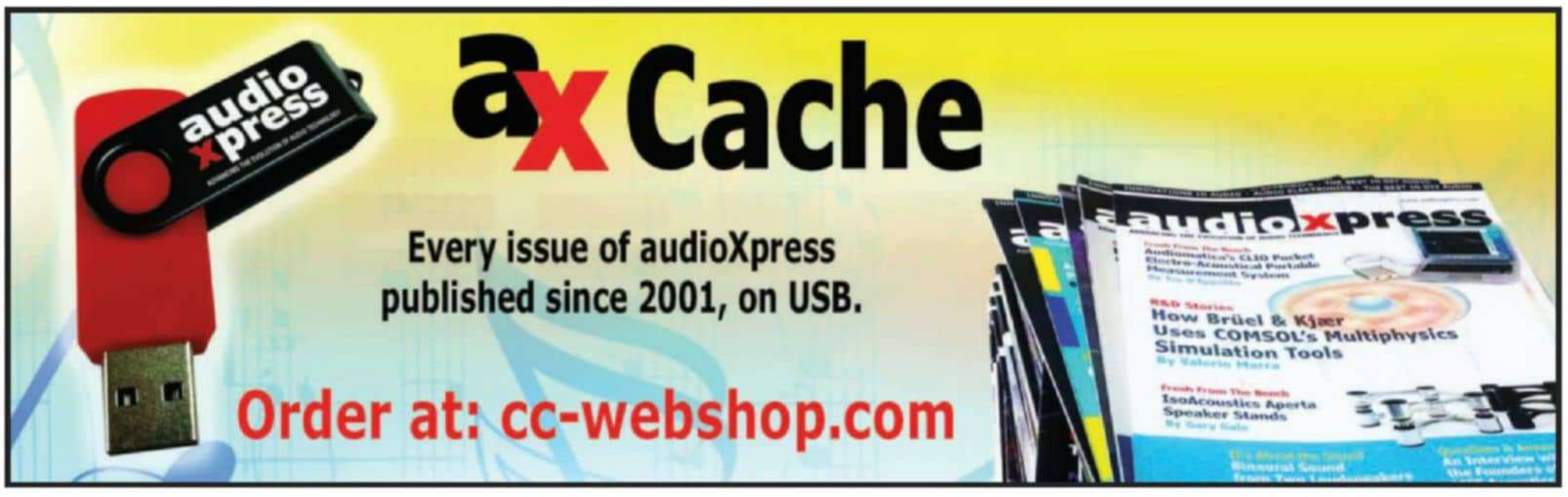
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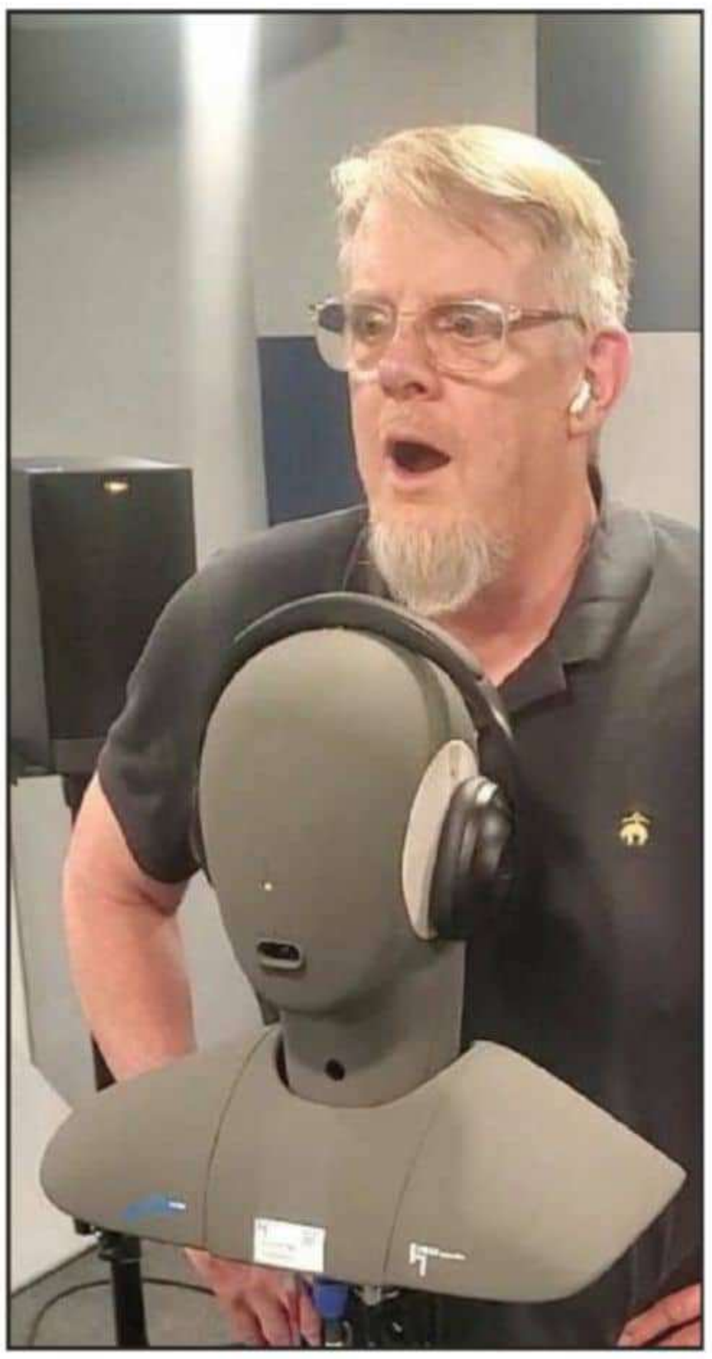


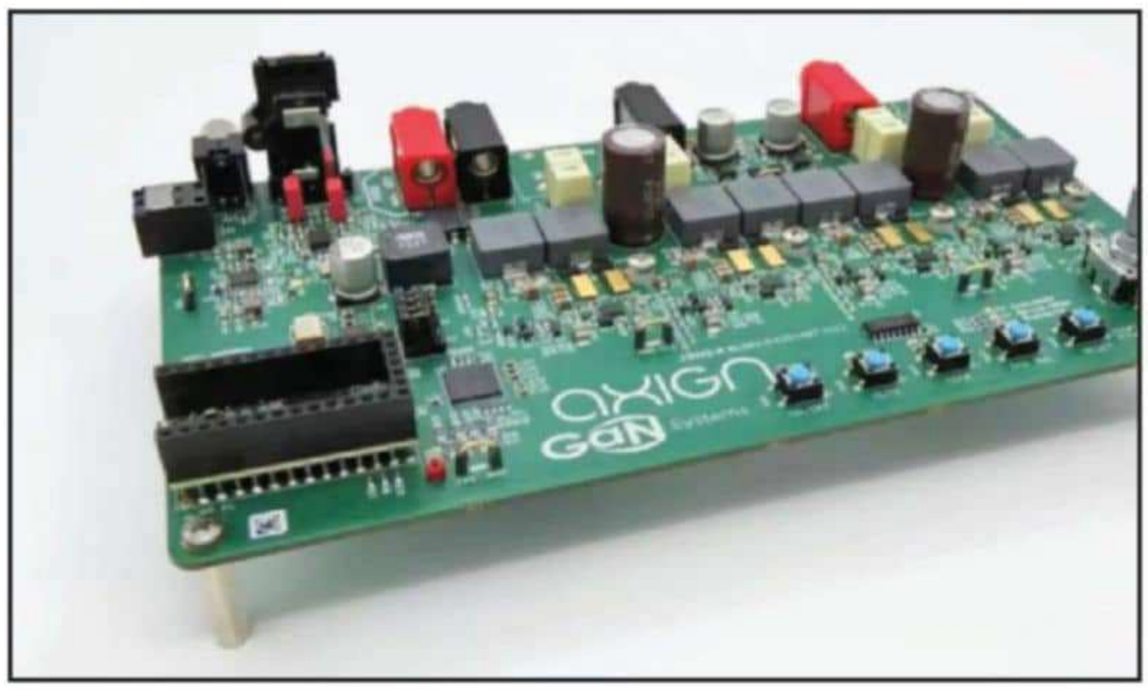


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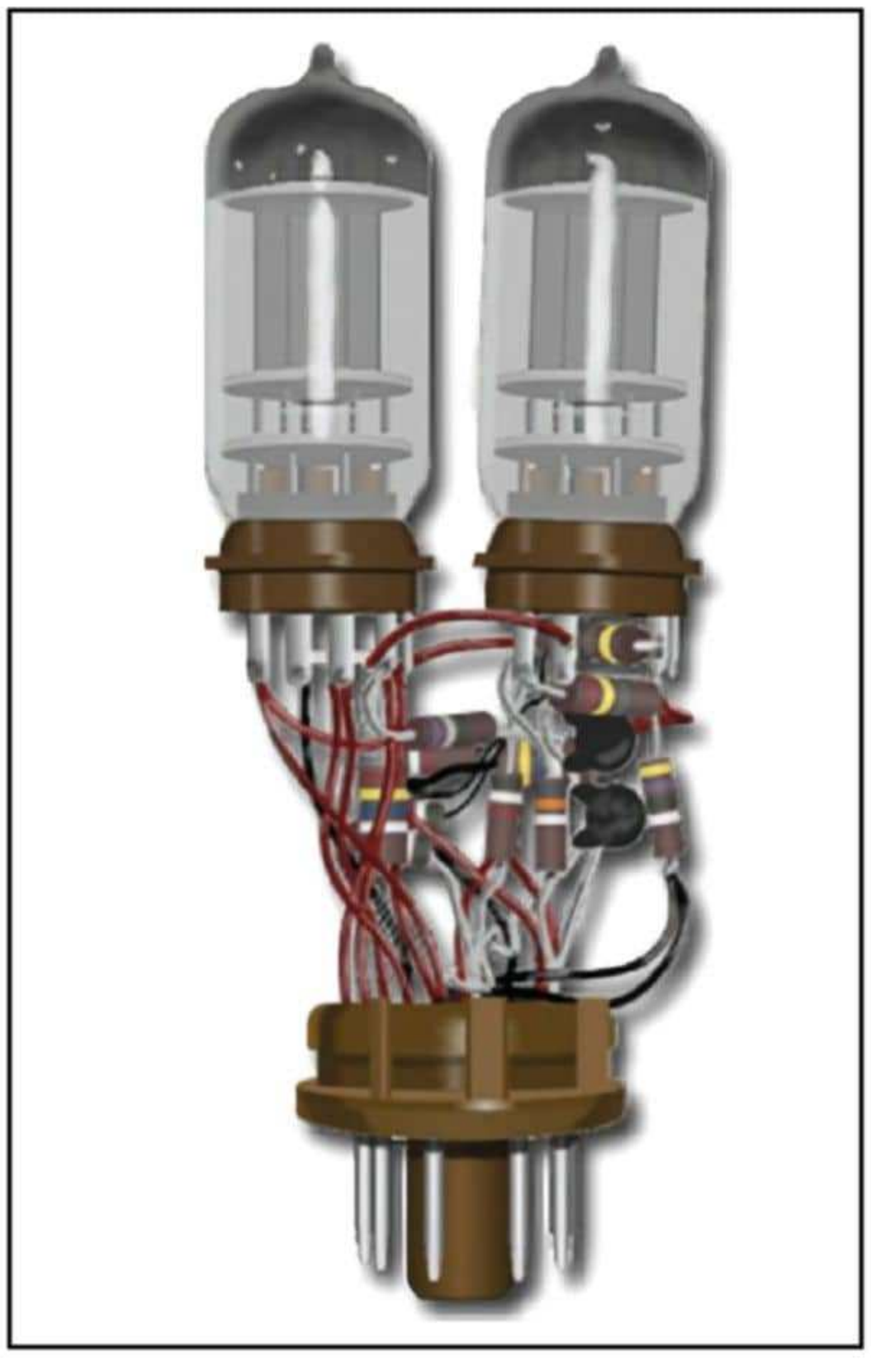
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Multiroom Audio From Wired to Wireless

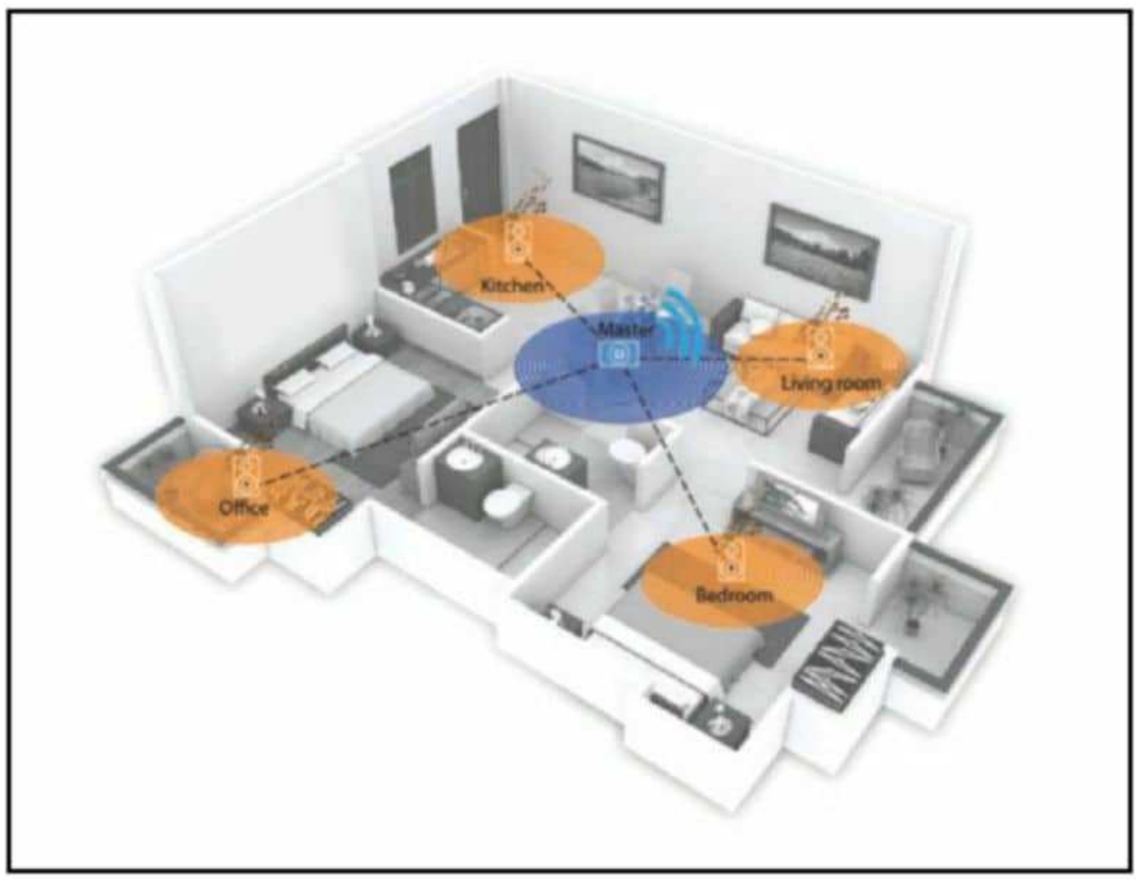


By
J. Martins

Building on top of the whole-home infrastructure, multizone and multiroom audio are exciting concepts in residential audio integration. This article traces recent developments in that area.

Wired or wireless, extending audio to the whole home is something that sounds increasingly appealing to homeowners, particularly those with the ability to benefit from different separate areas, including outdoors. The experience of recent years with consumers valuing their home spaces much more has caused a spike in demand for multiroom residential installations. Not everything is the full-featured custom-install, infrastructure-level type of projects with built-in wall and ceiling speakers—and where integrators now also tend to add value by mixing video distribution to go with audio, probably built on top of home networking, with security features and possibly a centralized control system to top it off.

Effectively, those fully featured infrastructure solutions are associated with the motivation to build smart homes, expanding and integrating all other disciplines, including window shades, lighting control, and HVAC, even before getting into the kitchen appliances universe—which is a realm of its own.



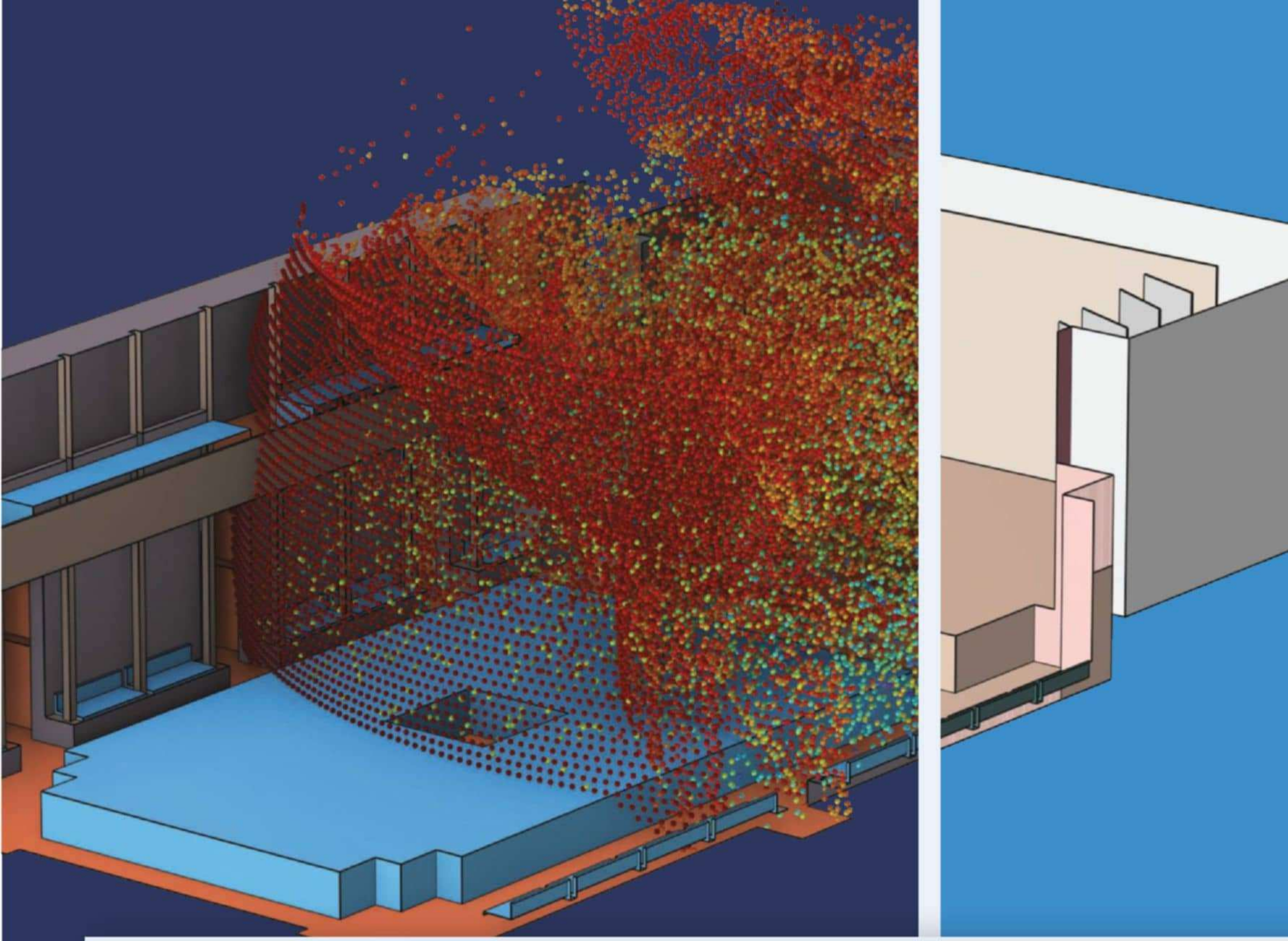
Original concepts of multiroom audio, wired and wireless, struggled with the inherent network architecture and the corresponding trade-offs between the number of connected points, range, and reliability under different network traffic loads. The centralized approach was normally the preferred option, which was good for playing the same music in every room, but wasn't ideal for having different sources playing in different rooms.

Consumers Want Control

Following the experience of a global pandemic, many homeowners have requested an upgrade to their home installs, and this includes multiroom entertainment options that can be audio/video or just audio. A very clear trend is the fact that consumers—even when calling for integration services—want to be able to fully control their systems and have the option to build on their investments, being able to add their own future gear acquisitions, and upgrade existing systems (e.g., buy a larger, better Smart TV), without having to redo the whole installation.

That is one of the reasons why separating the audio multiroom install makes increasingly more sense, since TVs are now almost always another signal source for their own dedicated areas, and very rarely would one wish to see the same content in more than one room of the home, unless it is a live broadcast event (such as sports). For those cases, every TV is a tuner/receiver on its own, and there's no advantage to distributing signals from point to point inside a regular home.

For dedicated multiroom audio, there are also many reasons why the systems should be separate since this allows users to benefit from a combination of technologies and evolve as things also change. Starting with the good old wired amplifier to speaker



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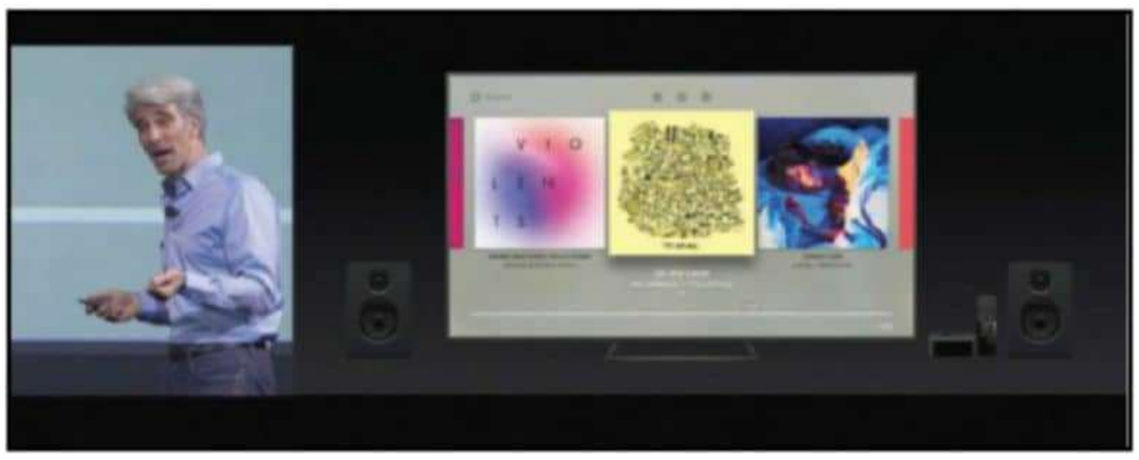
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Market Update — Residential Integration



With its AirPlay 2 update, Apple incremented support for multiroom audio, basically by reinforcing the control from Apple's Home app and multiple AirPlay-enabled devices, in order to allow configurations that go from two HomePod speakers as a Stereo Pair, and control which audio sources play on each device.



Unlike networked multiroom solutions such as Sonos or DTS Play-Fi, AirPlay 2 simply enables the control of supported devices in multiple rooms with full app integration. An Apple TV can stream music to a HomePod placed anywhere in the house over Wi-Fi, with the whole experience controlled from the TV interface or any iPhone. This also works with any speakers that support AirPlay 2.

line level systems—which are still great for having up to three zones in the home, including possibly an outside area.

For larger homes, particularly those that extend over a larger area, the second wired option is naturally to build a networked system and benefit from the extended range of products that now support Dante and AES67.

And finally, the wireless audio option, which in some cases can be there simply for convenience. As more homes adopt portable speakers that can be moved around, those speakers also can be integrated into the home network via built-in Bluetooth and Wi-Fi capabilities, and can be used as part of multiroom systems. Homeowners want their systems to be easily moved around, expanded, or updated, and they want to be able to control the systems they add, together with their subscribed services and existing whole-home audio systems, when there's one in place.

Increasingly, wireless audio solutions are also being adopted because they are really the only economical option in existing homes. Even residential integrators now prefer to embrace wireless audio systems than risk budgeting an overly ambitious whole-home wired infrastructure that will be out of reach for their customers.

The Wireless Option

For reasons that we have extensively explored in our latest residential integration Market Update, multiroom audio continues to be a discipline that is limited in the number of players in the market segment. Intellectual property in this domain was the product of extensive development and research at an early stage and remains highly protected, for good reason. As Sonos continues to successfully validate its pioneering efforts in court—including against mammoth corporations such as Google, which would gladly extend its Google Cast protocol to the whole-home—every other company in this space pays royalties to the rightful owners, and that includes mighty Apple.

In fact, Apple and its AirPlay wireless communication protocol and stack/suite is the second most common name cited when discussing multiroom audio options from manufacturers and the residential integration industry. And yet, AirPlay in itself doesn't support multiroom audio applications. AirPlay, even in the latest updated AirPlay 2 iteration is still very much a point-to-point technology. But AirPlay integrates well with existing whole-home and multiroom platforms.

AirPlay 2 basically extended the control capabilities for home audio systems and multiroom speakers throughout the entire house, but doesn't change the audio transport mechanism, which now also supports multichannel and object-based formats. Most manufacturers want to support AirPlay 2 because they see the demand, and it allows them to benefit from the compatible audio sources that can be streamed through their products—with iPhones and iPads being the most common streaming source and control interface in most residential audio installations. Bose, Devialet, Bang & Olufsen, Denon, Marantz, Bluesound, Polk Audio, Naim, Libratone, Linkplay/Wiix, and many other Wi-Fi-enabled speakers support AirPlay 2 as a core feature. All the residential integration brands also support it mainly to connect media sources to end destinations, while their own systems do the signal distribution throughout the home.

But things have evolved recently in this domain, as we are seeing a significant evolution in Wi-Fi home networks, with homes benefiting from faster Internet connections and services, the fast adoption of 6/6E home routers, and the perspective of a significant technology jump with upcoming Wi-Fi 7, also expanding the available wireless spectrum and creating interference free bands for audio/media applications.

audioXpress continues to follow this area closely with regular articles in its weekly newsletter covering related topics.

Wireless Audio Streaming from TVs

While sending a stereo analog audio source from one amplifier to two pairs of speakers is relatively simple to do—one speaker covering the dining area and another the kitchen, as an example—streaming digital audio wirelessly to two different sets of speakers is a whole different challenge.

We all have memories of once experimenting with turning on two radios (or two TVs) with the same programming in different rooms and realize how everything sounds "echoey" and strange. This is because the two sources, not being isolated, make sounds

to arrive timed differently as they are being combined in our ears—adding comb filtering and modulation effects that the acoustic environment can also cause. And this is from the same source. When adding inherent delays and frequency responses to those sources, such as when the received programs are broadcasted from two different channels, the whole experience becomes unbearable.

When trying to do the same using streamed sources over wireless protocols, the added latency can emulate those undesired acoustics effects. In fact, the same might happen even in wired networks. The way to avoid this is to synchronize the timing of the signals precisely—and that was one focus of the work of earlier multiroom protocol pioneers, who patented the solutions just to enable basic left/right distribution.

Even today, we are still struggling to do wireless stereo properly to two separate devices, as anyone who has tried to pair two wireless speakers over Bluetooth has found out. Latency in wireless audio can be tricky without a tight synchronization mechanism for left/right signals. And we can only imagine how it is to send 5.1 channels of surround sound, or other multichannel formats such as Dolby Atmos!

In fact, much of the excitement around the latest Bluetooth LE Audio and Auracast specifications result from the fact that those updates promise a platform that will be able to do stereo streaming to separate left and right receiving devices (true wireless stereo) and do broadcast—one source to many receivers (in mono only).

Basically, this is the same functionality that became available many years ago in the SKAA Wireless Audio proprietary technology developed by Canadian firm Eleven Engineering. SKAA is today available in numerous products, which receive audio via Wi-Fi or Bluetooth and then transmit using SKAA to multiple receivers.

To do wireless multichannel—accurately reproducing existing surround, immersive, and spatial audio formats—is a completely different set of challenges.

Many other problems are involved, depending on the source of format decoding and streaming, and overall control of the experience. And in general, having to accommodate the fact that signal distribution in the home normally sits on top of the most common wireless home network technology, which is Wi-Fi.

Wireless Multichannel at the Core

Efforts to deliver high-quality wireless audio for multichannel home systems have been evolving for some time. One notable example is the Wireless Speaker and Audio (WiSA) Association.

WiSA Technologies, the company responsible for the technology development and licensing, managed to successfully build a wireless multichannel solution for home cinema that used the available unlicensed spectrum, avoiding being in the same exact crowded bands with Bluetooth, Wi-Fi, and everything else. This proven approach creates a dedicated wireless network working on the more reliable U-NII 5GHz spectrum, with a unique signal latency of just 5.2 milliseconds, and 1 microsecond channel synchronization.

Currently designated as WiSA HT (for Home Theater), that wireless audio technology is supported by many manufacturers, including Bang & Olufsen. WiSA HT allows expansion to immersive configurations by supporting up to 12 speakers connected simultaneously (adding four height speakers to the standard surround 7.1 configurations) but remains confined to a single room.

In that same home theater segment, the market is changing fast, with smart TVs replacing set-top-boxes, and becoming the primary source of access to streaming services. At the same time, the consumer electronics market demands more affordable approaches for convenient wireless expansions of TV sound and soundbars, creating a formidable incentive to push wireless audio technologies as a mainstream solution. Those thinner LED and OLED TVs have created the need for soundbars, and consumers now associate even the most basic cinematic sound experience from a TV source as being Dolby Atmos, immersive formats and "spatial audio," which challenges what defines the home theater experience.

This is what is motivating WiSA to change strategies and now attempt to build a solution for wireless home theater that can be more easily implemented, directly supporting the increasingly sophisticated SoCs in smart TVs and soundbars. In the simplest use case, supporting two surround satellite speakers that extend the integrated left/center/right plus two up-firing channels of a soundbar, and which might even include the need to recognize the built-in TV speakers as one or two channels.

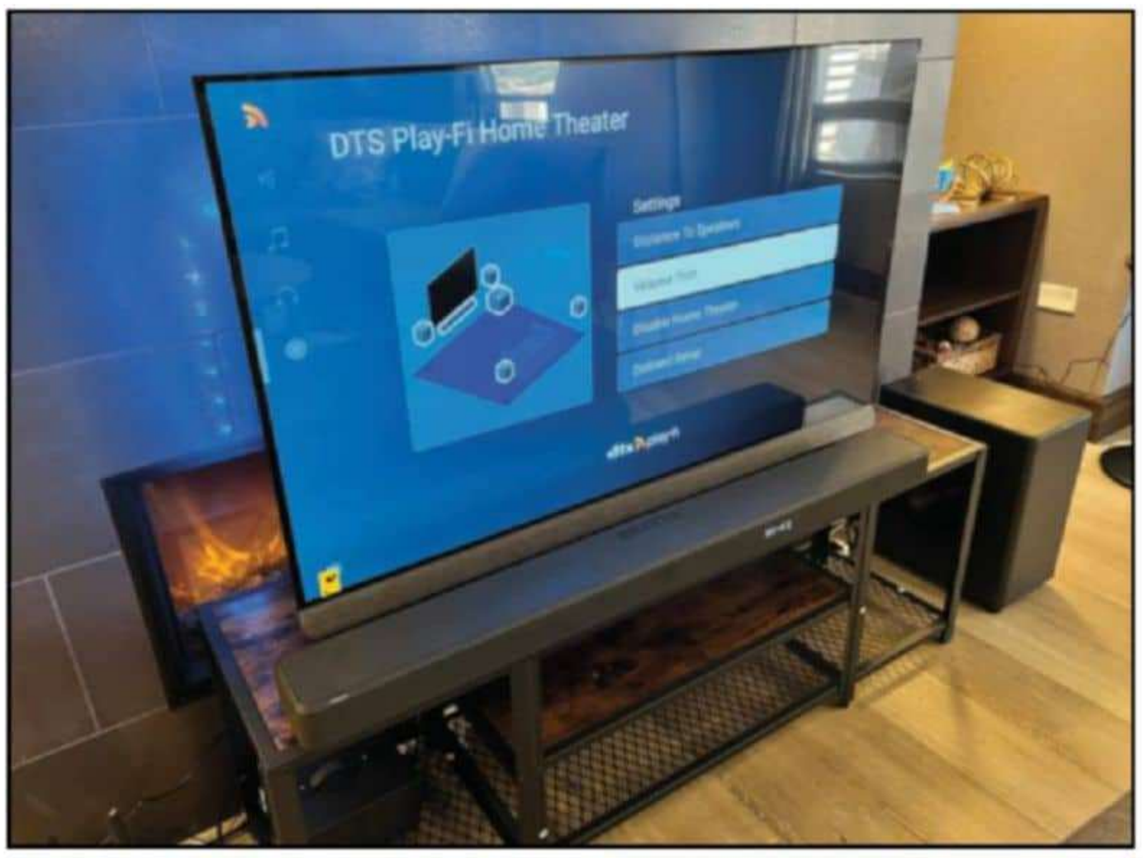
This was the reason why WiSA developed the WiSA Discrete System (DS), a module-based technology that simply transmits up to five channels—four full-range audio channels and one high-fidelity subwoofer channel—of uncompressed 16-bit/48kHz sound over a 2.4GHz Wi-Fi network it creates. This is a simpler implementation that is ideal for soundbar systems. When the soundbar does the core channels connected to the TV or STB source (e.g., HDMI), the five wireless channels become available to supplement the soundbar speakers with immersive channels.

A third approach that WiSA is aggressively pursuing is WiSA E (for Embedded). WiSA E can be implemented both as a plug-and-play module or as an IP/software license, enabling brands to embed WiSA E software directly into compatible SoC TV platforms. Using the 5GHz portion of the Wi-Fi band, WiSA E offers high-performance, high-quality wireless audio transmission and reception at an affordable price. It supports multichannel audio configurations transmitting uncompressed 24-bit/48kHz audio with a fixed latency of 20 milliseconds and tight channel synchronization.



WiSA and Platin Audio launched the Milan 5.1.4 Soundbar System, the industry's first to incorporate WiSA DS Technology to wirelessly power four channels of surround and true Dolby Atmos up-firing audio and a subwoofer from a single transmitter module.

Market Update — Residential Integration



New 2023 TVs with DTS Play-Fi demonstrated with wireless soundbars and speakers. Effectively, Play-Fi is a true aggregator for home audio entertainment that also does immersive home theater. For most CE manufacturers, this is a gateway to the whole home that represents an important upselling opportunity, starting at the entry-level product categories.

From TV to the Home

LG Electronics has been a strategic partner for WiSA since it announced the adoption of WiSA (HT) across all its display product ranges in 2019. But the WiSA DS module was also integrated into other TVs, including from TCL, in a sign of growing support from TV manufacturers to this type of technology. The clear advantage is to have the highest-quality immersive audio experience possible with external speakers, all from a wireless connection, originated directly by the TV and using standard Wi-Fi—no need for external transmitters, TV dongles, and AV receivers.

This is a similar approach to what has been actively pursued by DTS with its Play-Fi technology, with the difference that DTS Play-Fi was developed from early days as a true multiroom solution and remains one of the very few pioneers in that space.

The advantage of having these wireless audio technologies "embedded" in the TV is that other companies, including speaker companies, can now focus on designing compatible wireless speakers that extend the experience for the home. By incorporating the receiving Wi-Fi modules in each speaker, they are more easily able to dispute the living room, and eventually expand built-in features to smart home integration.

And with the very fast adoption of Wi-Fi 6 and 6E into the homes—not to mention upcoming Wi-Fi 7—the possibilities will continue to expand as the experience improves with faster speeds and even lower latency.

For DTS, this suddenly opened another exciting possibility, where the TV audio source can be directly connected to the whole-home audio system. As an example, Smart TVs support most major music streaming services, and the TVs can become the player that sends music to any room of the house, even to all the rooms at the same time.

The wireless sound will work from all TV sources including wired inputs (e.g., HDMI), on-board streaming apps, and wireless streaming protocols such as AirPlay and Chromecast. And consumers will be able to mix-and-match DTS Play-Fi-compatible soundbars, speakers, subwoofers, and amplifiers, regardless of the brand—an approach that DTS has followed since the early days of the technology. Any active speaker used since 2021 for Play-Fi Home Theater can be upgraded to be part of a DTS Play-Fi Immersive Home Theater system.

Play-Fi also works across various TV operating systems and supports all leading TV system-on-chip solutions, utilizing the same Wi-Fi radio already built into any smart TV—no additional hardware modules required. Active speakers will just need to be equipped with a DTS Play-Fi module.



Play-Fi Home Theater allows consumers to create their own home theater using any number of Play-Fi products as building blocks, with up to two subwoofers added to soundbars or stereo speaker configurations, and optional satellites added for up to 7.2.4 configurations. This latest 12-channel solution will be promoted in the market in 2024 as Play-Fi Immersive Home Theater.

Re-Enter DTS Play-Fi

It's interesting to explore what DTS is doing with Play-Fi, a technology with a unique and interesting trajectory. Having built a solid foundation for multichannel wireless audio over standard Wi-Fi, DTS Play-Fi is now reaping the benefits of more than a decade of persistence and building on the original vision of multiroom audio.

The Play-Fi team has been consistently working on the technology for more than 12 years. It was with a company called Phorus that the technology behind Play-Fi was developed. Before Phorus, the original team originated from Harman/JBL. Daniel Lau, who is now General Manager, Networked Audio Division at DTS, has been leading that effort since his company was acquired by DTS, and since DTS became part of Xperi.

The Play-Fi protocol leverages the important advantages of Wi-Fi standards and sharing metadata according to the Digital Living Network Alliance (DLNA) specifications, being able to support standard 16-bit/48kHz audio quality transmission all the way to high-resolution audio without compression up to 24-bit/192kHz. Typically, Play-Fi will use 16-bit for wireless multichannel. Hi-res is supported for two channels in Critical Listening mode (where

wired is still recommended). Wired sources such as HDMI eARC, and wireless protocols such as AirPlay and Google Cast are supported.

After DTS acquired Phorus in July of 2012, it launched Play-Fi in October of that year with the release of the Phorus PS-1 speaker and PR-1 receiver, later followed by the Phorus PS5. At the same time, the company began directly offering Play-Fi technology as a licensable, open solution through a simple-to-integrate, embeddable technology.

The original patented multiroom audio streaming technology for whole-home audio is today licensed to multiple audio brands, and the Play-Fi product support line includes multichannel wireless audio modules for CE manufacturers, embedded Play-Fi software stacks, OS-level Play-Fi drivers for Android, AndroidTV, Linux, and Windows, and apps for Android, iOS, Kindle Fire, and Windows. Since 2014, DTS offers its Play-Fi Certified ODM Program that includes leading audio manufacturers Eastech, Fenda, LiteOn Technology, Meiloon Industrial, Solidex Audio, Tymphany, Wistron, Zylux Acoustic, and Anam Electronics.

The company has now also successfully implemented a strategy to leverage the trend of using the TV as the central audio source for the living room and—uniquely—expand from the TV to the whole home, building unique multiroom and even personal wireless experiences.

Using standard Wi-Fi, Play-Fi now enables customizable home theater configurations from a simple wired or wireless soundbar connection to truly immersive configurations with discrete speakers, surrounds, and subwoofers, up to 7.2.4. Some of the wireless speakers may have built-in up firing drivers/channels and the system will recognize that. Users may even use the TV's internal speakers as part of a home theater system and can mix-and-match DTS Play-Fi soundbars, speakers, subwoofers, and amplifiers, regardless of the brand.

As Dannie Lau explains, Play-Fi is a natural fit for TV. "It works over Wi-Fi and now everybody has Wi-Fi on their TV. If the TV supports Atmos, it will decode it and give us up to 12 channels of PCM at 16-bit."

With the Play-Fi protocol, the TVs are aware of the connected speakers. As part of the process, all the active speakers are equipped with the Play-Fi module and, when they communicate with the TV, pass on their configurations—how many channels they support, maximum SPL, or if they contain immersive height elements. "Depending on which products and how many products you connect, the TV will determine what's the best rendering layout," Lau explains.

The current Play-Fi module that is integrated in speakers, amplifiers, and receivers has Wi-Fi built-in and has all the software needed for all sources, including Google Cast, AirPlay 2, Spotify Connect, Works with Alexa, and more. Generally, all manufacturers building Play-Fi-enabled products just take that module because they don't have to deal with the software or even the Wi-Fi implementation and certification.

"They just plug it in, and suddenly it's connected," Lau explains. "On the TV side, it is different. We do software porting for our TV partners, different SOCs and different operating systems. But for them this is the ideal solution. TV manufacturers use multiple



Dannie Lau, general manager, DTS Play-Fi, has been leading networked audio efforts since he founded Phorus, which was acquired by DTS in 2012.

operating systems. They might have Roku, they might have Android, they might have their own proprietary OS. But with Play-Fi, we're agnostic. They can use this across all the different operating systems, across all the different ICs, and have one technology to support their audio parts. We just want to sell attachments."

This approach has been followed by DTS since the launch of Play-Fi and the first licensing efforts. "Even if you bought a Play-Fi product 12 years ago from the Phorus brand using the first generation of modules, you can still use it with this TV in 2023. It won't have up-firing, but you can use it with the system," says Lau. "We're here as an ecosystem where interoperability and brand compatibility is maintained. You're supported the whole time. We don't abandon our partners. All these products still work with the latest developments that we do. I think that's what makes this ecosystem's story so powerful."

The Play-Fi vision was already perfectly clear when the company was still focused on creating a multiroom, whole-home audio solution, at a time when Sonos was still using its own proprietary



Dannie Lau and Justin Gibbins (DTS Play-Fi Product Manager) at CEDIA 2023, where the company promoted a demonstration of immersive audio support for DTS Play-Fi Home Theater, now supporting up to 12 channels of discrete sound.

mesh on wired networks. Fast forward 10 years, DTS Play-Fi holds some key advantages over other wireless audio technologies.

"A lot of people, 10 years ago, never thought we'd do surround sound over Wi-Fi. And we told people, "It can be done. And it's going to get better." At that time, we saw the features. Everything will be connected. Speakers, TVs, everything will have a Wi-Fi radio at some point. And Wi-Fi gets cheaper every year. And that's going to be the common denominator feature. That is what we set up to do back then, 12 years ago. Now, IoT chips have all the memory, processor, DRAM, Wi-Fi, all in one chip, and it's all inexpensive."

"Since 2014, we have been advertising Play-Fi as "Hi-fi over Wi-Fi." Our dream was always much bigger than just multiroom music," adds Lau.

One of the most important aspects is the fact that the Play-Fi protocol can have extremely low latency and multiroom audio synchronization between audio devices and video displays at sub one millisecond levels. Over standard Wi-Fi, other than Sonos, they are the only company to be able to do that.

The other aspect is the advanced integration and multi-service/multi-platform support that DTS Play-Fi now offers. The Play-Fi mobile app was completely redesigned in 2022 and provides all the functionality to work with all Play-Fi products, regardless of the brand, even integrating products that use AirPlay, Googlecast, or Alexa.

The interface offers an elegant way to find all the streaming services and allows full control over all services and features. That user interface even received a CES Innovation award nomination in the streaming category. And since a few years ago, Play-Fi has offered software for Windows, which is unique. Unlike Apple and AirPlay, Microsoft doesn't have streaming audio technology. So, this means that users can control their Play-Fi experience from any mobile OS, Mac OS, or Windows, and always have audio-video synchronization.

One other unique feature of the Play-Fi mobile app is that it can be used in conjunction with headphones for private listening at home. This means taking TV sound directly into the phone and using it as a portable listening device, while walking around the home, or while cooking in the kitchen, knowing that the audio will be in sync with the TV, and there will be no lip sync issues when looking at the screen. Up to four people can use this feature at the same time, and this can run simultaneously in conjunction with other speakers, all within the Wi-Fi range.

And now, DTS Play-Fi is finally able to offer a seamless wireless experience to consumers, from surround to immersive sound formats. "We think we have a big lead over our competitors. And then we've done a lot of work to combine that all into a singular user experience so that all works seamlessly for end users," Lau states confidently, knowing that it will only get better as Wi-Fi home networks are improving fast with Wi-Fi 6, 6E, and eventually Wi-Fi 7.

The Integration Control Ecosystems

It is basically over these wireless technologies or using wired solutions—analog, digital, or networked audio—that the whole residential integration industry works to create multiroom audio. Not being able to determine the types of sources or formats, manufacturers in this space are basically "aggregators" and platform integrators, where the largest advantage resides in the reliability of the installed systems and the convenience of single control interfaces.

A great example in this space is Sonance, one of the most familiar names in the custom integration industry. The San Clemente, CA-based company has been providing loudspeaker solutions and more since the mid-1980s. The company entered a new growth phase in 2019 with the acquisition and merger with James Loudspeaker, and the launch of a new brand identity. The goal was to streamline the number of brands they promote (a common error in the residential integration space, where large groups maintain dozens of separate brands that over time become irrelevant). While Sonance was originally known primarily for its architectural speakers, the same team also created IPORT, a brand that turned tablets into powerful home controllers; and TRUFIG, a flush mounting process that has become a favorite among architects, designers, and custom installers. In 2023, the Sonance brand has become front and center, and introduced a new important strategy—its close alliance with Sonos, creating the Sonos Architectural by Sonance range of products.

Originally, these were created as architectural speakers, designed for the Sonos Amp, which automatically recognizes the Sonance In-Wall and In-Ceiling passive speakers and optimizes its output. Taking things one step further, the Sonos Architectural by Sonance are the only third-party speakers to be TruePlay compatible, making the fine-tune adjustments required during custom installation more seamless than ever. The latest range includes Outdoor speakers with a weatherproof design as part of the Sonos Architectural by Sonance collection. The whole system includes an easy-to-use app that works with virtually any streaming service, Apple AirPlay 2, and offers a simple way to expand any residential system with more Sonos speakers and components. For a Sonos-native multiroom strategy, this Sonance partnership fits perfectly in the current market, allowing a professional approach to the Sonos ecosystem.

Another major player in this integration space is Savant, a company that recently fully embraced its smart home vision, in an effort to make professionally oriented projects to compete as that concept evolves. The company's latest innovation in this space embraces integration with the Apple ecosystem to allow users to conveniently control and automate their living spaces—which includes the Savant's growing range of smart power products. Using AI integration with Siri through the Apple iPhone, iPad, Apple Watch, HomePod, and CarPlay, Savant built a complete home solution that goes beyond voice recognition.

Over the last five years, Savant has developed AI-driven voice capabilities that have been designed to interpret natural language via Siri, to easily control entertainment, comfort, lighting, energy, and security throughout the entire home. Homeowners can trigger personalized Savant scenes by tapping an iPhone to an Apple AirTag or near-field communications (NFC) tags, setting up geofencing, and more possibilities.

"As the first complete control and automation solution built on an Apple platform, we're committed to providing Savant and Apple users with an advanced and best-in-class smart home experience

for every living space inside and out," says Bob Madonna, CEO of Savant. "And the combination of Savant AI technology with Siri allows for seemingly limitless applications driven by the imagination of the integrator and homeowner."

This allows users to control the whole multiroom experience in the home, such as having a podcast source playing through Apple Music to follow a user throughout the home. Or use an Apple Watch to activate a Savant Scene that plays an Apple Music playlist anywhere, all while light colors dynamically react to the beat and rhythm of the music. Or ask Siri on an Apple HomePod to dim the living room lights, turn on the Apple TV, and launch Disney+. In addition to these Apple devices, the dedicated Savant Pro Remote, which already integrates with Siri, is another voice input in the home that can leverage Savant AI technology for easy control.

The possibilities for applications indoor and out extend into the vehicle where Savant AI technology via Apple CarPlay can enhance personal safety and security as the user approaches the home.

With the recent launch of its daVinci 10.4 control system, Savant has also made it possible to build Apple Music into Savant Scenes and custom automations. Apple Music joined the Savant whole-home music experience and features a personalized dashboard, plus the ability to search and create playlists across multiple music services without the hassle of switching apps.

These latest integrations are built on existing efforts by the company, which include Amazon Alexa and Amazon Music as part of its Smart Home Entertainment Experience. This integration enables Savant homeowners to enjoy lossless streaming up to 24-bit/192kHz in their installations. Within the Savant smart home app, users can access Amazon Music, add their favorite playlists to the Savant Music home screen for convenient access, and build Amazon Music into any Savant Scene and scheduled event and access it through any Savant control point.

We will also mention Snap One, the leading manufacturer and distributor of "smart living technology," as the company describes itself these days. The combination of a succession of mergers and brand combinations, including what was previously Control4, Snap One can offer complete residential integration solutions that include multiroom and home theater audio.

Recent announcements in control and media distribution from Snap One include the company's Episode MoIP Amplifier and a variety of revamped, next-generation Triad speaker solutions and electronics featuring upgraded components, from soundbars and subwoofers to streaming amplifiers and total room control.

"Home entertainment is among the most reliable drivers for custom integration projects, and Snap One is pursuing innovative designs and leveraging new technologies to deliver everything our Partners need to create stunning home media experiences and spaces," said Alex Zaliauskas, Director of Product Management for Speakers at Snap One.

The new Episode MoIP Amplifier Series, the company's first audio-over-IP solution, offers streamlined integration to receive HDMI and line level signals from Binary MoIP media distribution transmitters. The three amplifier models make it easy to design IP-based rooms, with one-, three- and six-zone versions offering three, six, or 12 channels, respectively.



Most residential integration manufacturers still offer convenient and simple-to-install multizone amplifiers, which can be Class-D inside, and fully digitally controlled but remain essentially traditional analog audio designs. This is a recent example from Nice, the new company that resulted from the merger of all Nortek brands including ELAN, SpeakerCraft, and many others.

Binary MoIP provides flexibility to design expandable systems with affordable Cat5 or Cat6 cabling and place the amplifiers where it is convenient, including close to speakers to reduce speaker wire lengths and improve quality. MoIP requires fewer audio cables than traditional installations and enables rapid deployment, offering a total media distribution solution for residential and light commercial needs. These amplifiers are also able to receive network-based streaming audio from Control4 without traditional digital or analog inputs. This enables attractive IP-based multiroom audio solutions that are compatible with all major home automation systems and ready for expansion. Snap One's MoIP ecosystem allows creating centralized or decentralized media systems, making them easier to install and more flexible, compared to traditional setups.

Another new highlight from Snap One is the Triad SA1 Streaming Amplifier that replaces the Triad One as a high-quality option for streaming amplification. Offering two 200W channels, HDMI ARC/eARC support and support for Dolby Atmos and DTS formats, it's a solution for single-room media systems or 2.1 audio zones. The small-format device offers extensive DSP controls including tone, subwoofer EQ, room EQ, and speaker EQ presets for all Triad and Episode installation speaker models. The Triad SA1 Streaming Amplifier is powered by the Ryff audio platform within Control4, allowing users to stream high-resolution, lossless audio.



The Sonos Architectural by Sonance is a collection of passive speakers for indoor and outdoor use, designed and optimized for the Sonos Amp, developed and marketed in partnership with Sonance. Sonos sells these speakers directly to consumers, and Sonance handles the integration channel.

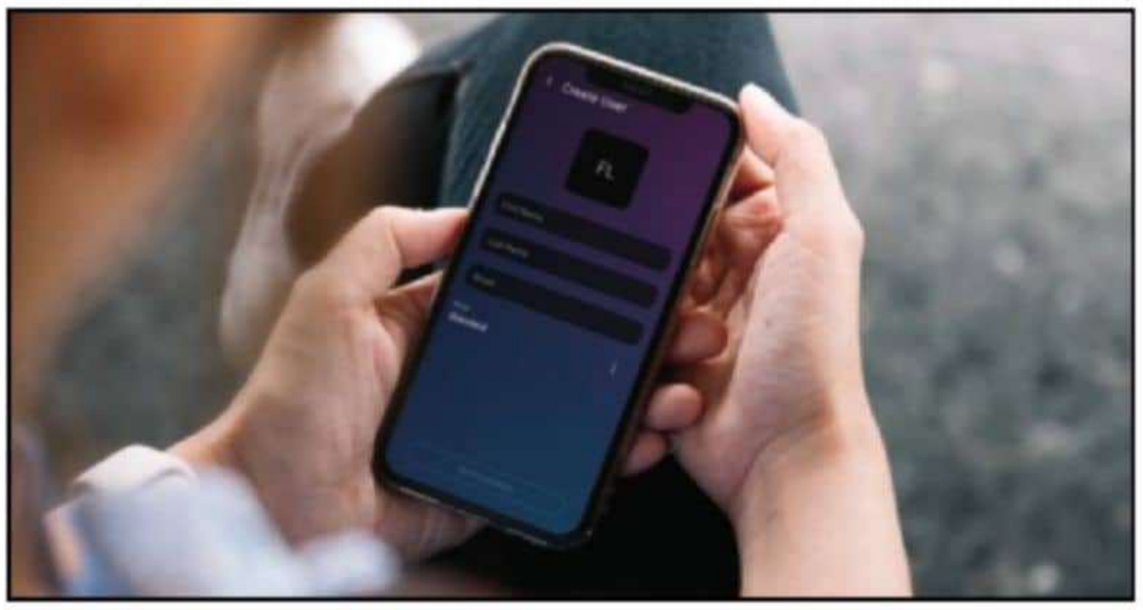


The Savant App is recognized in the residential integration industry for its elegance and powerful control over Savant home systems. Already supporting Amazon Alexa and Google Home/Google Assistant voice control, Savant now expanded its Savant AI technology with the first complete control and automation solution built on Apple Siri. This even extends outside the home with Apple CarPlay integration.

To power multiple audio zones, the new 4-Zone and 8-Zone Triad PAMP V2 amplifiers introduce a new visual design, even higher performance, and scalability. The Triad PAMP V2 boasts 25% more power for higher output with lower distortion, using Class-D amplifier modules from ICEpower. Global inputs and outputs, independent linkage, and bridgeable speaker outputs provide maximum flexibility for even the highest output installs.

And with native Control4 compatibility across the entire Snap One speaker lineup, integrators can easily manage and expand the system in the future if desired.

The latest Control4 OS 3.4 software update introduces features based on user feedback that are intended to increase end-user satisfaction. Named "Customer Empowerment features" by the company, the updates enable users to manage their own cloud-service subscriptions and allows both admin and standard account types, for an additional layer of security. Control4 admins can invite new users, edit existing users, and define roles for each user, but to ensure privacy and security, only admins will be able to see account information and update the property details.



Snap One recently launched the new Control4 OS 3.4, an experience-focused software update that empowers end users with streamlined account management features and user experience and introduces support for high-resolution audio streaming options from up to 12 native services, access to cloud subscriptions, and support for Amazon Alexa and Google Assistant.

The new OS also empowers users to create new notifications and modify their event notification settings easily from the Control4 App, with no separate portal or integrator assistance required.

In line with industry trends, Snap One's own research identified that users are showing the need for a balance between the "Do-It-For-Me" experience, a system designed and managed by trusted professionals, and DIY essentials, which the company says, "will mean happier customers and less-inundated technicians."

Finally, and because these integrated systems consistently rely on networking for signal distribution, Snap One is also increasing its Control & Networking Room portfolio with products from Araknis, Access Networks, Binary, and WattBox. This allows integrators of all specialties to benefit from the latest innovations and protocols integrated in the new lineups, including Wi-Fi 6 and advanced remote system access. Offering network switches with higher networking distribution speeds, up to 10Gbps connections with Layer 3 functionality, as well as static IP routing and multicast routing, the latest Araknis range of switches are ideal to support the demanding needs of wired and wireless multiroom systems.

Crestron Audio Solutions

I'll close with a mandatory summary of the segment leader. Crestron is going through a strong innovation and transformation stage, leaving the black-box approach behind, and focusing on the integrated experience, including systems outsourced from other manufacturers, such as speakers from Origin Acoustics.

On top of the wide range of new products and solutions launched from Crestron over the last three years sits the latest Crestron Home OS 4 home automation and control platform.

Once again, the focus is totally on the end-user, with a complete rethink of the user interface, allowing users to do a lot more by themselves.

At CEDIA 2023 in Denver, CO, Crestron even used the new Home OS 4 to run its entire booth, showcasing how powerful the platform can be even for the largest projects. Crestron Home OS 4 is a free software upgrade to any existing customer with a Crestron Home, and further improvements will continue to be added for no additional charge, and that includes additional drivers and third-party integrations.

As the modern smart home continues to evolve, Crestron homeowners are connecting more and more devices in their residences and taking more control of what used to be essentially an integrator-defined solution. Simultaneously, Crestron customers are asking for more natural and effortless ways to control and manage their lighting, comfort, entertainment, and more.

Crestron Home OS 4 offers interoperability and control of both Crestron and thousands of third-party smart devices and delivers a new level of user-friendliness, functionality, customization, and whole home control from a remote, touch screen, keypad, or mobile device for everyone, including residents, children, and even house guests.

The update also takes homeowner preferences to a new level, offering granular system customization, and an updated end-user schedule feature for events and automated tasks—all without a call to the dealer.

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PMR-5 G2	6	150W	4	-	-
PMR-6 G2	4	150W	4	2	-
PMR-7 G2	6	150W	4	2	-
PMR-8 G2	2	150W	4	1	-
PMR-9 G2	4	150W	4	2	1
PMR-10 G2	6	150W	4	2	1
PMR-11 G2	4	150W	4	2	2
PMR-12 G2	6	150W	4	2	2
PMR-13 G2	2	150W	4	1	1

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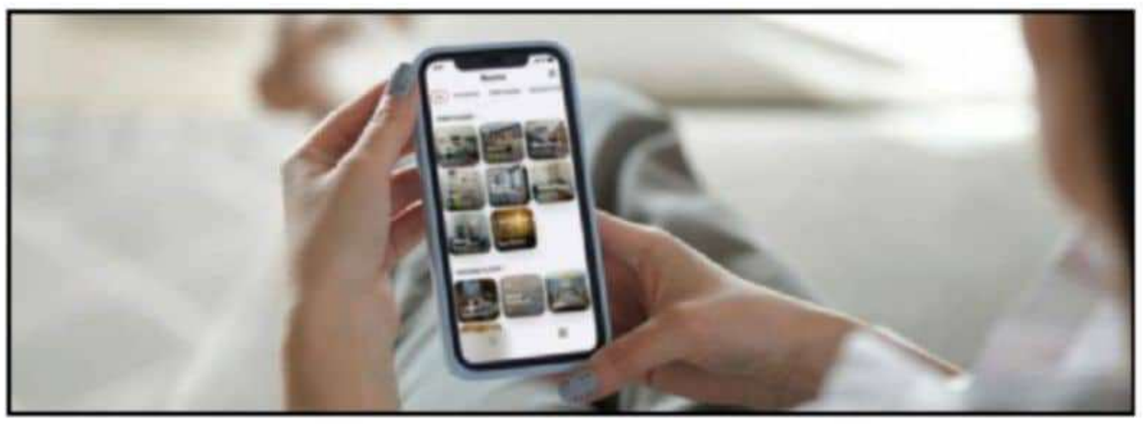
App for IOS and Android.
Drivers for automation: Crestron, Savant, Control4 and others.

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Market Update — Residential Integration



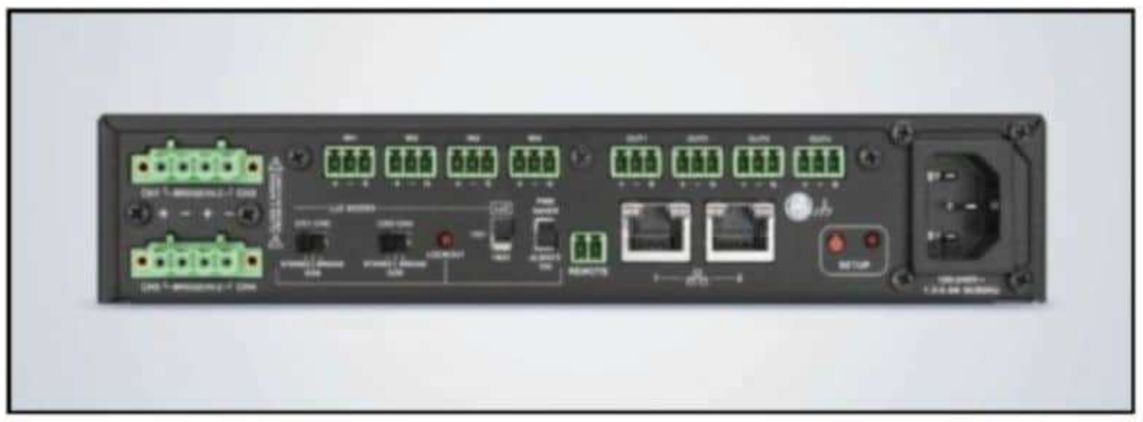
"Homeowners are much more knowledgeable and discerning when it comes to the technology in their home. Their expectations for home control, connectivity, and device security and privacy have soared since 2020," says Crestron when introducing its latest Home OS 4 home control software update.



Crestron Home OS 4 refines the smart home experience, showcasing a new look and feel with further personalization, and includes the ability for dealers to add almost any third-party integration.



The new DM-NAX-4ZSA-50 GaN amplifier, the DM-NAX-XSP decoding digital signal processor, and the DM-NAX-BTIO-1G wall plate.



When Ekin Binal and his team sat down to design the latest amplifier for the Crestron DM NAX line, the result was a flexible, 300W powerhouse with a diminutive form factor called the DM-NAX-AMP-X300. This solution accepts microphone signals, line-ins, network inputs, and output them on line or network outputs in native Dante, DM NAX or AES67 audio modes—the last ones being more suitable for residential projects.

Additional Crestron Home OS 4 upgrades include new voice control functionality for Xfinity systems, and Crestron touch screens running OS 4 can now also provide whole-home paging to make announcements to the entire household or a select group of rooms.

But moving on to new audio products, Crestron has now expanded its successful DM NAX Audio-over-IP technology line, which is a perfect solution for multizone installations, based on Ethernet infrastructure and compatible with the AES67 standard.

Completely new is the DM-NAX-4ZSA-50 GaN amplifier, the spiritual successor to the C2N-AMP-6X100/4X100 model, now delivering DM NAX technology at a lower price point. This 4+1 zone streaming amp (that's four amplified zones with a bonus line output for an audio-visual receiver) offers the latest GaN amp technology, and local inputs, all wrapped up in an elegant, passively cooled, surface-mountable half-rack box. With eight channels rated at 50W (RMS) each, this silent, fanless amp features five dedicated media players, four audio inputs (two for analog sources and one each for coax and digital), and two network ports. And the DM-NAX-4ZSA-50 offers native Crestron Home platform integration.

Crestron also debuted its DM-NAX-XSP surround-sound eARC solution—the company's first decoding DM NAX digital signal processor—to bridge streaming TVs to a multiroom audio distribution system. By leveraging ARC/eARC, the device can grab the audio content from smart TVs and tie that to a DM NAX over IP solution, where a second unit can feed a traditional AVR from there. An ideal solution to expand immersive audio formats to ceiling speakers. The device offers multi-channel audio inputs with a two-channel DM NAX downmix out with 8K HDMI inputs and outputs, along with two network ports, one with PoE.

Finally, Crestron introduced the DM-NAX-BTIO-1G, the first of several audio accessories coming to the DM NAX platform. The DM-NAX-BTIO-1G is a wall plate with Bluetooth communications and analog input/outputs that allows to easily add Bluetooth audio to any DM NAX Audio-over-IP, DM NVX AV-over-IP, Dante audio network, or AES67 AoIP system. The DM-NAX-BTIO-1G acts as something of a "wildcard," as it supports any audio source to bridge to DM NAX technology.

Resources

Creston | www.crestron.com

DTS Play-Fi | www.play-fi.com

Nice North America | www.niceforyou.com

Savant Systems, Inc. | www.savant.com

SKAA | www.skaa.com

Snap One, LLC | www.snapav.com

Sonance | www.sonance.com

Sonos, Inc. | www.sonos.com

WiSA Technologies, Inc. | www.wisatechnologies.com

XPERI | www.xperi.com

Advanced Audio Technologies

Brazilian Residential Audio Brand AAT Expands to North America



The AAT PMR-12 G2 is the second generation of the AAT multiroom amplifier range, offering six Stereo Zones with 12 mono channels with 150W RMS for each zone (900W RMS total power), six inputs (four analog and two optical/SPDIF), two built-in streamers with AirPlay, and 4x6 audio matrix. All outputs for each audio zone are preamplified, with one line subwoofer output for each zone.

Advanced Audio Technologies (AAT) manufactures a complete line of residential audio, home theater, and ambient sound solutions, offering a catalog of more than 90 high-quality products, all totally "Made in Brasil." From speakers to multiroom amplifiers, the manufacturer has a history of success in its home market, which is now expanding internationally, including the US.

With 16 years of history, AAT is a brand that is associated with advanced technology, quality and innovation. Brazil is a country with an extensive tradition as a manufacturer of loudspeakers and audio electronics, particularly for music, sound reinforcement, and car audio. Residential integration is a segment that has seen considerable growth in the last two decades, and the solutions manufactured by AAT stand out for their design and performance, becoming a reference in the country for consumers and system integrators.

Engineer João Yazbek, AAT's founder and managing director, has more than 30 years of experience in the Consumer Electronics (CE) industry and is recognized as one of the most influential audio and video professionals in Brazil. With a career founded on a solid academic background, and after a long period working at Philips Consumer Electronics in Brazil, he set up his own company and AAT was born. The mission was founded in his vision of turning dreams into reality, with modern products of the highest-standards.



The AAT PMR digital range of high-quality Class-D multiroom amplifiers with embedded streamers allows multiple control options, with automation drivers for Crestron, Savant, and Control4 (among others), and a dedicated Mobile App for iOs and Android. Installers will find a dedicated menu and password protection.

"AAT has total control over all stages of the production process, which is very rare these days and increases agility when producing high-performance equipment connected with the customers' needs," explains Yazbek. "We built the brand with a lot of work and creativity and the ambition of surpassing ourselves with each new project."

Such care and dedication meant that the brand quickly became one of the most respected in the segment in Brazil, enjoying the same prestige and recognition as manufacturers from abroad. Today, AAT offers the largest and most complete line of subwoofers in Brazil, and a variety of speaker options, of all types and sizes, for any type of project. AAT manufactures tower, bookshelf, wall-mounted, and ceiling speakers to match different customer profiles and architectural concepts, serving interior and outdoor environments.

With this strong technological DNA, AAT has developed a complete range of AAT PMR multiroom amplifiers, which are the brand's best sellers. Currently offering 13 models, from one to six stereo zones, with analog and digital control, and options

that feature one or even two integrated audio streamers, for online music playback in various environments at the highest quality. The flexibility of the AAT Multiroom digital line stands out, with the ability to control these amplifiers via Ethernet, Serial interface, infrared (IR) and directly from the front panel encoder. All digital AAT PMR models benefit from a dedicated mobile app (available both in iOS and Android) and are supported by an extensive series of automation drivers, including Crestron, Savant, and Control4, among others.

And these multiroom amplifiers are complemented with a range of audio streamers with Wi-Fi, Airplay and Bluetooth connectivity, amplifiers for speakers and subwoofers, and a complete line of Power Conditioners developed for the tricky conditions of the country's mains supply, which ensures maximum system protection. As an example, the AAT PWC-8 model is among the most advanced power conditioner in the world due to its Internet Monitoring and Reset feature, allowing remote power monitoring in real time, and the possibility for full control over the Internet. This advanced power conditioner is also supported with drivers for the most used automation vendors.

More information about the company and its products is available on the company's website.

www.aataudio.com.br

Origin Acoustics

WiiM Professional CI Series Smart Audio Solutions

Origin Acoustics, the dynamic and fast-growing architectural audio and electronic solutions manufacturer and distributor, announced a strategic partnership with WiiM, the streaming audio technology brand created by Linkplay to provide high-quality smart home experiences through accessible AI-connected products. Origin Acoustics and WiiM have joined forces to co-develop cutting-edge WiiM Professional CI Series products, designed to cater to both residential and commercial custom installations. Origin Acoustics will become a crucial distribution partner for all WiiM Home and Professional products across the United States and Canada.

Linkplay's expertise in streaming audio software and hardware has already led to the creation of several best-selling music streamers—the WiiM Mini, WiiM Pro and its latest innovation, the WiiM Pro Plus. The skilled team leading the WiiM brand effort, comprising hardware and software engineers and industry veterans, has proven the versatility, user-friendliness, affordability, and the quality of its technology platform in a market space with very few alternatives that are truly comprehensive, ecosystem-independent, and interoperable.

Origin Acoustics is expanding its market approach and providing insightful direction to WiiM, resulting in the co-development of the brand's first-ever Professional Smart Audio products designed specifically for custom integration in residential and commercial environments. Each product within the new WiiM Professional CI series features the same user-friendliness and compatibility with the various control platforms, already proven in the direct-to-consumer range of products launched so far. Now, with the launch of the new WiiM Professional CI Series, origin is also offering a profitable avenue for dealers.

Four new products from the WiiM Professional CI Series already available include the WiiM CI MOD S1 single-zone streamer, WiiM CI MOD S4 four-zone streamer, WiiM CI MOD A-110 single-zone 110W amplifier, and the WiiM MOD CI A1-60 single-zone 60W amplifier. Noteworthy integration features encompass ESS DACs, rack-mountable form factors, audio inputs and subwoofer line outs.

The WiiM streamers support AirPlay 2, Alexa Casting, Spotify Connect, TIDAL Connect, Qobuz, Amazon Music, TIDAL, Deezer, TuneIn, iHeartRadio, and more. It also feature Chromecast Audio and are extremely easy to set up and configure with home systems, supporting OTA software updates.

Additionally, Origin Acoustics, a company known for its effective channel management practices, has been selected to distribute WiiM's Home and Professional products in the US and Canada.

w**ww.linkplay.com | www.originacoustics.com**
www.wiimhome.com



Origin Acoustics and WiiM have joined forces to co-develop cutting-edge WiiM Professional CI Series products, designed to cater to both residential and commercial custom installations.

Lenbrook International

BluOS 4.0 Reimagines the Multiroom Mobile App Experience

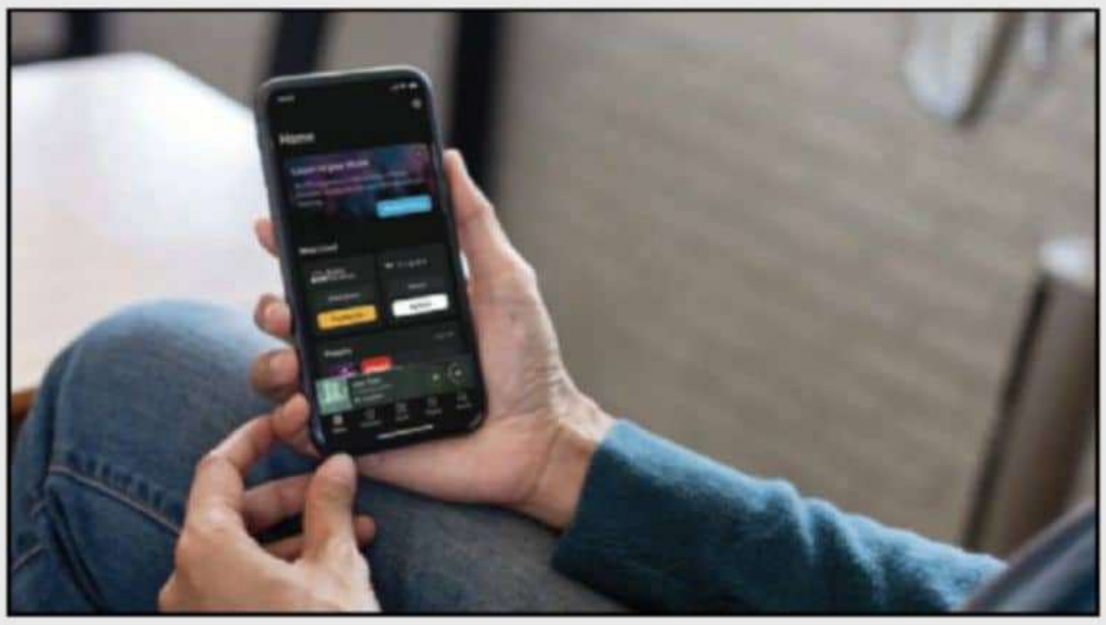
BluOS is already one of the most elegant software environments for home entertainment, and its importance has gradually expanded in the residential integration market. Lenbrook International has now introduced the latest update to its premium multiroom music management platform BluOS 4.0.

Made up of an operating system and a control application for smartphones, tablets, and PC desktops, BluOS is a premium multiroom audio ecosystem that manages stored and cloud music sources and playback, with support for high-resolution audio streams up to 24-bit/192kHz. Adopted by some of the most renowned residential hi-fi and commercial audio brands and integrated with numerous connected devices and voice control systems, BluOS allows for interoperability among enabled devices across brands for maximum versatility and use cases.

Integrations with popular streaming music services like Tidal, Spotify, SoundMachine, and SiriusXM for Business, as well as featuring support for MQA, FLAC, AAC, and other high-resolution formats and codecs, BluOS offers virtually unlimited access to music of all genres, for any occasion.

This 4.0 update expands how the final user interacts with all the connected systems in a more personal way.

A key aspect for BluOS 4.0 was to improve navigation with easily identifiable icons, adding a favorites screen for quick



access to playlists, artists, albums, songs and stations. The Music screen combines music sources, music services, and device inputs between connected zones, all in one area. The Player can now toggle between Now Playing and Play Queue screens, there's an icon for grouping, and indicators for sampled audio quality and zone playing. Finally, a new dedicated Search screen searches by music service for more specific and faster results.

www.bluos.net
www.lenbrook.com



Improving GaN with Digital Control

Axign Class-D Audio Amplifier PWM Controller Demonstration Board

The AX5689 is an eight-channel audio amplifier controller IC with digital inputs and CMOS level PWM outputs, enabling high-order digital control loops with post-filter feedback. This article reviews the reference design board using GaN Systems output devices, configured as a two 500W channel BTL (2Ω) at 50V.



By
Stuart Yaniger

An apocryphal story about Alexander the Great says that, upon surveying his domain, he wept because he had no more worlds to conquer. Every time I test a new piece of audio electronics, the same thought crosses my mind—Audio engineers seem to have conquered all of the traditional limitations and problems of the earlier generations of equipment in the electronics chain, and I imagine that, paradoxically, this can in itself be a bit of a frustration. "Have I innovated myself out of a job?"

Perhaps this is nowhere more evident than for power amplification, where distortion has been banished to the nearly unmeasurable, source impedances are smaller than the resistance of binding posts, massive amounts of power (almost unthinkable at the beginning of the 21st century) are available in tiny and cool-running packages, and prices have plummeted. We've also hit asymptotic limits on efficiency, well surpassing the "80" part of Pareto's 80:20 rule.

It appears the principal differentiators among modern amps are packaging, features, and price. Likewise, amplification for OEM products (e.g., active speakers) has even less room to grow and improve since there's no consumer-facing packaging. Could there be any new approaches left that bring tangible advantages to the table?

The Technology

Enter Axign's new Class-D amplifier controller chips. Axign is a Netherlands-based fabless chip supplier. Its new PWM Class-D controllers are supplied in two types: the AX5688 for two-channel bridge-tied load (BTL) output configuration and the AX5689 configured for a four-channel BTL. The outputs are pulse-width modulation (PWM) at CMOS level to drive the output stage driver circuits. So far, all conventional. But looking more closely, the Axign chips have some very novel elements.

A block diagram of a system using the Axign controller is shown in **Figure 1**. First, unlike conventional Class-D controllers, the inputs are natively digital, nothing within the chip (with one exception—a group of ADCs, which we will discuss a little later in this article) is analog until the output. The chip's input is addressed via the usual I^2S protocol. The open loop gain versus frequency is also implemented digitally, allowing a high open loop gain, with a fifth-order roll-off. This configuration allows the distortion to be consistent across the audio band, but also leads to high suppression of the carrier.

So, you might wonder how feedback is done, since that usually involves taking the difference between input and a fraction of the output signal, usually an analog task. The answer involves another interesting innovation—digital feedback.

This technique was described in a series of papers and patents from Daniel Schinkel and Fred Mostert (see Resources). Instead of a conventional feedback loop, the differential-mode amplifier output is digitized by a high-speed low latency Delta Sigma

(ΔΣ) ADC, then input and output are subtracted in the digital domain.

Why hasn't this conceptually simple approach been done before? This path is not an easy one. The ADC is required to have high bandwidth and low latency for stability. ΔΣ ADCs can have issues with high quantization noise. Digitally filtering it introduces latency and phase shift, which complicate stability. So, almost on-theme here, the ADC derives feedback via a DAC that has a low-pass filter function. This allows compensation for digital filtering and greatly reduced noise in the audio band. In the implementation tested here, the sampling is done every 20ns, which is more than fast enough to achieve a stable control loop well beyond the audible range. In the Axign chip, the ADCs are doubled up to get a 3dB increase in signal-to-noise. Surprisingly, there is no dissipation penalty for the ADCs. Axign's design reduces their power use to 15mW each.

The output drive is configured for multiphase, which greatly reduces electromagnetic interference (EMI) and demands on the output filters, and the feedback is taken following the output filters. It's well known that nonlinearities in output inductors are often the limiting factor for Class-D amplifier performance. This can be overcome at significant money, space, and heat costs with inductor design, but even more effectively with post-filter feedback. I've seen this used in several other premium Class-D amplifiers (e.g., Orchard, Hypex, Purifi), but in combination with multiphase drive, efficiencies, and performance can be maximized at low cost.

Speaking of efficiency, the demo board uses GaN output devices. As I've discussed in a couple other articles, GaN brings a lot of benefits, specifically from the higher charge carrier mobility and thus lower channel resistance. In theory, one can achieve higher switching speeds with GaN compared to silicon, but this goes well past the point of diminishing returns. Where GaN really shines is exploiting the low resistance (and thus low I^2R heating) to drive very low impedance loads.

For active speaker use (a key target for the Axign chips), efficiency is particularly important since the amp will most often be built inside a box without particularly good thermal flow. Of course, at high powers, efficiency is high because of Class D (and the use of GaN output devices), but realistically most operation is done at lower powers, where Class D has far less of an efficiency advantage. Any amp has what I call a "power overhead," that is, the power needed to light the lights, run the logic and input stages, and of course any ADC or DAC. So as a practical matter, the highly touted Class-D efficiency advantage is honored more in the breach.

Axign prioritized efficiency at lower power by at least two means. First, the low power consumption of the ADCs; and second, through the use of a novel PWM scheme, which takes advantage of the differential outputs and modulates the common-mode voltage. This also greatly reduces the stress and dissipation of the inductors used in the output filters. Because of the low power consumption resulting from the design, the dissipation of the amps at idle and low power is under 1.5W per channel, enabling low heat output and higher reliability in active speaker use. Of course, at high power, the Class-D advantages dominate, and the use of low on-resistance GaN output devices further increases the advantage.

The Axign GaN Demo Board

As mentioned, the board (**Photo 1**) is a collaboration between GaN Systems and Axign—since its release, GaN Systems was acquired by Infineon, so there's some question about the continuation of the corporate relationship, but the

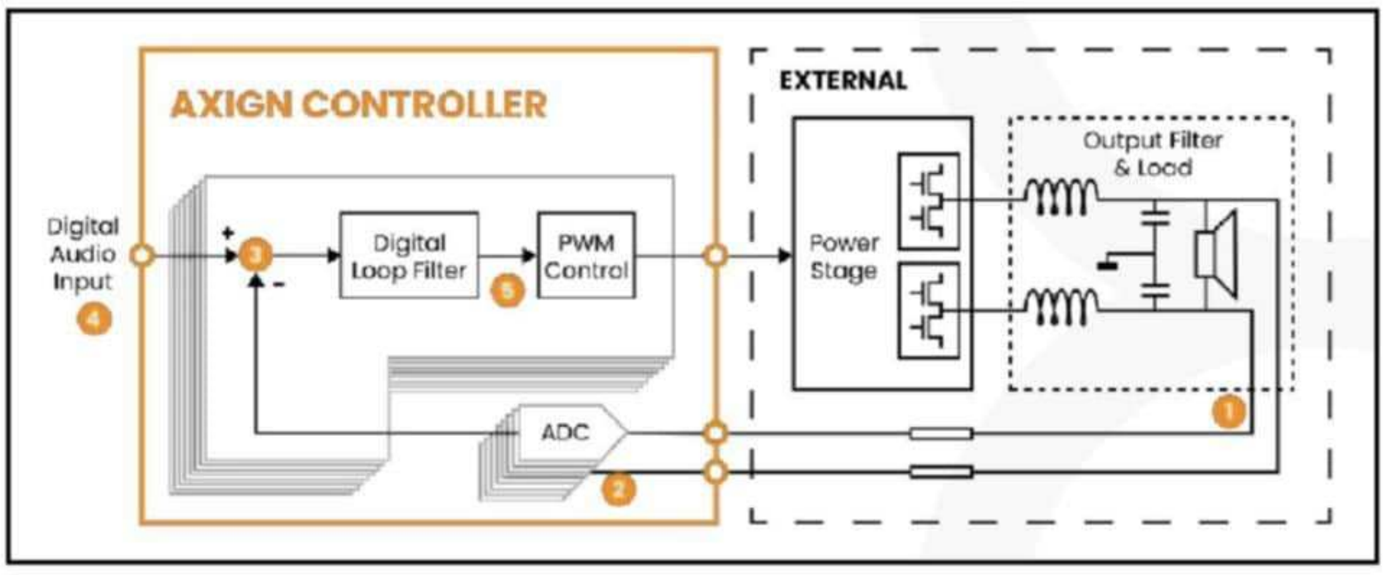


Figure 1: A basic block diagram of an amplifier incorporating the Axign controller, which uses digitized post filter feedback.

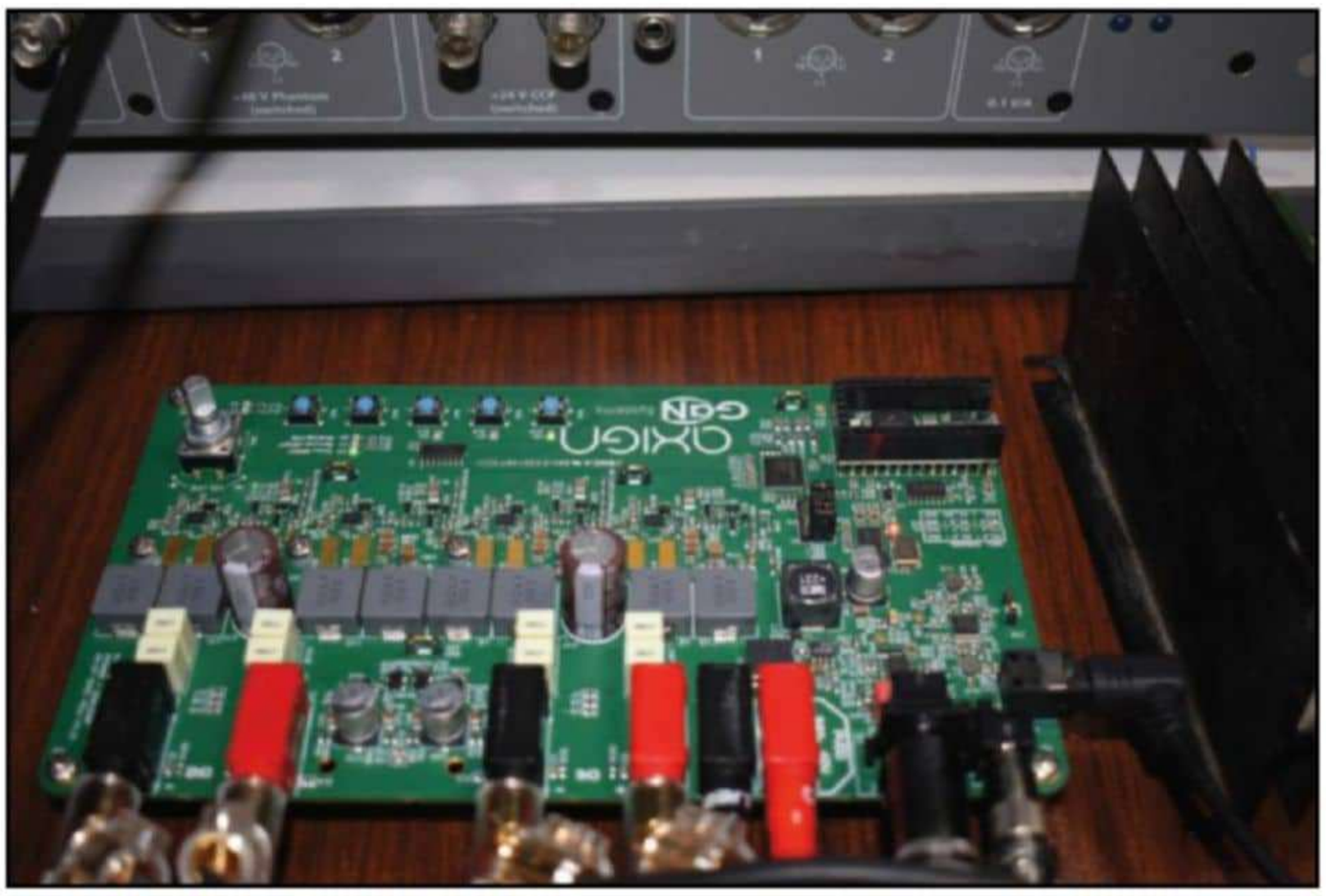


Photo 1: The Axign/GaN Systems Class-D GaN audio amplifier board seen here under test.

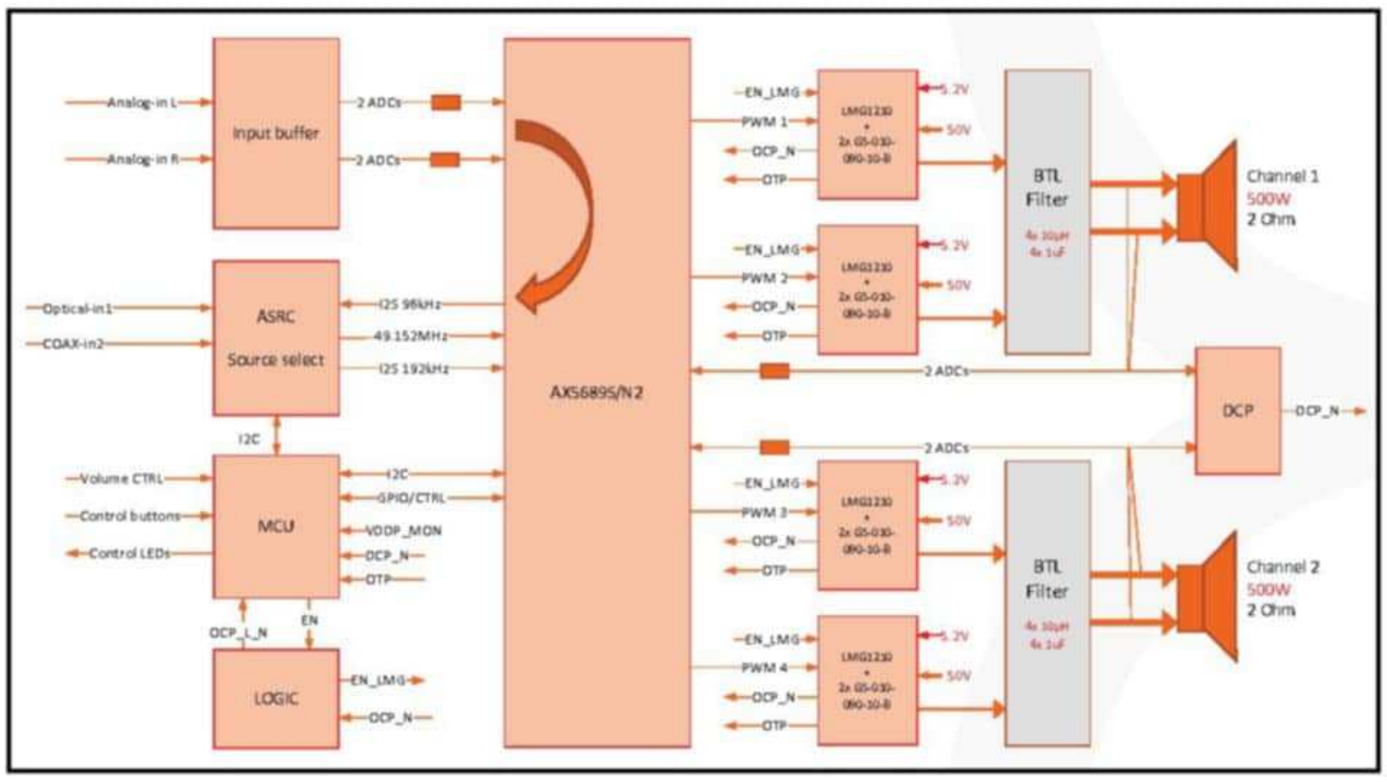


Figure 2: The GaN Systems-Axign demo board features a variety of input options and controls.

Axign controller chips could be used with other output transistors (e.g., silicon MOSFETs, SiC, other GaN devices) just as easily. So, although details of this particular board may vary a bit for future designs, the basic concept has a lot of flexibility in application.

A block diagram of the demo board is shown in **Figure 2**. You can see that the provided inputs include SPDIF coaxial and optical, single-ended analog, and I²S. Since the controller is natively digital, the analog inputs go through a buffer and a high-speed ADC at the input before the AX5689S controller. The ADC and buffer are good enough that there's no measured difference in performance between using analog and digital inputs. Digital inputs are first fed to a high-performance CS8422 sample rate converter where they are resampled and sent to the controller via an I²S bus at 192kHz.

Other controls on the board are managed by a Teensy 3.2 microcontroller, including thermal management, DC offset, PWM controller programming, and sample rate conversion. There's an optical rotary controller for volume level.

The output stage uses GaN Systems's GS61008 transistors, which are bottom-cooled. They feature an extremely low 7mΩ on resistance, which is key to the ability to run cool without external heatsinking. The driver chip for the output stage is a Texas Instruments (TI) LMG1210. Of course, other devices and drivers could be used in applications based off the demo board's reference design.

The demo board requires a power supply of 50V nominal. Because the board includes a power inverter onboard, a single-ended supply is sufficient. I bought a (relatively) inexpensive 48V supply rated at 20A to run the board. Conveniently, the power connections to the board are banana plugs rather than a proprietary connector or (worse yet) screw terminals. The first power-up produced no sparks or smoke, which was a good omen.

Measuring the Axign GaN Demo Board

Measuring modern Class-D amps is usually an exercise in ennui. They are much of a muchness. A pretty excellent muchness, but not a lot of variation. The Axign demo board held a few surprises.

As usual, my test setup comprises an Audio Precision APx525 running the APx software version 6.1. I have a set of 8R and 4R high power dummy loads and a small two-way bookshelf speaker for monitoring. Voltage levels are double-checked with an HP 3466A precision multimeter. Most measurements were taken with the APx525's built-in AES17 20kHz filter, but it was obviously turned off for the wideband noise measurements.

The noise spectrum of the demo board is shown in **Figure 3**. There are two things of note here. First, one can see the noise-shaping, which removes any issues of audible hiss even with extremely sensitive speakers. The low 1/f corner is icing on the cake. I measured under 140µV in the audio band (20Hz to 20kHz), and the white spectrum means most of even that minuscule amount that lies outside the regions of maximum human acuity. Second... where's the carrier? Normally, it shows up in the 100mV range for higher-end Class-D amplifiers, which isn't really an issue, but the outstanding result here speaks to the effectiveness of the multiphase drive and balanced output filtering, as well as taking feedback from the load side of the filter.

Output impedance is one of the most critical performance attributes of a power amplifier. Although humans have relatively low sensitivity to distortion, we can pick up very small deviations in frequency response. Because an amp's output impedance forms a voltage divider with the load impedance, lower values mean lower frequency response variations with actual speaker loads. There are several ways of performing this measurement. The simplest is to input a signal, measure the output voltage, change the load impedance, and see how the measured voltage changes. The source impedance can then be easily calculated from the

About the Author

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



voltage divider equations and Kirchoff's Law. Ideally, this is done at various frequencies across the audio spectrum since the source impedance usually varies with frequency.

Imagine my surprise when changing the load from 10R to 2R caused NO change in the output voltage! This was true at several spot frequencies (50Hz, 200Hz, 1kHz, 5kHz, 15kHz), and is a remarkable result, predicting that the amp's sound will remain uncolored even using speakers with wildly swinging impedance. A nice illustration of this is shown in **Figure 4**, which gives the level relative to 2.83V over the usual 20Hz to 20kHz frequency range for a pure 8Ω resistance and a two-way sealed box bookshelf speaker. Note the highly expanded scale. The amp's basic frequency response has an inaudibly small 0.08dB dip in the top octave, but there was absolutely no measurable difference using a test resistor and a speaker.

Other than frequency response, primus inter pares for amplifier characteristics is power delivery. **Figure 5** and **Figure 6** show the demo board's distortion versus power for two different loads, 8R and 4R, respectively. The distortion knees are at somewhat over 100W and 200W, also respectively, with the 1% distortion point at about 140W and 280W for the two loads. Note that the curves

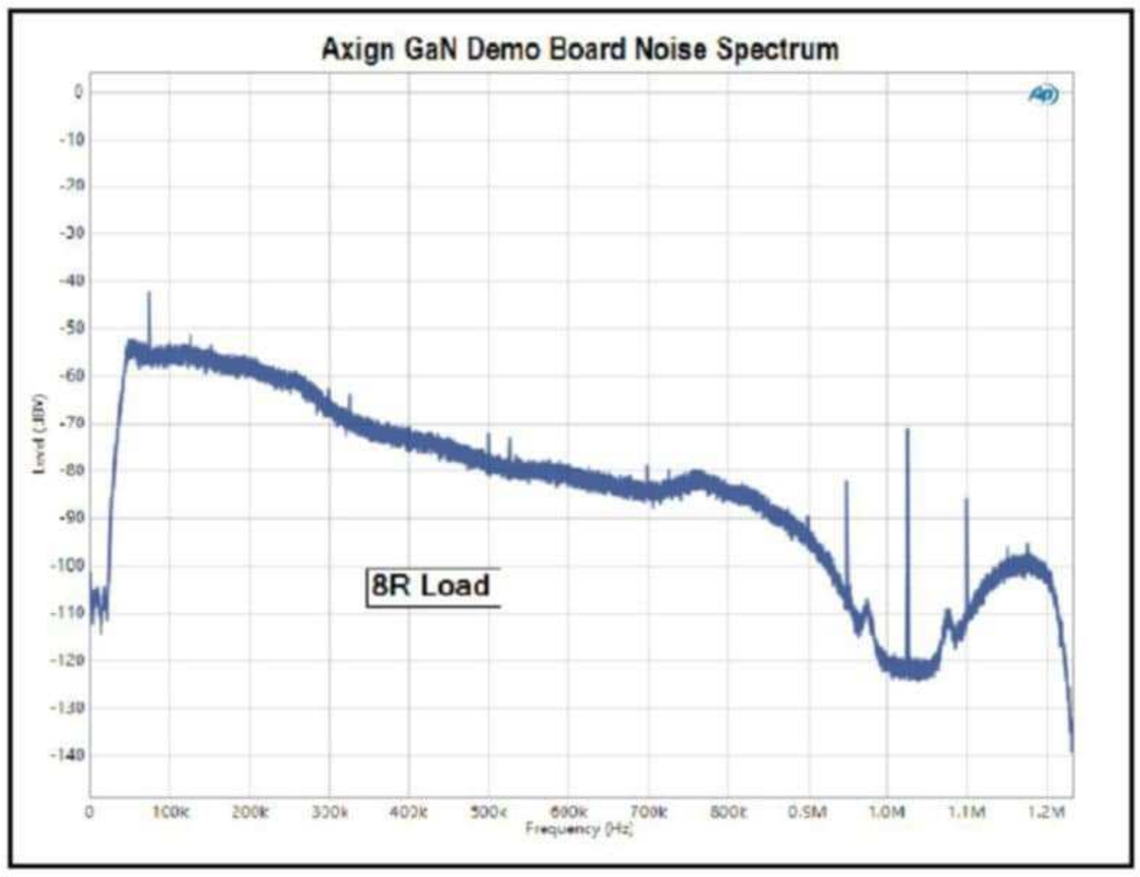


Figure 3: The idle noise spectrum of the demo board is remarkably quiet in the audio range and shows essentially no trace of carrier frequency.

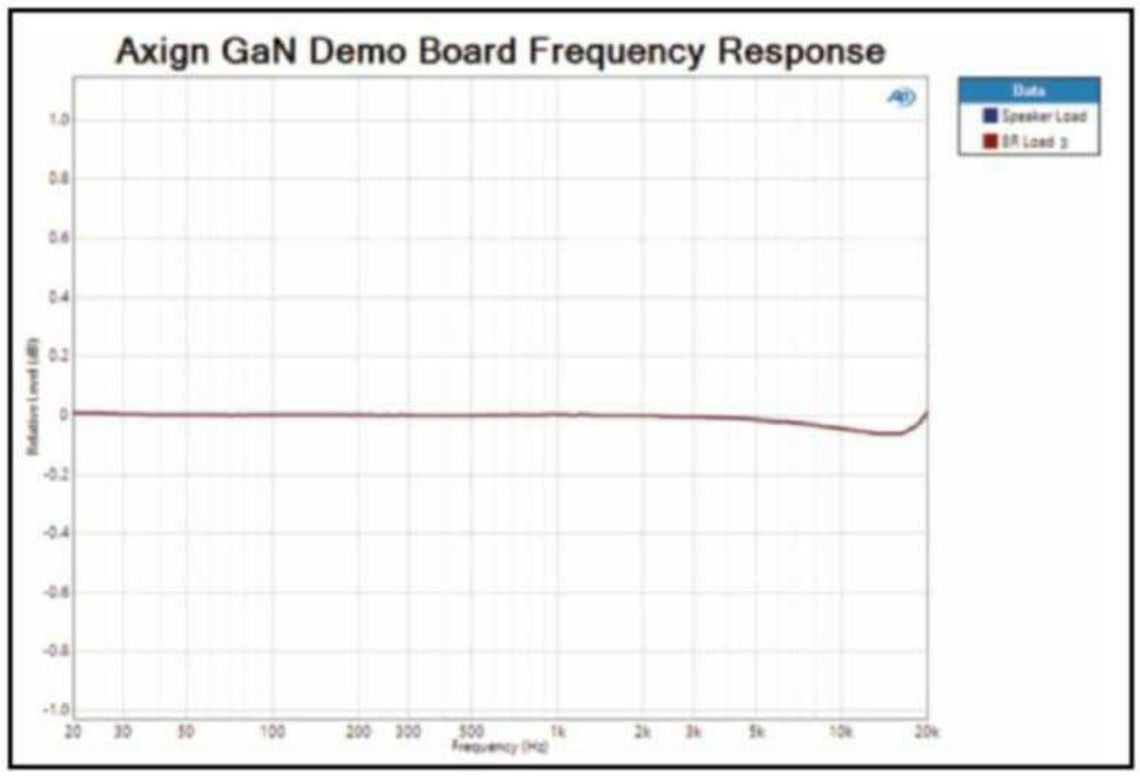


Figure 4: The frequency response is unaffected by the load, a function of the demo board's exceptionally low output impedance.

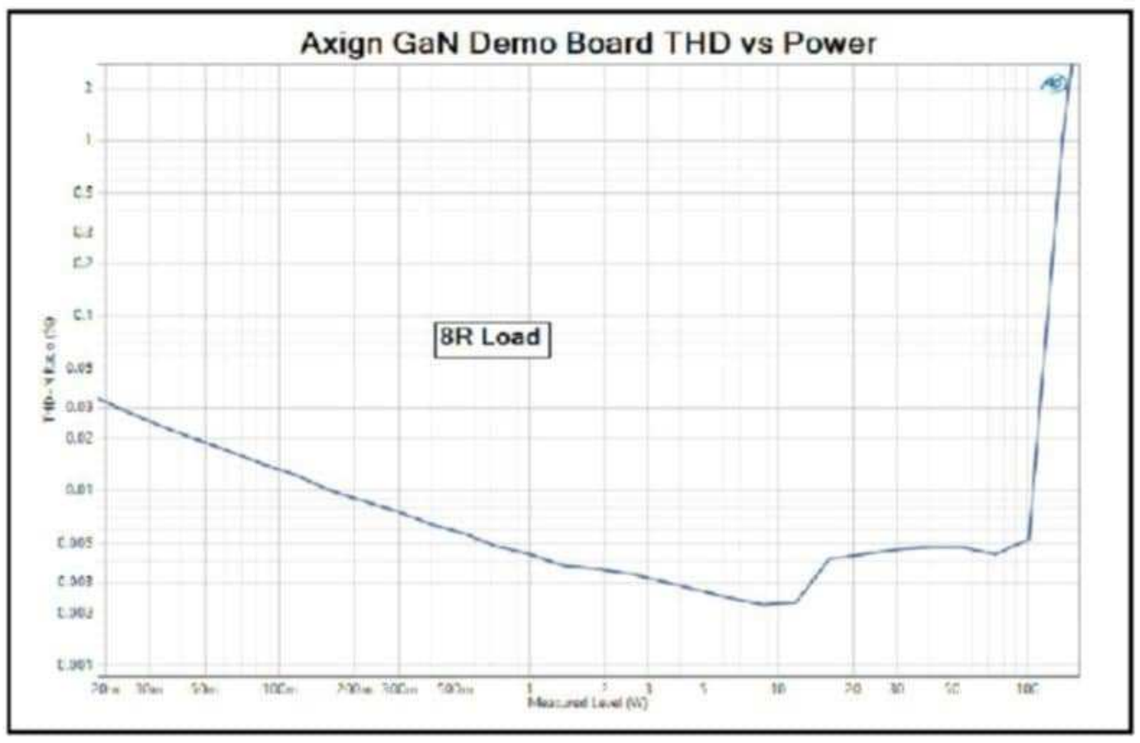


Figure 5: The power delivery at 8R well exceeds 100W. The distortion step at about 12W is a measurement artifact (see text).

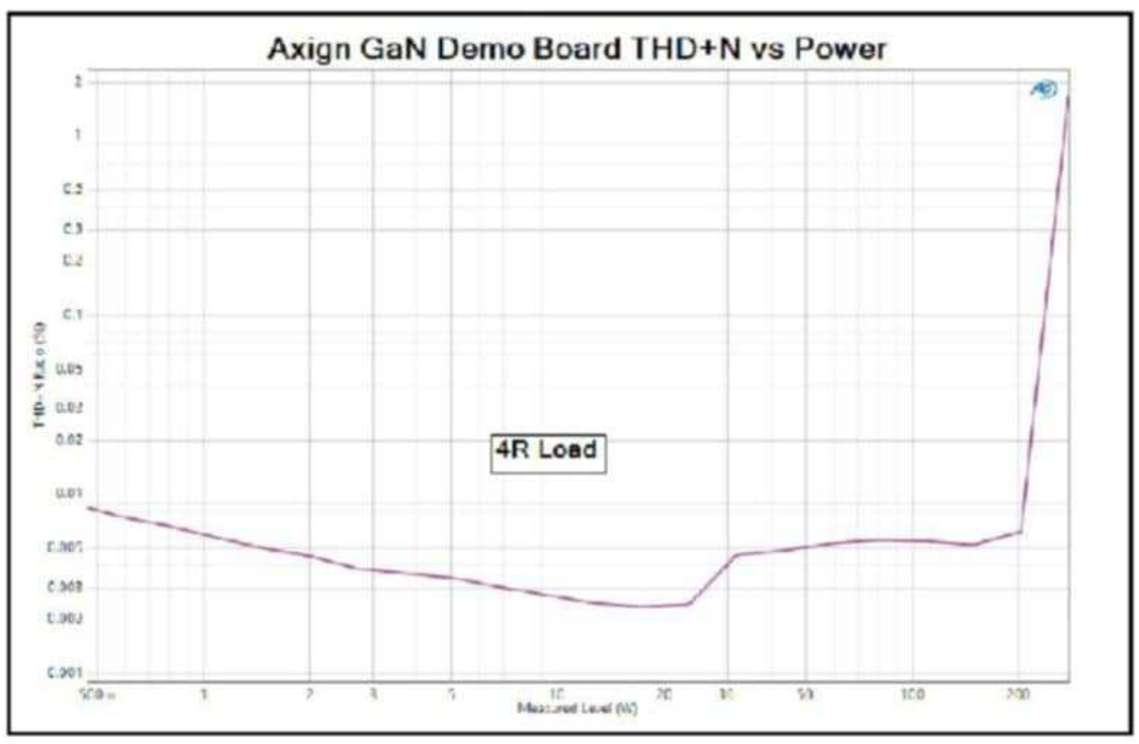


Figure 6: The power delivery at 4R is almost perfectly double that at 8R.

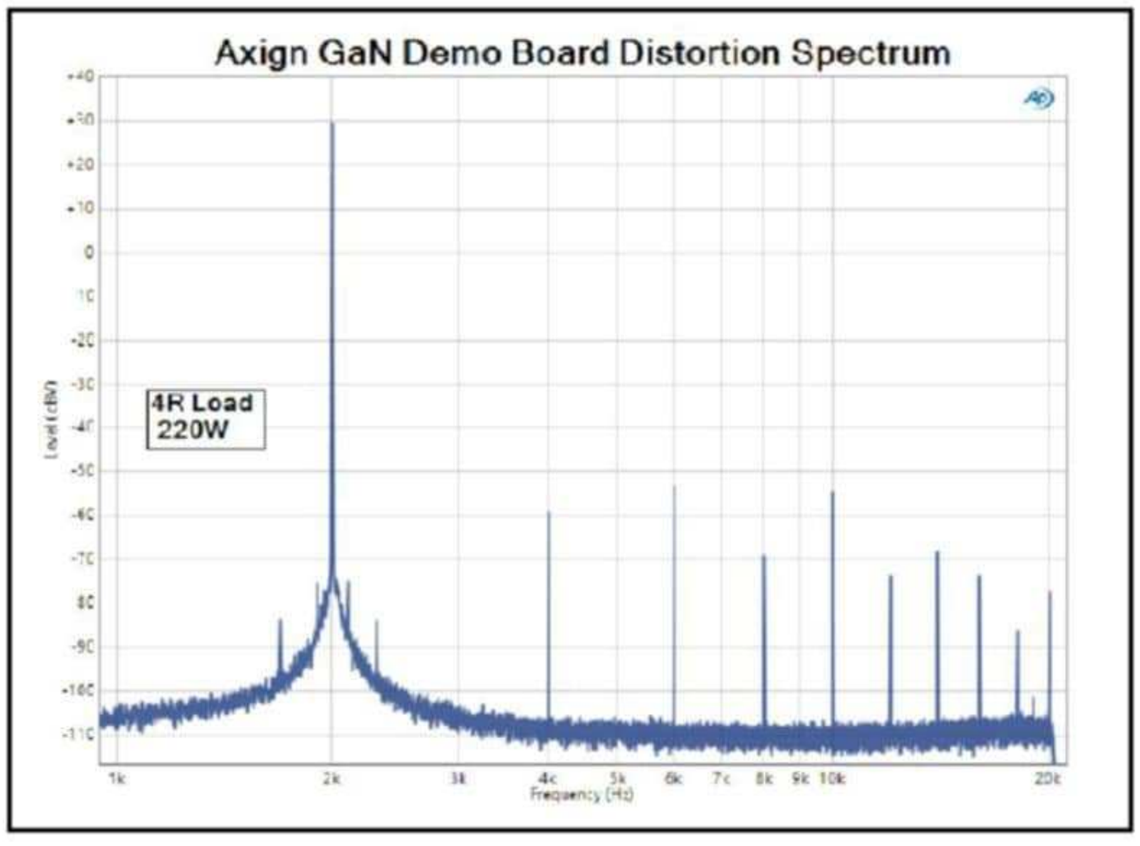


Figure 7: The harmonic distortion spectrum at 220W for a 4R load shows mostly odd harmonics at a low level.

look almost identical but scaled for power by the load resistances. This is suggested by the source impedance measurements, but it's nice to see that the almost perfect doubling of power going from 8R to 4R loads is maintained even at high power.

The step in distortion seen at about 12W and 24W for the 8R and 4R measurements is actually a measurement artifact and not an actual distortion increase in the amplifier. The artifact is a result of the previously mentioned common-mode modulation and corresponds to the range changing point of the APx525 analyzer. Of course, this is not an issue for loudspeaker loads, which are driven purely by the differential signal.

Figure 7 shows the distortion spectrum for a 4R load with 220W. It's dominated by the third harmonic at about -85dB, but is otherwise unremarkable. Likewise, a pair of 10V signals at 10kHz and 11kHz across an 8R load gives the intermodulation spectrum shown in **Figure 8**. The largest error is a sideband at -96dB; this is quite an excellent result.

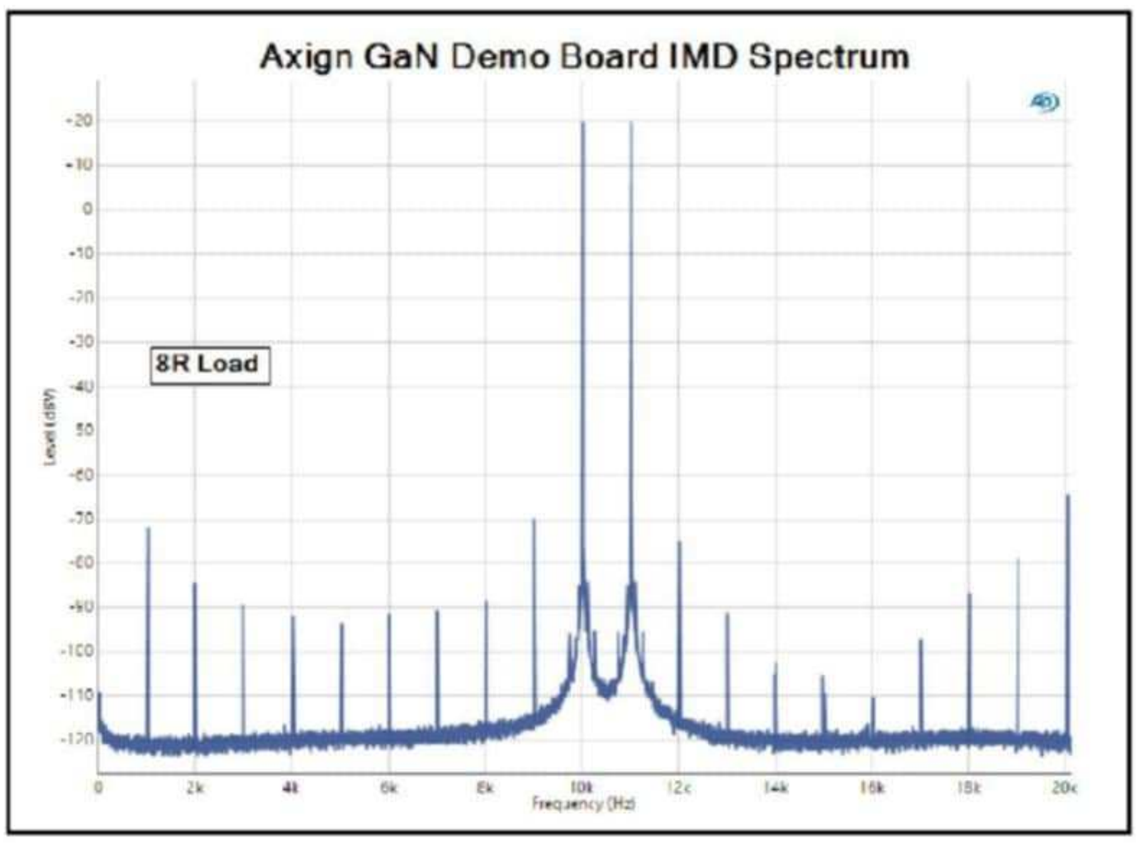


Figure 8: The demo board's intermodulation distortion spectrum for a mixed 10kHz/11kHz signal at 10V per component is very clean.

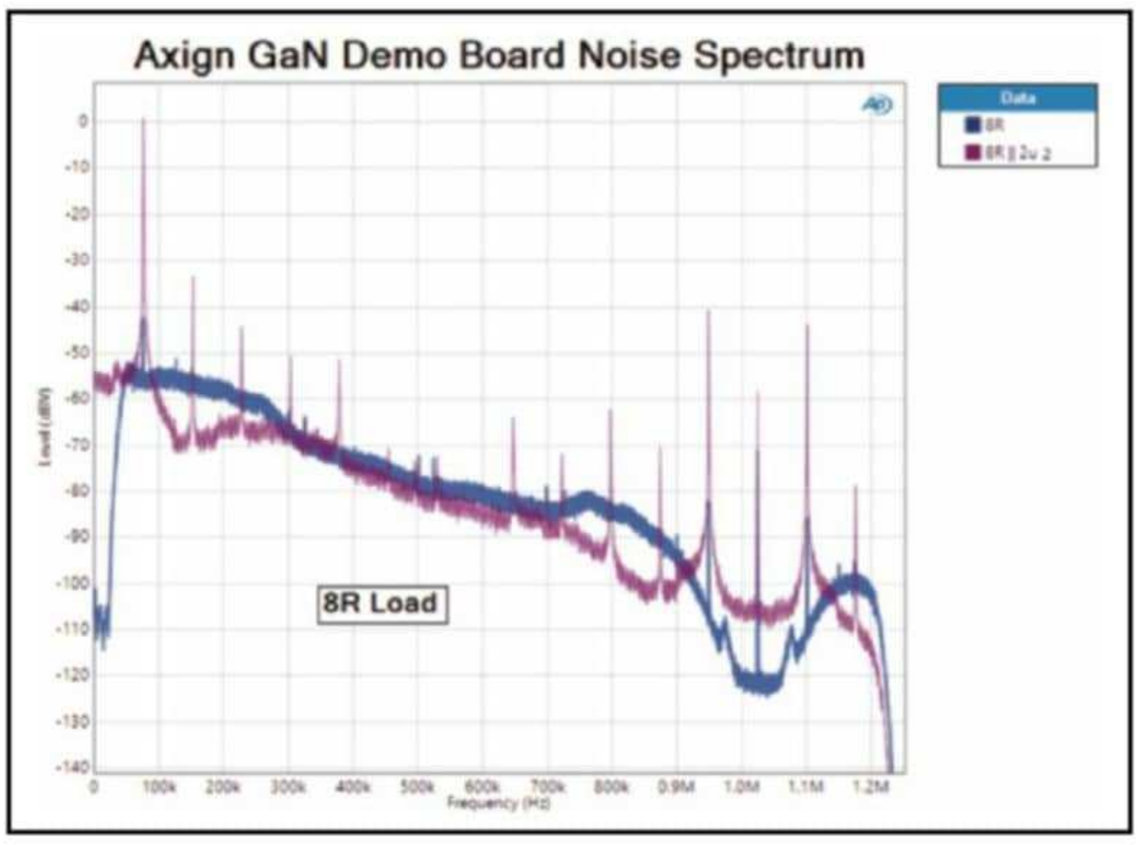


Figure 9: Adding a severe capacitance to the test load caused a rise in noise and ultrasonic oscillation.

One test that I've incorporated into my measurement suite is driving a heavily capacitive load. I used a 2µF capacitor across an 8R load. This is a far larger capacitance than nearly every electrostatic loudspeaker, but there are a very few edge cases which could approach that. The demo board was not at all happy with this load! **Figure 9** shows the change in the noise spectrum between the 8R load and the 8R paralleled with the capacitor. Large discrete noise spikes arise (near oscillation) at ultrasonic frequencies and the noise level in the audio band rises sharply. The idle waveforms (**Figure 10**) show the oscillatory behavior induced by this extremely difficult load. For the very few ESLs that have this high capacitance, there could be an incompatibility issue, but for the 99.9% of more normal speakers, this is not a factor at all. Certainly, with the ESLs I used for listening tests, this problem did not arise, and I reiterate that this test is meant to be an extreme case and not representative of any dynamic speaker or the majority of ESLs.

A sweep of distortion versus frequency at 4V out for 8R, 4R, and 8R paralleled with the 2µF capacitor is shown in **Figure 11**.

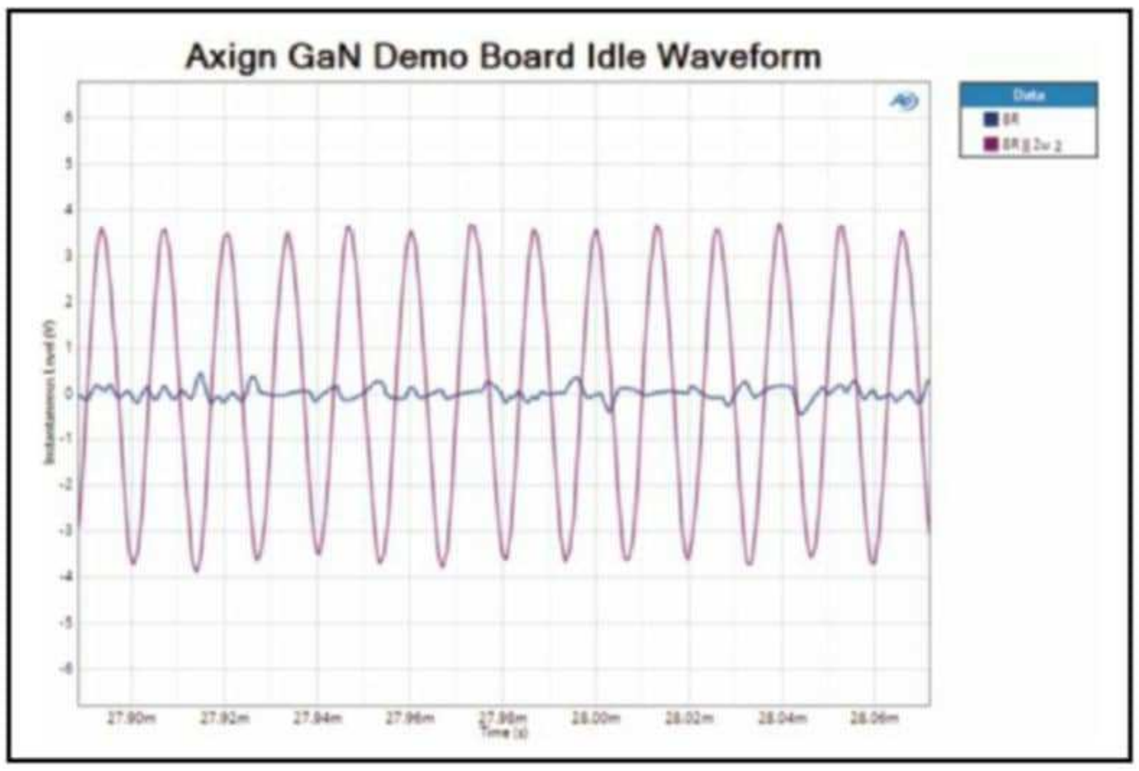


Figure 10: The idle waveform for a normal load and a severe capacitive load shows the induced oscillation.

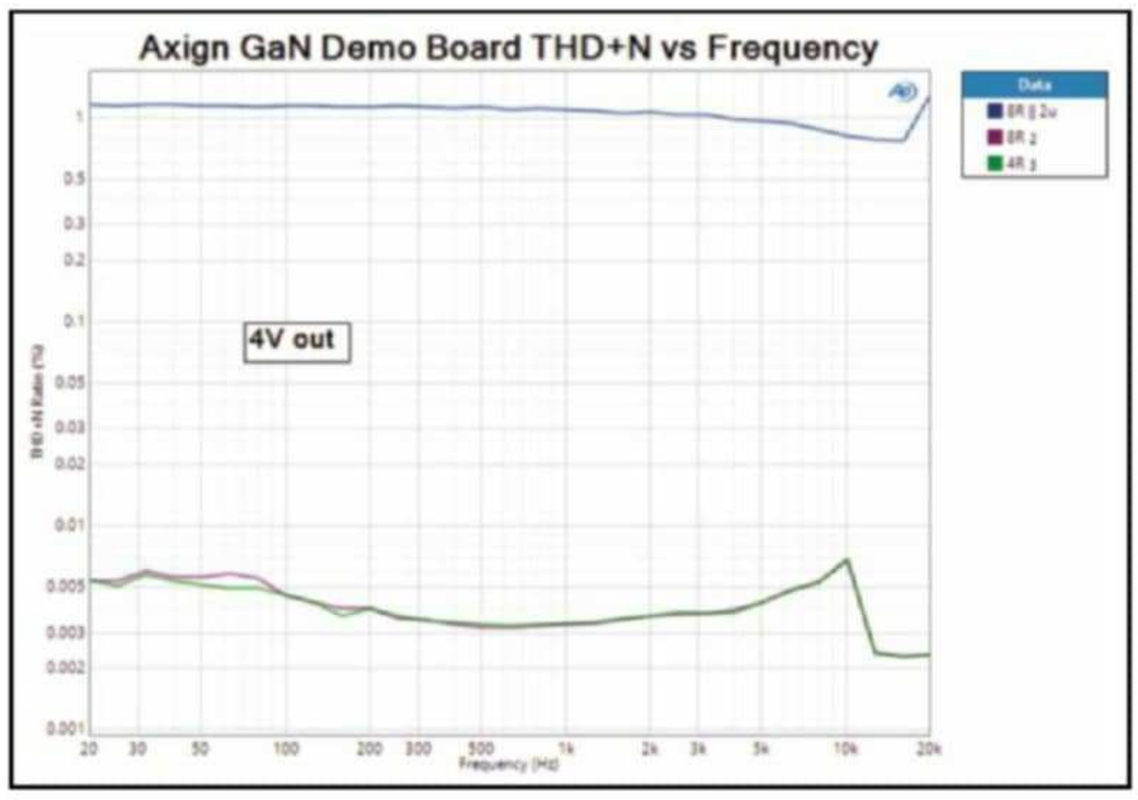


Figure 11: A sweep of distortion versus frequency shows very little high frequency rise and consistent results for 8R and 4R loads.

For the resistive loads, it remains low across the audio band. With the extreme capacitive load, the rise in noise dominates and the result is less good. Nonetheless, with more normal loading, the lack of significant rise at higher frequencies reflects the shaping for the open loop transfer characteristic.

The Axign Demo Board In Use

For the listening portion of the review (my favorite part!), I used the board with two different systems. System 1 has the WiiM Pro streamer as a source, connected via an optical cable, consistent with my view that digital is always more precise and reliable than analog. The Axign amps then drove a pair of Quad 988 electrostatic speakers. For System 2, the Axign board's input was connected to the preamp output of an NAD M10 streamer/integrated at the Axign's analog inputs to drive the upper section of a modified and biamped pair of NHT 3.3 dynamic speakers. The change in source when switching between these two systems was because the WiiM Pro's analog outputs have mediocre performance (this has been claimed to be corrected in the newest version, the WiiM Pro Plus) and because I use the Dirac feature of the M10 with the dynamic speakers but not the ESLs.

Not unexpectedly, the Axign amps functioned flawlessly. There were no turn-on or turn-off pops or noises, the speakers were absolutely silent until the music played, and everything was tight, stable, and audibly in control. The low source impedance ensured that there was no tonal coloration. Whether I played the delightful "Sandman" from Count Basie's *Kansas City 3* on the always-reliable Pablo label, "Funny Ways" from the superb Steven Wilson remaster of Gentle Giant's eponymous debut album, my homemade recording of Lee Barber's "All Night Long," or heavier rockers like Dream Theater's "This Dying Soul" from their *Train of Thought* album on Elektra, the sound was always clear, clean, uncolored, and unstressed.

Swapping the Axign board back and forth with the Nilai DIY amps currently in my system, the sound remained unchanged—a high compliment! Obviously, the Axign ran out of gas before the more powerful Nilai on heavy metal turned up to 11, but recovery from clipping was totally without drama.

Also, a high compliment: During the hot and humid summer here in rural New York, the amp put essentially no heat into the room. I couldn't get the board anywhere near hot even with extended high-power testing.

My only regret is that I didn't have one of those idiosyncratic boutique speakers with really wild impedance dips to show off the low impedance drive capability, but I would have no hesitation pairing a design like this with one of those oddities. And of course, for active speaker use, this would be a non-issue.

Conclusion

The Axign chips used in a reference design deliver good distortion performance, very low noise, excellent carrier suppression, exceptionally low source impedance and drive capability, and thus absolutely no coloration to the sound. It took major effort to make the amplifier misbehave. All this while running absolutely cool with no external heatsinking! The demo board reference design shows the sort of performance possible for amplification integrated into active speakers or for a stand-alone amplifier using the remarkably versatile and efficient Axign controller chips. There's a lot of innovation packed into a few square centimeters.

Author Acknowledgements: I thank Axign and Richard Langezaal for the use of their diagrams as well as patient explanations, and Dimitri Danyuk from Harman for giving me background and user feedback.

Resources

W. Maas, "Axign of Things to Come," *audioXpress*, December 2017, https://audioxpress.com/article/amplifier-series-axign-of-things-to-come

F. Mostert et al., "A 5×80W 0.004% THD+N Automotive Multiphase, Class-D Audio Amplifier with Integrated Low-Latency ΔΣ ADCs for Digitized Feedback after the Output Filter," Session 5, Analog Techniques, 5.1, Institute of Electrical and Electronics Engineers (IEEE), 2017, https://ieeexplore.ieee.org/document/7870273

Daniel Schinkel et al., "A Multiphase Class-D Automotive Audio Amplifier with Integrated Low-Latency ADCs for Digitized Feedback After the Output Filter," *IEEE Journal of Solid-State Circuits*, Volume 52, No. 12, December 2017.

S. Yaniger, "GaN Technology in Audio Power Amplification," *audioXpress*, April 2022.



New Ways to Measure Noise Cancelling, Transparency, and Other Modes in Headphones and Earphones

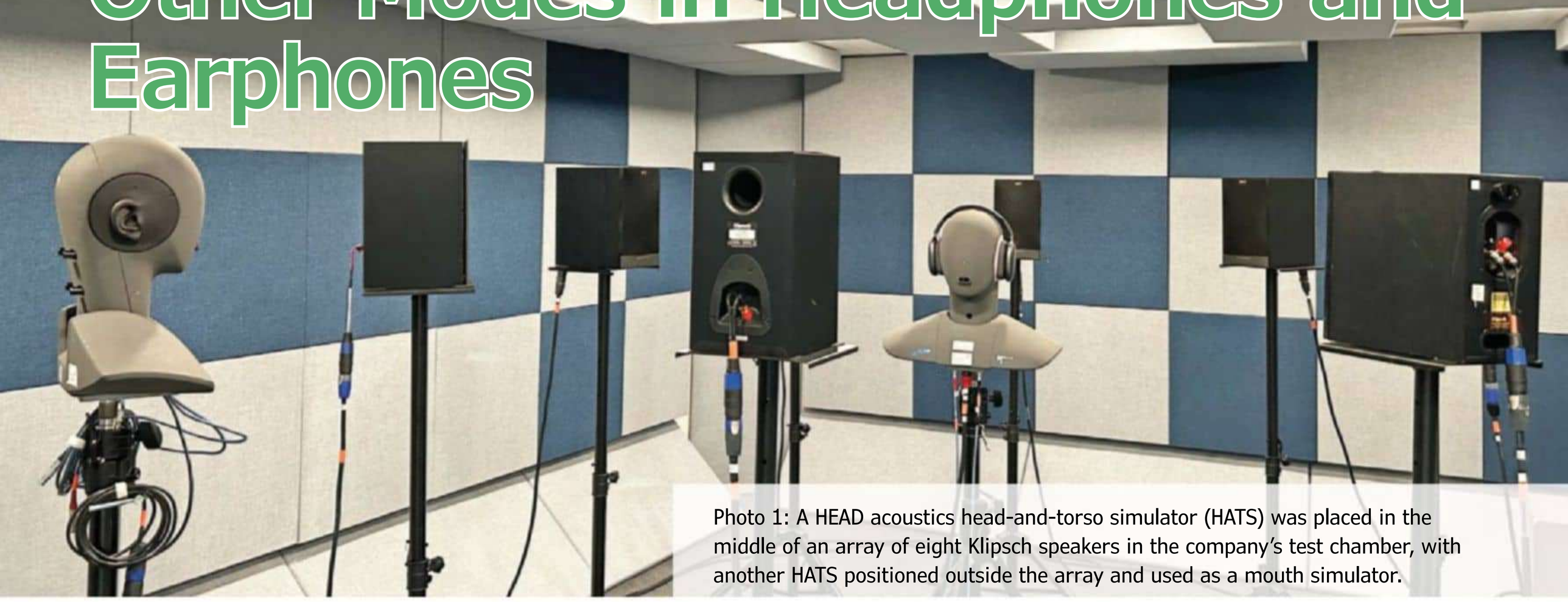


Photo 1: A HEAD acoustics head-and-torso simulator (HATS) was placed in the middle of an array of eight Klipsch speakers in the company's test chamber, with another HATS positioned outside the array and used as a mouth simulator.

By
Brent Butterworth

Now that headphones and earphones are adding situational awareness, hearing protection, and hearing enhancement features, it's time to come up with new test methods to evaluate these functions. HEAD acoustics has some ideas.

Probably the most interesting new function in headphones and earphones emerged in large part due to two recent health-related matters: a growing concern for hearing health and the COVID-19 pandemic. Thanks in part to these events, headphones and earphones are now something we don't just listen to—we also hear through them.

In 2017, the US Congress directed the US Food and Drug Administration (FDA) to develop regulations for over-the-counter hearing aids that could be sold without consultation from an audiologist. Although the FDA didn't release these regulations until 2022, many manufacturers jumped the gun by launching "hearables"—products that weren't sold as hearing aids, but nonetheless were intended to enhance hearing.

Products such as the NuHeara IQbuds2 Max (reviewed in the July 2021 *audioXpress*) were, in essence, true wireless earphones with hearing-assistance features built in. Apple added hearing-assistance functions to its hyper-popular AirPods Pro earphones, and later added a hearing-protection function—a hear-through mode intended to limit any sound reaching the eardrums to 85dB, a level generally accepted to be safe for eight hours of exposure.

Now let's get to the pandemic. Millions of office workers were forced to work from home, where their attempts to focus on spreadsheets or make Zoom calls were often disrupted by noise from other family members. Interest in noise-cancelling headphones rose fast, even though most headphone noise-cancelling systems are designed primarily to reduce airplane cabin noise, not voices or blaring TV sets.

Adaptive noise cancelling—launched most prominently in the Apple AirPods Pro in 2019, and now available in countless headphones and earphones—promised to tune the function of noise-cancelling systems to better suit other environments in addition to airplanes, in some cases blocking low-frequency noise while allowing speech in.

Because of these developments, high-end wireless headphones and earphones have evolved from being mere playback devices into products that promise total control over what enters your ears—reducing unwanted noise and blocking dangerously loud sounds, while enhancing the sounds you do want to hear, such as conversations.

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The Challenge

For engineers and reviewers, though, these new functions present considerable difficulties when it comes time to evaluate their efficacy. The standard method for measuring isolation provided by headphones and earphones—whether through active noise cancelling (ANC) or passive noise blocking—is described in the IEC 60628-7 standard. It employs uncorrelated pink noise played from eight sources surrounding a measurement fixture, such as an ear/cheek simulator or head-and-torso simulator (HATS). The headphones or earphones are placed on the test fixture, and a spectrum analysis is performed to show how and at what frequencies the headphones or earphones reduce the level of sound reaching the fixture's test microphone.

This method worked well in the days when ANC was a simple "on/off" feature. But because the method presents only one stimulus at a fixed level, it offers no way of evaluating how well an adaptive noise cancelling system adjusts itself to optimize performance in different environments. And while this method is somewhat useful in evaluating transparency (or hear-through), voice-enhancement and environmental noise functions, the results are difficult to interpret and not particularly reflective of real-world situations.

"Pink noise is great as a development tool, but it's not realistic," Jacob Soendergaard, account manager at HEAD acoustics, said. "It doesn't simulate user situations, and it doesn't stress the DSP in a realistic way."

I had made some efforts to characterize these functions in my work for Wirecutter, a product-recommendation site, resorting to such seat-of-the-pants techniques as flanking my GRAS 43AG ear-cheek simulator with a pair of PA speakers playing a simulated music clip (crafted to sound like music but produce a pink noise-like spectrum and level consistency) at 105dBA. After seeing a video I made about my testing, Soendergaard told me that HEAD acoustics had worked out a much more formalized version of what I was attempting, and asked if I'd like to visit the company's US headquarters in Brighton, MI, to see what they're doing.

Quantifying the Previously Unquantifiable

Soendergaard explained that in keeping with HEAD acoustics' preference for relying on existing standards, the company based its work on measurement techniques outlined in ETSI TS103 224. The goal was to simulate environments that a consumer might typically encounter when using headphones and earphones equipped with various hearing enhancement and protection features. These can include any combination of ANC (or passive noise blocking), a hear-through function, voice enhancement, and hearing protection. They can also be used to evaluate earplugs and active hearing protection products.

HEAD acoustics recorded five different sound environments using an eight-microphone array, with the intent that these signals be played back on an eight-speaker system in a controlled environment (i.e., a largely absorptive room, but not necessarily an anechoic chamber), with a HEAD acoustics HATS (**Photo 1**) positioned in the middle of the speaker array, and the headphones, earphones or earplugs placed on the HATS. By recording and analyzing the signals coming through the microphones in the HATS, using a HEAD acoustics LabCore hardware interface (**Photo 2**) and ACQUA analysis software (**Photo 3**), the company (and its customers) can gauge how effectively headphones and earphones deal with these sonic environments.

Here are the environments that HEAD acoustics recorded, and the average sound pressure levels (SPLs) of the recordings:

- Electric leaf blower (96.13dBA)
- Riding lawnmower (95.90dBA)
- Woodshop with miter and table saws (98.23dBA)
- Music venue, electronic dance music (91.45dBA)
- Airplane cabin (82.15 dBA)

To gauge how well headphones and earphones allow voices to pass through—so they can be heard when the products are worn, even in noisy environments—HEAD acoustics adds a mouth simulator positioned between two of the front test-signal speakers.

In this case, the mouth simulator is a tiny transducer that reproduces a human voice, with a dispersion pattern and acoustic response that mimic the characteristics of a human mouth. The mouth simulator was built into a HEAD acoustics HATS.



Photo 2: The HEAD acoustics LabCore audio measurement interface

The signals arriving at the test fixture microphones—after being emitted from the eight-speaker array, then filtered through the headphones or earphones—are captured, then analyzed through HEAD acoustics ACQUA software.

The entire process is automated, and typically takes about 2 ½ hours. That's 90 minutes for signal acquisition—including 30 minutes for transparency mode, 30 minutes for noise cancelling on, and 30 minutes for noise cancelling off—followed by 60 minutes for processing. "You wouldn't do this for every iteration of a product design, just for major evaluation steps," Soendergaard said.

The processing computes the performance of the device under test (DUT) for several benchmarks.

Overall attenuation in decibels: a single, unweighted number to show how much the product attenuates the sound level at the eardrum, versus having no headset in place. This is the most basic measurement of headphone and earphone isolation, but because it doesn't take psychoacoustics or the content of environmental noise into consideration, it doesn't accurately reflect the level of attenuation that listeners will perceive.

Spectrum of attenuation: a plot to show the attenuation at different frequencies. This is the graph that most of the reviewers who measure headphone and earphone isolation (including me) usually present. If you have some knowledge of psychoacoustics and the spectrum of common types of environmental noise, you can use these charts to get a rough idea of how much isolation listeners will experience when they use the product, but the result won't allow quick, easy comparison of the perceptual performance of two devices under test, or two iterations of DSP.

Attenuation vs. time: measurement shows device performance over time to track attack times. This is useful for testing adaptive noise cancelling functions, to see how quickly and effectively they respond to changing circumstances.

LAeq: A-weighted noise level over a certain span of time, a measure commonly used by regulatory agencies (such as US Occupational Safety and Health Administration, or OSHA) to gauge noise exposure in workplaces, concerts, and more.

Loudness level reduction: applies psychoacoustic principles to noise levels at the ears to gauge the subjective noise reduction experience the DUT will produce. It's the same as overall attenuation in decibels but weighted on a subjective scale and expressed in phons. For example, a 20dB reduction in noise at 1kHz will be much more noticeable than the same reduction at 40Hz or 18kHz, because the ear is less sensitive at those frequencies.

Speech Intelligibility Index (SII): analyzes the noise arriving at the eardrum to estimate how difficult it would be to understand voices in a specific environment while using the DUT. The result in this test is the estimated percentage of speech that will be intelligible—100 being perfect intelligibility, 0 being a complete inability to understand speech. SII is especially appropriate for measuring the efficacy of the transparency (or hear-through) modes found in many noise-cancelling headphones and earphones, which are intended to allow the wearer to participate in a conversation without removing their headphones. However, SII wasn't designed to measure speech intelligibility in a noisy environment.

Listening effort: similar to SII, this test (which HEAD acoustics developed, and is now standardized as ETSI TS103 558) analyzes the speech signal arriving at the eardrum to estimate how difficult it would be to understand speech in specific situations while using the DUT. Like SII, this test is especially valuable in evaluating the efficacy of transparency modes. But it's specifically designed for measuring speech intelligibility in a noisy environment. A rating of 5 indicates no effort, similar to conversing with someone at a range of 1 meter in a quiet

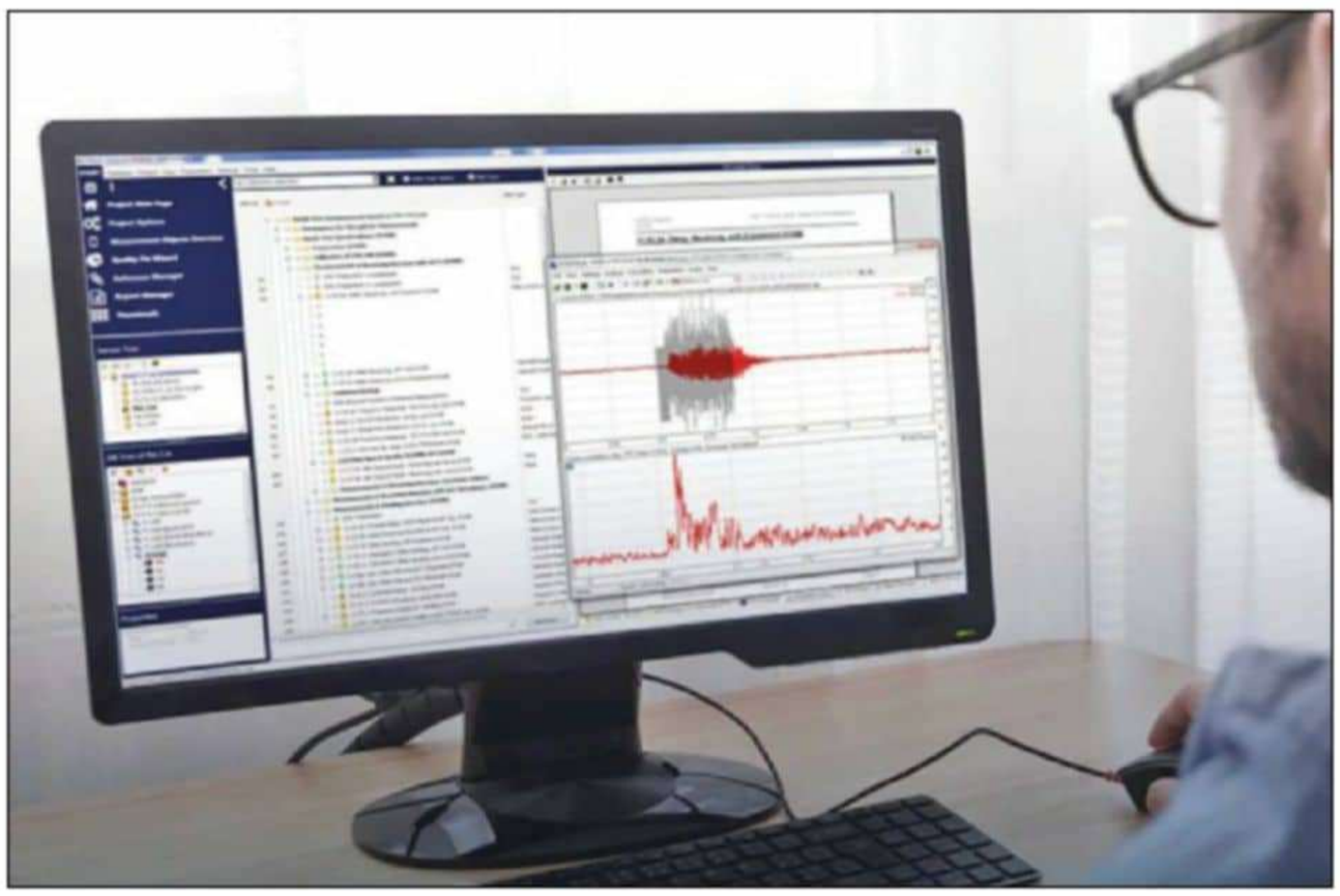


Photo 3: HEAD acoustics ACQUA audio analysis software

About the Author



Brent Butterworth has been a professional audio journalist since 1989, and has measured and evaluated thousands of audio products. His headphone measurements and reviews appear on soundstage.com and wirecutter.com. He has served as editor-in-chief of *Home Theater* and *Home Entertainment* magazines, contributing technical editor for *Sound & Vision* magazine, senior editor of *Video* magazine, and reviews editor of *Windows Sources* magazine. He has also worked as marketing director for Dolby Laboratories, and as a consultant on the design and tuning of speakers, subwoofers, and headphones.

environment, while a rating of 1 indicates a total inability to comprehend speech in a given situation.

The Results

During my visit, we measured four products, a process that consumed the better part of a day. The products were the Bose Noise Cancelling Headphones 700 (which include noise cancelling and transparency modes); Apple AirPods Pro 2 (which include both modes and a hearing protection function); the Loop Experience earplugs; and a generic set of foam earplugs.

To get an impression of what was going on in the tests, and what the program material sounded like, I entered the sealed test chamber wearing the Apple AirPods Pro 2 earphones (**Photo 4**) set to the transparency mode, which would, in theory, limit the sound reaching my ears to 85dB SPL, even though some of the test material reached levels as high as 98dB SPL. Although the loud volumes weren't pleasant, I was delighted to hear, when I removed the earphones, how natural the simulations sounded from the eight-speaker array when I was standing directly behind the HATS.



Photo 4: The author in the HEAD acoustics test chamber experiencing the environmental noise simulations while wearing the Apple AirPods Pro 2 earphones.

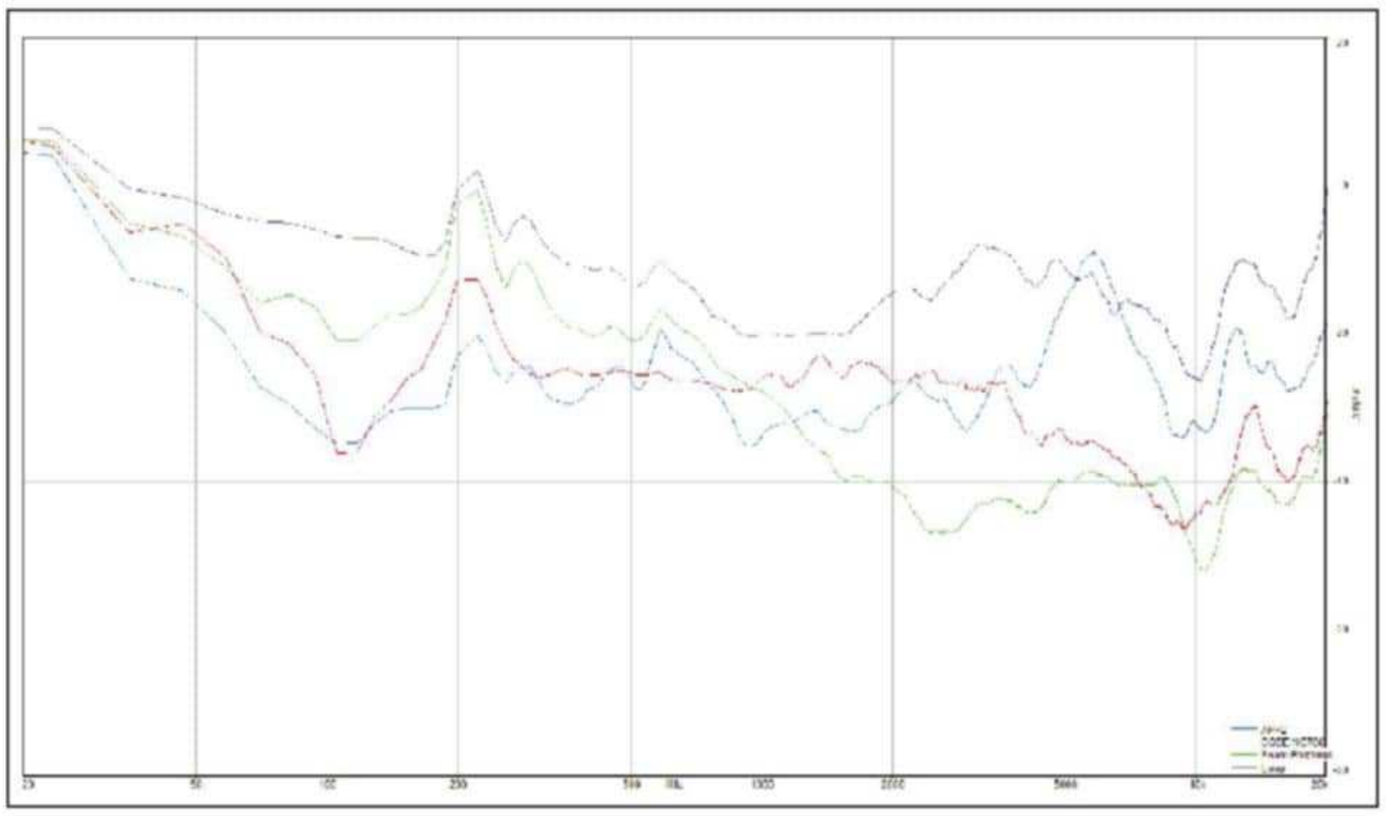


Figure 1: Noise reduction/cancellation provided by the Apple AirPods Pro 2 (blue), Bose NC700 (red), Loop Experience earplugs (purple) and generic foam earplugs (green), in the presence of simulated airplane cabin noise at 82.15dB SPL average. The 0dB line represents the noise level with the ears on the head-and-torso simulator unoccluded.

Soendergaard later provided me with a document showing the results of all the tests we'd done. The results included a matrix chart of single-number ratings for each DUT in specific conditions, and charts Soendergaard put together comparing the performance of the devices at different audio frequencies, time windows, noise levels, etc.

Just a glance at the matrix chart made it easy to see the differences in performance among these products in the various test settings. For example, in the overall attenuation test, the foam earplugs easily beat the Bose headphones, Apple earphones, and Loop earplugs in the leaf blower and workshop simulations. In the leaf blower test, for instance, they reduced outside noise by 41.57dB, while the other products achieved at best a 32.72dB reduction (for the Bose headphones).

Yet in tests where the environmental noise had more low-frequency content (e.g., airplane cabin noise and a music venue), the noise-cancelling-equipped Bose and Apple products did far better—because passive devices such as the foam plugs generally do a poor job of blocking low frequencies. The Apple AirPods Pro 2 performed best on both tests, achieving a noise reduction of about 25dB, while the foam plugs reduced the noise by only about 11dB. Having tested many earplugs and noise-cancelling headphones, these results didn't surprise me.

What was more interesting to me, though, was the listening effort tests, particularly in the case of the Apple and Bose products. In this case, the listening effort in the presence of noise with no headphone or earphone installed on the HATS was rated at 2.5 on a scale of 1 to 5—1 representing total inability to comprehend speech, and 5 effectively representing no listening effort. In noise cancelling mode, the AirPods Pro 2 did not lessen listening effort, but neither did they make it worse: the result was still 2.5. However, in transparency mode, which is intended to reduce listening effort by allowing the voice to "leak through" the earphones, the result was 2.3—slightly worse than achieved with noise cancelling, and more importantly, worse even than with the earphones removed. The results with the Bose NC700 were 2.1 with noise cancelling on, 2.4 in transparency mode, so with the Bose product, the

transparency mode does seem to work as intended, although the result in transparency mode still isn't as good as the result with no headphones at all.

These results show how valuable this type of testing is. Transparency modes are intended to make speech at least as intelligible as it would be with no headphones or earphones on. Given the amount of DSP in noise-cancelling headphones and earphones, it seems that transparency modes might even be able to enhance speech relative to the results with an unoccluded ear canal. If speech intelligibility in transparency mode is worse than it would be when no headphones or earphones are used, it's likely that consumers will take off the products while they're conversing with others, thus negating the benefit of a transparency mode.

The charts allowed more analysis that shows the "why" that explains the numerical results. In the chart showing the noise reduction in the presence of simulated airplane cabin noise (**Figure 1**), for instance, we can see that the noise-cancelling products—the Apple AirPods Pro 2 and the Bose NC700—provide excellent noise cancelling at low frequencies (where the rumbling drone of jet engine noise occurs). The foam earplugs do the best job of blocking high-frequency sounds (e.g., the hiss of airplane HVAC systems), but the Bose NC700's excellent passive isolation gets the headphones pretty close to the high-frequency isolation of the earplugs.

The next chart (**Figure 2**) shows the results of the same test with the simulated noise from an electric leaf blower. Here the noise is dominated by high frequencies, and the foam earplugs clearly work best, followed by the Bose NC700 headphones.

The growth of adaptive noise cancelling systems, which automatically adjust their function to suit different sonic environments, has proved particularly challenging for those of us doing headphone measurements, because we must guess what environmental characteristics the headphone designers anticipated, and guess again at how best to simulate them. Thus, I was especially curious about the HEAD acoustics system's ability to do dynamic measurements—testing a headphone's isolation in changing conditions.

You can see some of the results in the third chart (**Figure 3**), which shows the measured level of the DUTs at the eardrum (or measurement microphone) in various levels of simulated live-concert sound. Except for the Apple AirPods Pro 2, all the devices respond in a linear fashion—the level at the eardrum increases directly with the level of the outside noise. With the AirPods Pro 2, set for transparency mode, there's a decreasing rise of level at the eardrum as the external noise rises above about 85dB SPL.

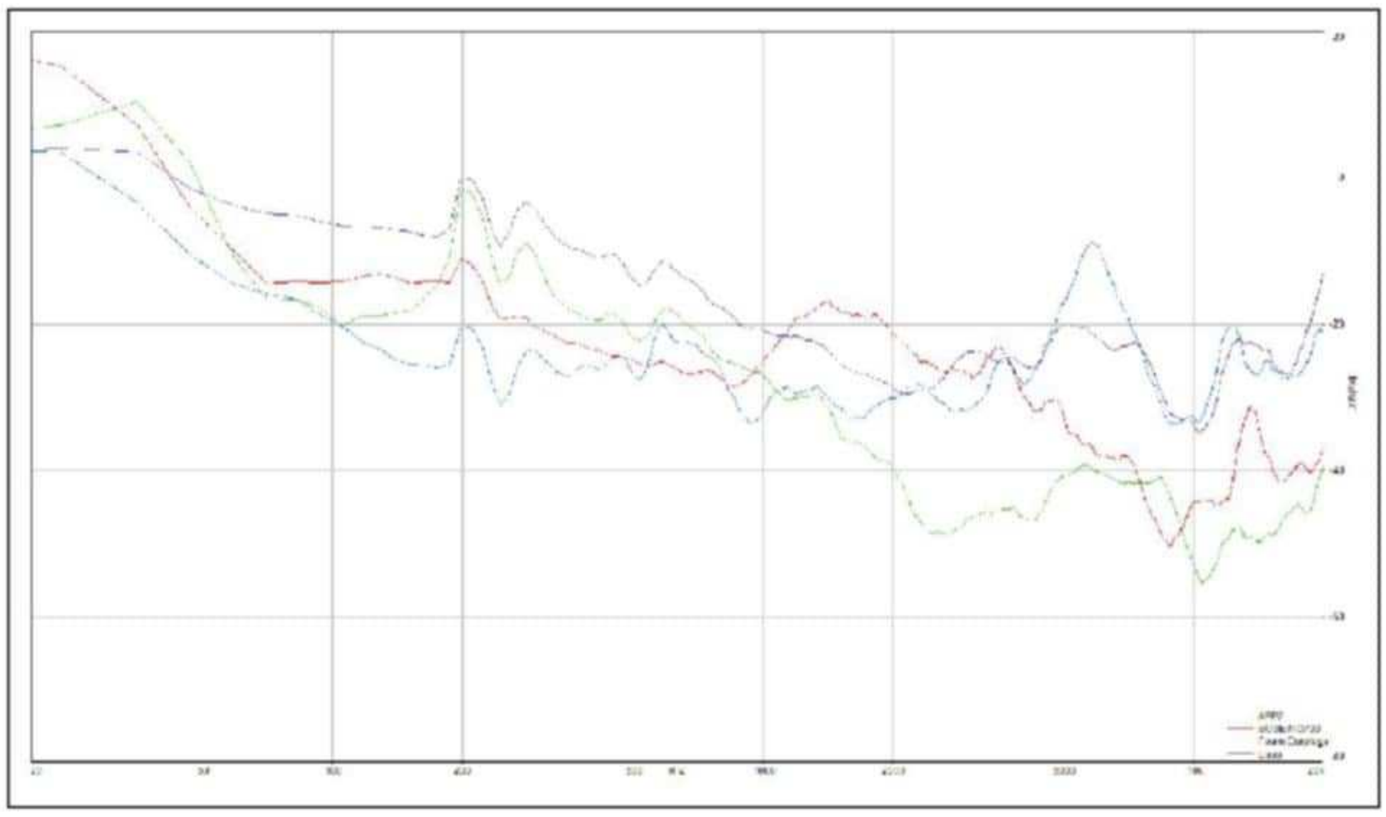


Figure 2: Noise reduction/cancellation provided by the Apple AirPods Pro 2 (blue), Bose NC700 (red), Loop Experience earplugs (purple) and generic foam earplugs (green), in the presence of simulated electric leaf blower noise at 96.13dB SPL average.

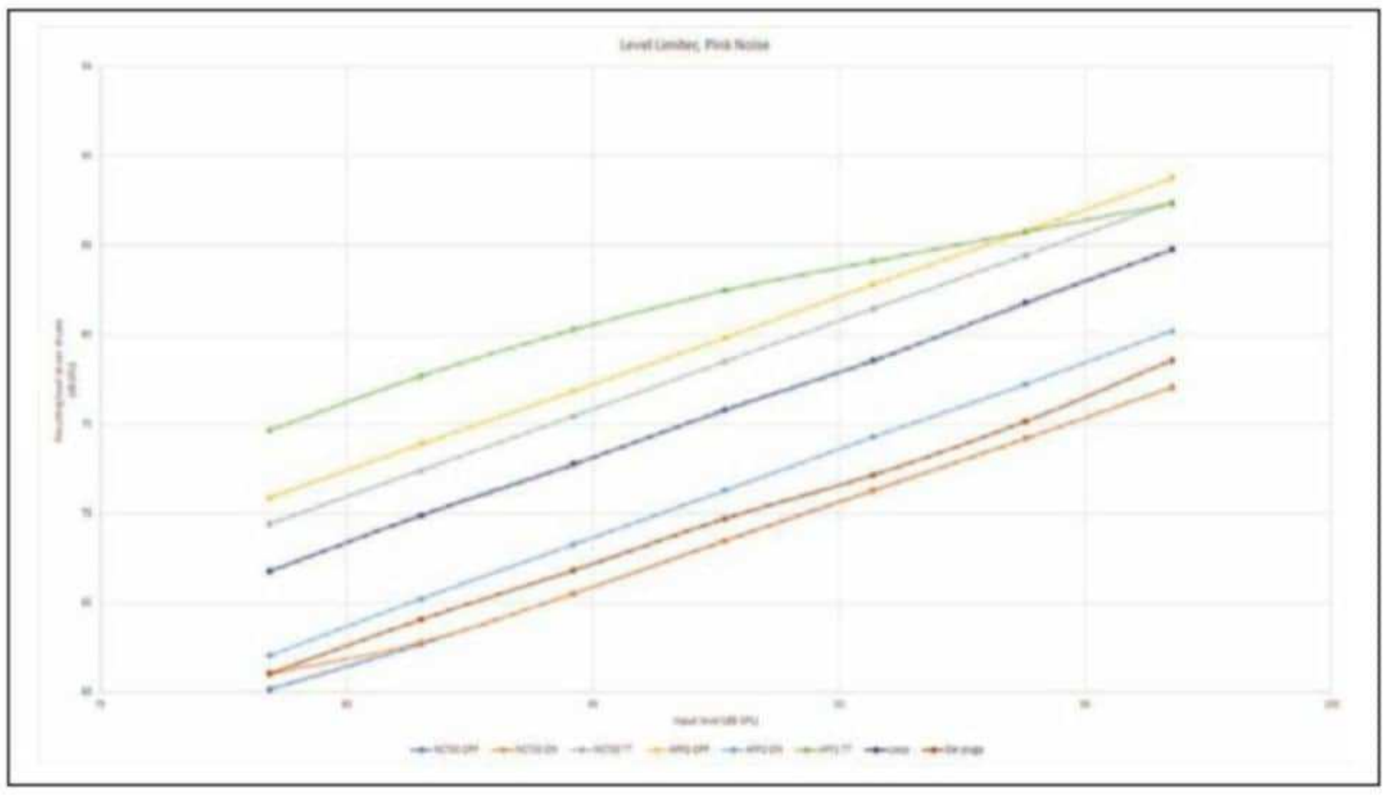


Figure 3: Level measured at the eardrum (or measurement mic) versus the level of external noise for the Apple AirPods Pro 2 (with noise cancelling on and off, and in transparency mode) and Bose N700NC (with noise cancelling on and off), and Loop Experience and generic foam earplugs.

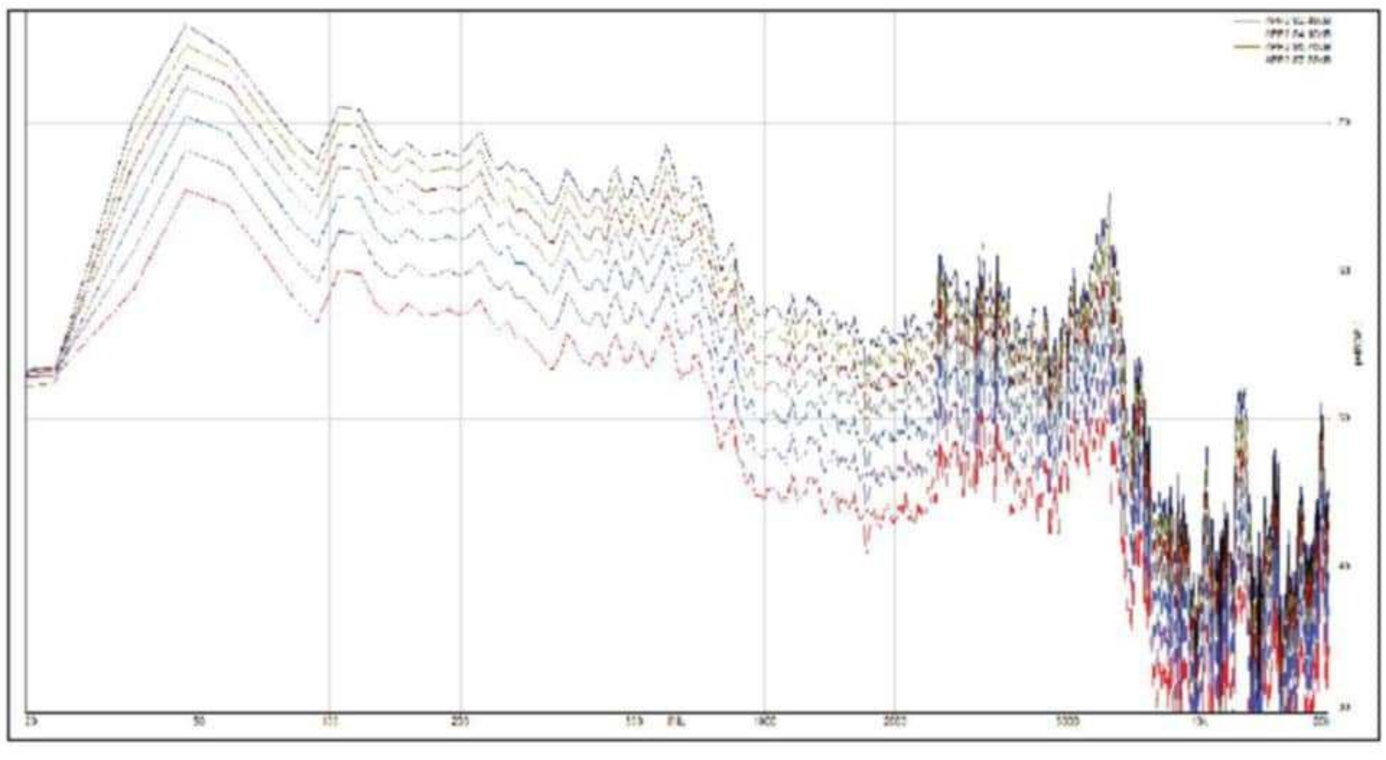


Figure 4: Isolation of Apple AirPods Pro 2 in the presence of external noise ranging from 74.65dB SPL to 87.32dB SPL.

Apple has stated that the transparency mode limits the level at the eardrum to 85dB SPL; this test suggests that the transparency mode is working in a manner similar to what Apple has described, although the level did rise to about 87.5dB once the external sound level rose to 97dB. Interestingly, the AirPods Pro 2 reduced the level at the eardrum much more in noise cancelling mode, but the reduction is linear and does not change as the external sound level increases.

The final chart (**Figure 4**) extends this dynamic measurement concept by examining the spectrum of the AirPods Pro 2's noise cancelling and reduction in different levels of external sound. As you can see, the isolation provided by the earphones remains consistent at all frequencies no matter how loud the external sound gets. This suggests that the adaptive function of the AirPods Pro 2's adaptive noise cancelling adjusts only the level of the noise cancelling, not the frequency profile. (Although who knows what new functions will arrive in Apple's next update?)

Conclusion

Although we spent nearly an entire day doing measurements, I feel like we just scratched the surface of what the HEAD acoustics measurement system can do. Of course, I'm not sure what kinds of functions headphone manufacturers will add to their products in the coming years, or how their designers will employ DSP to add new adaptive functions to these products, but I'm pretty confident that this system, and the measurement standards it's based on, will be a big help in creating and optimizing new smart and adaptive functions in headphones, earphones, hearing aids, and hearing protectors.

HEAD acoustics MDAQS

A More Comprehensive Concept for Audio Measurement

In my audio product consulting work, many clients have asked me to create a testing protocol that would give them clear, straightforward sound quality ratings of their products—ratings that would tell them how they stack up against the competition, whether an engineering change produced a notable improvement, and whether a cost-reducing alteration had a perceptible effect on performance. Although I was able to steer them toward some useful measurements, I always recommended they assemble listening panels that could provide a more comprehensive assessment of a product's sound quality—especially in the case of low-cost active products, which often employ dynamic processing and frequency response shaping to wring every last bit of performance out of their tiny drivers and inexpensive monolithic amplifiers, and thus can't be fully evaluated through conventional frequency response measurements.

"Frequency response measurements are a little bit like zero to 60 times for a vehicle," HEAD acoustics' Jacob Soendergaard told me. "They'll tell you something about the performance but will not provide a holistic perspective on the quality of the vehicle."

However, assembling listening panels is time-consuming, and finding people who want to sit through frequent, extensive, and tedious testing sessions is difficult. None of my clients ended up taking my advice, preferring instead to trust a limited series of measurements plus my opinions and perhaps the opinions of an in-house expert or two. Sometimes this method worked, sometimes it didn't, but it always took a lot of time and produced imprecise results.

To make sound quality assessments faster, more precise and more liable, HEAD acoustics created the Multi-Dimensional Audio Quality Score (MDAQS). It's an effort to provide a technically exhaustive yet easy-to-grasp assessment of the quality of practically any audio playback system, be it a wireless speaker, a set of headphones or even a car audio system. MDAQS is an optional feature in HEAD acoustics ACQUA audio analyzer software, activated through the purchase of the ACOPT 36 extension. Surprisingly—but conveniently—it uses the same measurement hardware and software for all applications, based around a HEAD acoustics head-and-torso simulator (HATS). The advantage of using the HATS is that it provides binaural results that can be used to gauge stereo immersion, something not possible with a standard measurement microphone.

The basis of MDAQS is a series of tests HEAD acoustics conducted with more than 200 participants, who performed an extensive series of blind A/B preference comparisons between audio signals that had been processed in different ways, using a quality scale of 1 to 5. The tests revealed that characterizing timbre, distortion, and immersiveness could provide a simple, comprehensive, and accurate assessment of an audio system's subjective quality.

HEAD acoustics' engineers then developed a set of measurements that could evaluate DUTs on the basis of these three parameters. The measurements employ a combination of music clips and swept tones, and they're part of an automated routine that runs on the ACQUA system.

For every product tested, the end result is a series of four mean opinion score (MOS) ratings for a product: MOST (for timbre), MOSD (for distortion), MOSI (for immersion), and MOSO (for overall). All these scores are expressed as a single number extended to one decimal point, with 1.0 being bad and 5.0 being excellent. By looking at these four numbers—rather than a chart full of numbers—engineers can easily compare different iterations of a product, compare past products to new designs, benchmark their products against competitors, and figure out which of a products' different parameters they need to adjust to get better results. Although I haven't used MDAQS, and thus can't confirm its efficacy and accuracy, it does seem to be the sort of "measurement holy grail" that many audio manufacturers have been seeking.

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Conference Room Acoustics

Architects, interior designers, and facility owners now have interesting options for conference rooms, including materials to use in glass surfaces, that don't compromise the acoustic performance.

By
Richard Honeycutt

Photo 1: This simple conference room is typical of late-20th-century designs.

Many commercial and educational buildings incorporate one or more conference rooms. The first conference rooms I ever saw were rectangular and had wood-paneled or gypsum-board walls and mineral-fiber tile ceilings. Sometimes the tiled ceiling area was recessed above soffits in which HVAC ducts were concealed. Centered in the room was a large table surrounded by chairs. The floors were usually carpeted, and no sound-reinforcement system was installed (**Photo 1**).

Over the years, the architectural approach to conference room design has changed. Today, although rectangular shapes prevail, circular, octagonal, or other shapes may be employed. Such designs carry the likelihood of acoustical focusing, a potential cause of echoes that can be annoying at the least and can impair speech intelligibility. While wood-paneled walls are still common, the last decades have seen an increase in the use of glass walls for conference rooms. Ceilings may be acoustically reflective and are sometimes shaped to reflect sound from participants back to the other participants, either using flat wood-paneled or molded gypsum or plastic clouds. **Photo 2** shows a conference room using acoustically reflective/diffusive clouds. **Photo 3** illustrates a circular conference room with glass walls. **Photo 4** shows a glass side wall and part of the front wall in a conference designed to help participants feel "in touch with nature." The rear wall of this room and the central portion of the front wall are wood-paneled.

The conference room shown in **Photo 5** is the most acoustically challenging of the rectangular conference rooms shown so far, because not only does it have a substantial fraction of the area of each wall glassed, but the reflective wood ceiling and the thin carpet-over-concrete floor add little acoustical absorption.



Photo 2: This 21st-century conference room incorporates curved clouds above the table to reflect and diffuse sound from participants' voices.



Photo 3: A circular shape combined with glass walls allows a view of the adjacent production area from this industrial conference room.

Desired Conference Room Acoustical Performance

All conference rooms need to have a low level of background noise: a level of NC 35-40 is recommended (corresponding to a noise level of 40dBA to 45dBA). This noise level requires proper HVAC design and good isolation from noise sources external to the room. In some cases, speech privacy for the proceedings inside the conference room requires even better isolation than would be needed for noise reasons alone. Common reasons for excess external noise include large areas of single-pane glass, hollow-core doors or even solid-core doors with poor seals, and

partition walls extending only to the ceiling, leaving flanking paths for external noise to enter above the walls and down through the conference room ceiling.

Slap echoes in rooms having any dimension greater than about 56′, or flutter echoes in smaller rooms, can decrease speech intelligibility. So can excessive reverberation. **Figure 1** shows the recommended RT plotted vs. the room's cubic volume for a well-designed conference room.

Design Example

As an example of the effects of design variables upon conference room acoustical performance, I built an EASE model of the room shown in Photo 5. You will note from the photo that the room has a (mostly) glass wall at the front and that about



Photo 4: Glass walls add an outdoor ambiance to this rectangular conference room.



Photo 5: This rectangular room incorporates both wood and glass wall materials, plus a reflective wood cloud over the table.

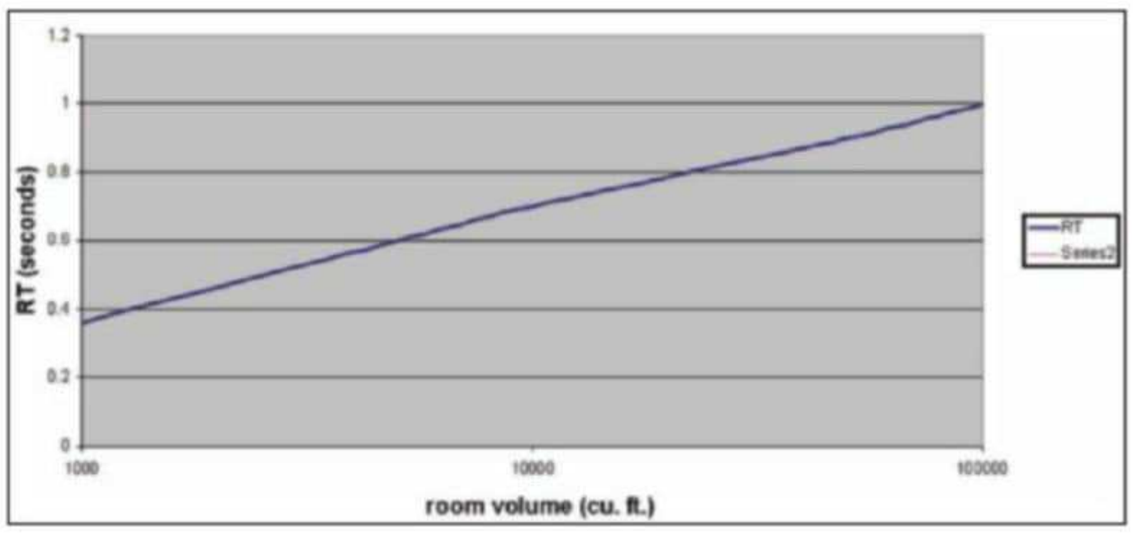


Figure 1: The recommended RT for a conference room depends somewhat upon the cubic volume of the room.

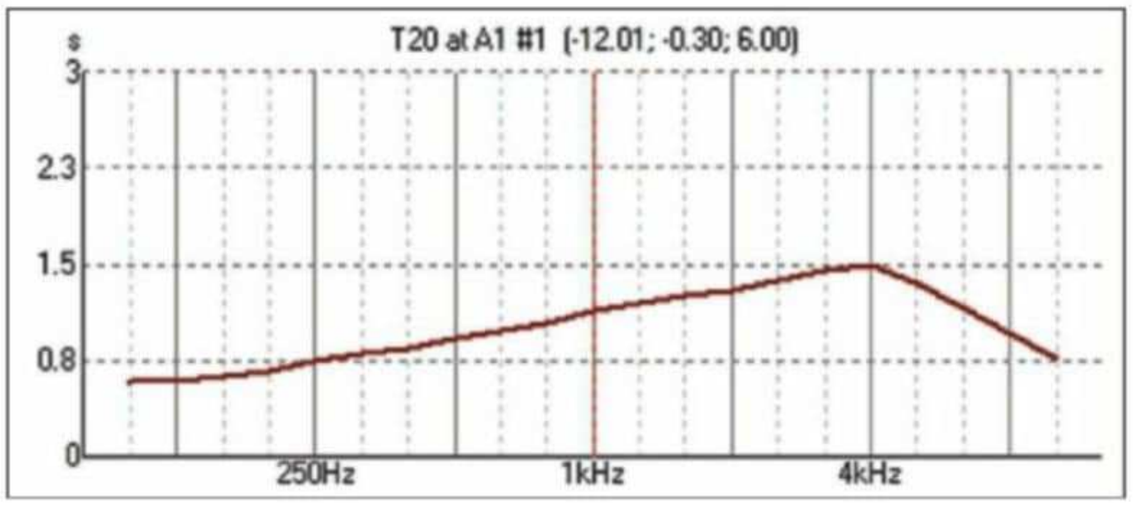


Figure 2: The average RT peaks at 4kHz.

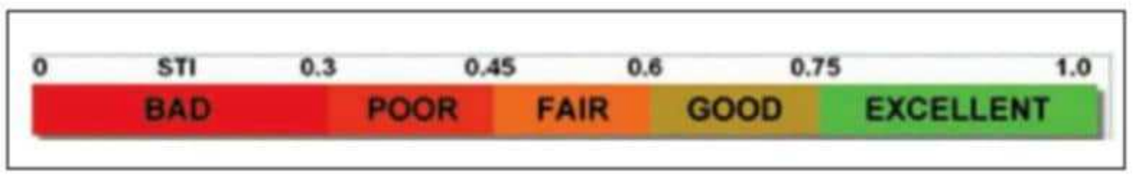


Figure 3: The STI ratings quantify speech intelligibility.

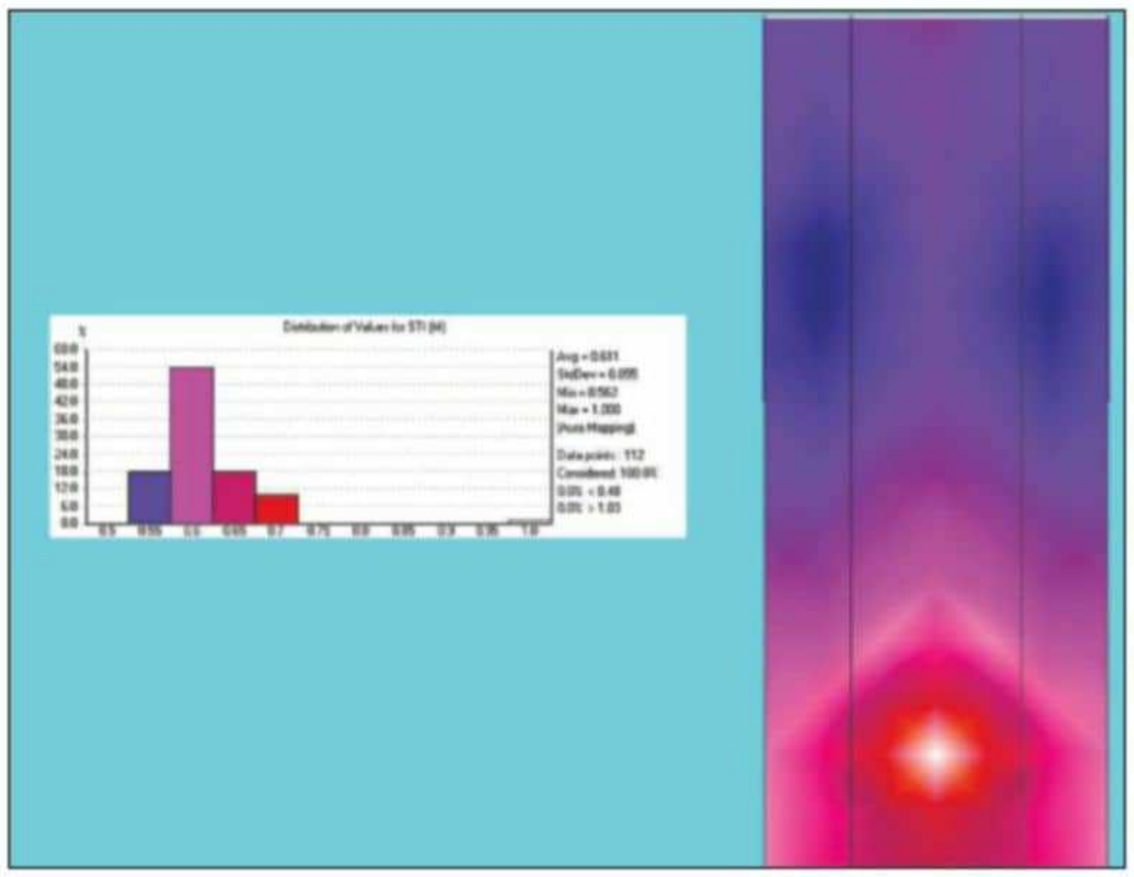


Figure 4: The STI of the conference room as originally designed is about 0.6.



Photo 6: This image shows a view through a Clearsorber film-treated window into an adjacent room.

one-third of the side wall area is glass. Not visible in the photo is the rear wall, also mostly glass. Almost all the remaining wall area is reflective wood. Thus, there are a lot of parallel reflective surfaces. The owner requested a marble floor, although as you can see, thin carpet over concrete was the final choice. My model includes the originally planned marble floor.

Room dimensions are 24'×60'×13', except that the wooden cloud over the table has a height of 11'. Cubic volume is 15,780ft^3. The target RT for face-to-face conferences is 0.6 seconds. However, teleconferencing requires a lower RT for optimal conditions. The reason is that the human ear/brain system automatically evaluates the static acoustical conditions of the environment and cancels these from the perceived sound stream. The listener in a teleconference experiences this effect in regard to his/her listening room. However, if the sound from the talker contains significant acoustical cues about the talker's room, the listener's ear/brain system is confused—trying to eliminate two different rooms' acoustical signatures at the same time. This is distracting for the listener and can be annoying. So not only is a lower RT necessary for acoustical comfort during teleconferencing, but it improves the performance of the participants.

In the case of this room, the optimal RT for teleconferencing would be 0.3-0.4 seconds. **Figure 2** shows that the RT, which is about 1 second at 1kHz, peaks at 1.5 seconds near 4kHz. This would make speech sound somewhat harsh or metallic in the room and would decrease speech intelligibility.

The most straightforward metric for evaluating the acoustical design of a conference room is the Speech Transmission Index (STI). **Figure 3** shows the perceptual ratings of speech intelligibility associated with specific STI numbers. As you can see from **Figure 4**, speech intelligibility is fair to good. (The person talking is located at the bottom of the STI map; hence, the better STI values in that area.) For hearing-challenged conferees, especially those using hearing aids, understanding speech in this room may be a challenge. As-is, the room is not ideal for teleconferencing.



Figure 6: The addition of one layer of Clearsorber Foil improves the RT.

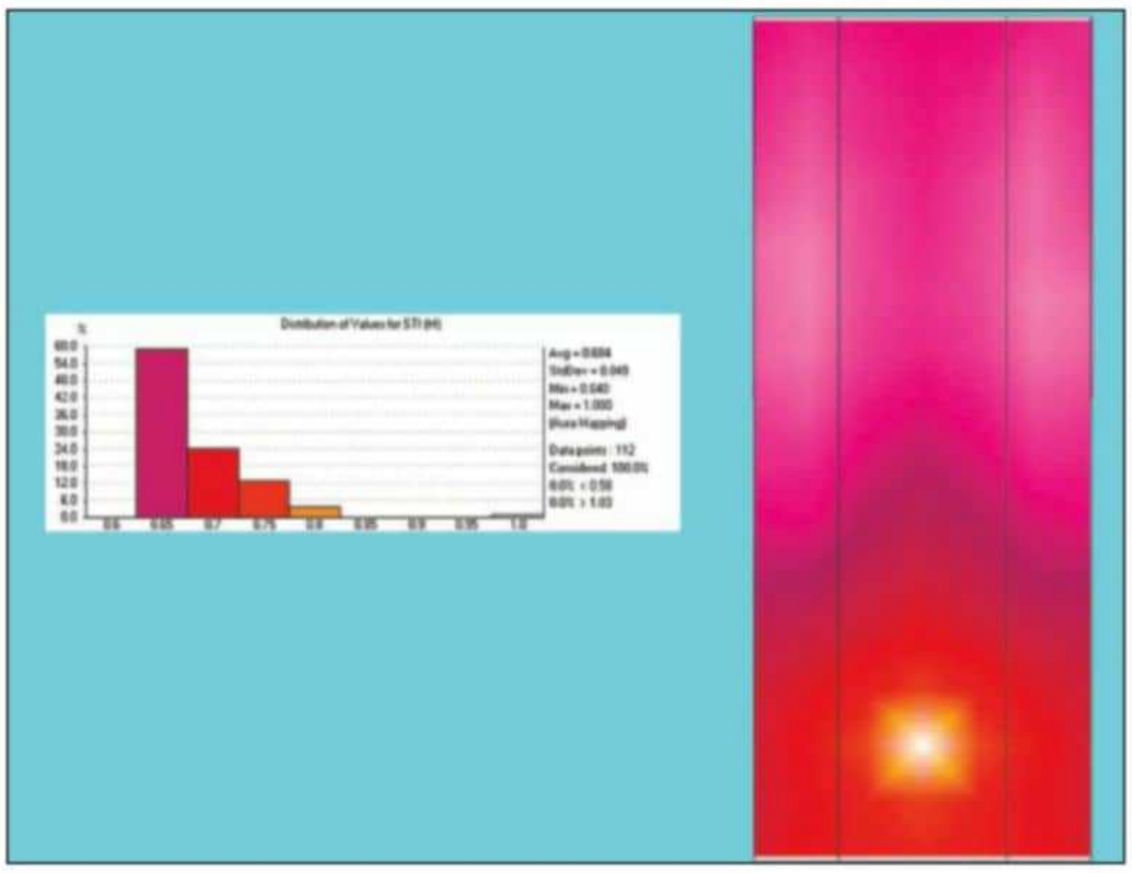


Figure 7: Adding a single layer of Clearsorber Foil improves the STI.



Photo 7: Clearsorber Foil is a transparent, acoustically absorptive material.

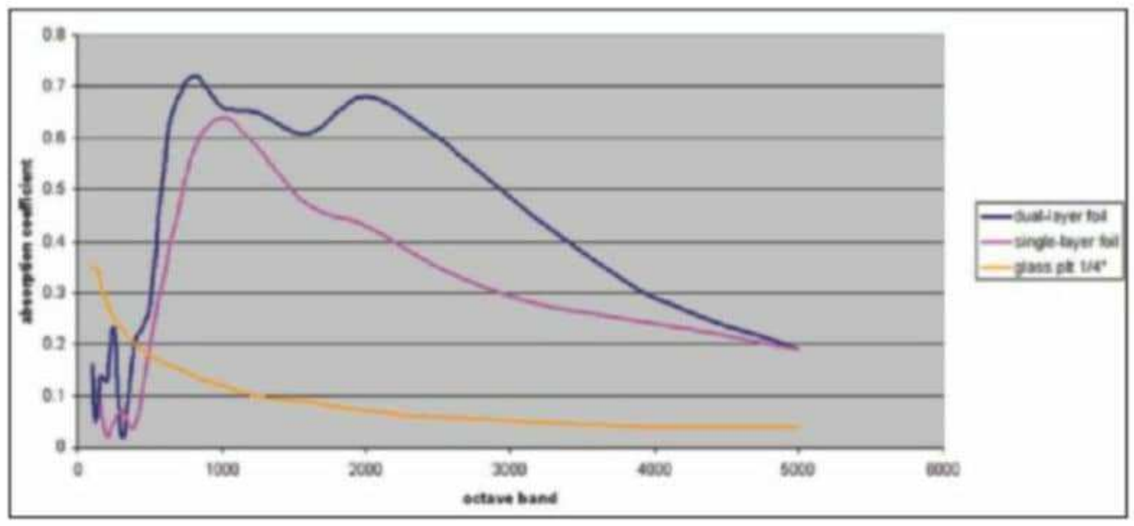


Figure 5: Both single-layer and dual-layer Clearsorber provide superior sound absorption, compared to plate glass.



Photo 8: Armstrong Woodworks micro-perforated material provides excellent sound absorption and an elegant look.

Solutions for Glass-Walled Conference Rooms

Traditionally used materials for adding acoustical absorption to reduce RT and improve STI include felt, fiberglass, acoustical plaster, or mineral fiber ceiling tiles. None of these materials provide the transparency of glass or the elegance of wood panels. However, RPG distributes a product manufactured by Acoustic GRG Products of the UK: a clear acoustically absorptive material named Clearsorber (www.rpgacoustic.com/data-sheet-library), a microperforated polycarbonate or acrylic panel or film.

Photo 6 shows a view through an office's Clearsorber-paneled window. The Clearsorber product line includes (2) transparent solutions:

- Clearsorber Foil—a thin, crystal clear micro-perforated film that is either mounted in a fixed manner off a glass surface (such as to the surrounding mullions) or it can be mounted in a roller blind (manual, remote or hardwired) for an operable solution.
- Clearsorber Panel—a 5mm thick , crystal clear micro-sliced resin panel is mounted in a fixed manner off a glass surface typically with a standoff.

The Foil is the clearest, as the perforations are round, and the pattern is not too dense (**Photo 7**). The Panel material has similar clarity, however the slices used to get best efficiency through a thicker panel create areas that are more translucent than transparent. **Figure 5** compares the octave-band absorption coefficients of 1/4" plate glass with dual-layer and single-layer Clearsorber Foil spaced 2" away from the glass.

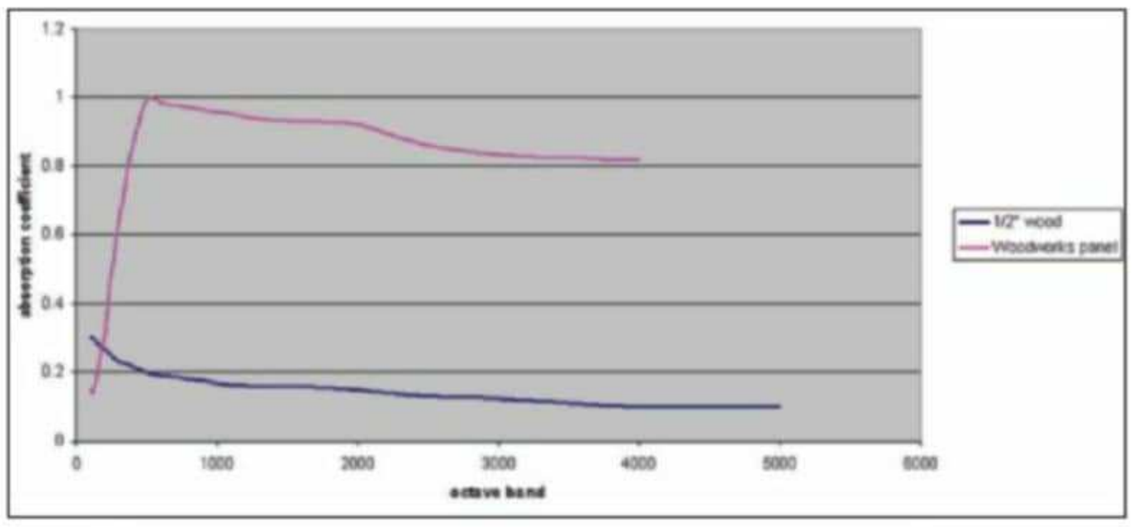


Figure 8: The acoustical absorption of Woodworks micro-perf panels far exceeds that of plain wooden panels.

Applying one layer of Clearsorber Foil to each of the glass surfaces in the conference room eliminates the 4kHz peak in RT and decreases the 1kHz RT to 0.5 seconds, as shown in **Figure 6**. **Figure 7** illustrates the improvement in STI, which has been increased to 0.65 (solidly in the "good" range.)

Further improvement in this room's acoustical performance is needed if it is to serve as a good teleconferencing space. This improvement can be achieved by replacing the wooden panels with Armstrong Woodworks micro-perforated panels (**Photo 8**). The acoustical absorption of Woodworks 5474WW7NDC panels over a 1" fiberglass backer is compared with that of 1/2" non-perf wood panels (**Figure 8**).

By replacing the wooden side wall panels with Woodworks micro-perf panels, we can improve both the RT and the STI of this conference room to meet teleconferencing standards (**Figure 9** and **Figure 10**).

Summary

We have seen how visual elements preferred by architects, interior designers, and facility owners can limit the acoustical performance of a conference room, especially in the crucial area of speech intelligibility. Using modern microperforated materials is one way to maintain the desired visual design while optimizing acoustics.



Figure 9: The increased sound absorption of Woodworks panels decreases the 1kHz RT to 0.37 seconds.

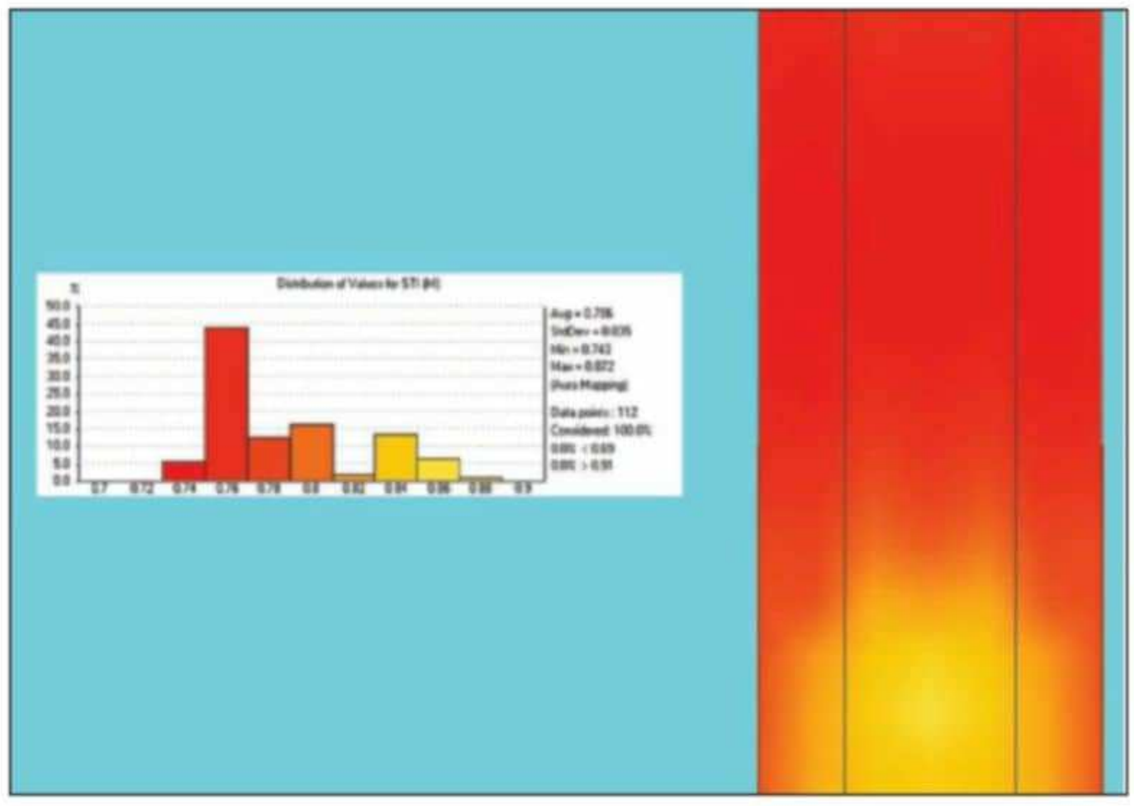


Figure 10: The combined effects of Clearsorber Foil and Woodworks panels increase the STI to 0.76—in the "excellent" range

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.

Optimizing Variable-Frequency Tone Controls

Say what you like, but I think bass and treble tone controls are essential in a hi-fi system, and I will go further to say that variable–frequency bass and treble tone controls are very desirable because of their much greater flexibility. This article discusses optimizing a variable high-frequency and low-frequency equalizer.

By
Douglas Self

In the design considered here, in each band the frequency at which control begins is variable over a 16 to 1 range, making it much more useful for correcting speaker deficiencies or room anomalies. Very few manufacturers have offered this. The only examples that come to mind are the Yamaha preamps; the C6 (1980-81), the C70 (1982-83), the C80 (1984), and the C85 (1986-88)—all of which had low and high frequencies variable over wide ranges (they could overlap in the middle) and Q controls for each band as well, as the tone controls were of the peaking rather than shelving kind, rather like the parametric EQ in a mixing console. These preamps today change hands for startling sums of money.

In 1996 I published a design for what is commonly known as the Precision Preamplifier in *Electronics World* [1], and one of its main features was a treble/bass tone control with the frequencies variable over a 10:1 range. When I mentioned on the DIYaudio board in May 2012 [2] that I had developed an improved version of this tone-control, I was surprised at the enthusiasm for its immediate publication. It duly appeared in Jan Didden's *Linear Audio* Volume 5 [3].

The tone-control was quite sophisticated; apart from variable frequencies, it also gave Return-To-Flat (RTF: also known as Return To Zero decibel) operation. The responses are caused to revert back to unity gain outside the audio band. Strenuous efforts were made to ensure that the response was not affected by variations in pot track resistance, which are usually ±20%. This used a lot of op-amp sections (six in each channel of the tone control stage) and it seemed to me that if we compromised a little on pot-track dependency, two much simpler stages dealing with bass and treble separately, and borrowed from mixing console technology with only two op-amp sections per stage, might work pretty well. The price of freedom from pot-track dependency is not eternal vigilance but more op-amps. As a tone-control it can be added to existing preamps that are not so blessed, to give great flexibility. Be aware a low-impedance drive is required if the equalizer's response is to be predictable.

Optimizing a High-Frequency Equalizer

A typical variable frequency high-frequency shelving equalizer is shown in **Figure 1**. R1 and R2 set the basic gain to -1 and ensure that there is DC feedback to fix the operating conditions. When the wiper of RV1 is at the output end, positive feedback partially cancels the negative feedback through R2 and the gain increases. When the wiper of RV1 is at the input end, the signal fed through the frequency-sensitive sidechain R3, R4, A2, etc., causes partial cancellation of the input signal, so gain is reduced. The signal from pot RV1 is scaled by divider R3, R4, which set maximum cut/boost. This signal is buffered by voltage-follower A2 and fed to a high-pass RC network comprising C1 and (R5 + RV2). Only high frequencies are passed, so the equalizer has no effect at low frequency.

Specifying the frequencies at which equalizers work is difficult. One possibility is stating the frequency at which the response is 3dB less than the shelving level for maximum boost, but this is useless for any other boost/cut setting. In the past

I have used the frequency at which the response is 1dB more or less than the flat response, calling it the break frequency. The downside is that the break frequency is some ways from where the main action happens.

This EQ section was and is pretty much the industry standard in mixing consoles that did not aspire to parametric EQs, and I certainly do not claim to have originated it. It gives a lot of flexibility with a minimum of components. Because of its mixer ancestry I tend to think of it as an equalizer rather than a tone-control, but I have used the terms interchangeably here.

Originally, both op-amps would have been TL072 sections, for the powerful reason that they were the cheapest reasonable op-amps available. Another more subtle reason is that as a JFET-input part, the input bias currents were very low, and they could be connected directly to a pot wiper without crackling noises when the control was moved. This saved a lot of money on DC-blocking capacitors. Crackling pots on a mixer are bad because you may move the control for dynamic effects, but I suggest there is rather less of an issue in a hi-fi preamp, so I have here chosen to concentrate on noise.

The use of the well outmoded TL072 means the stage is noisy by modern standards. You could update the stage by using a more modern JFET-input op-amp such as the OPA2143, but in a world where supply chains are still a mess I decided to focus on the cheapest and most easily obtainable op-amp: the 5532. This is of course a BJT-input part, which gives low noise with low source impedances, and has good load-driving abilities, allowing us to use Low Impedance Design, which reduces both resistor Johnson noise and the effect of op-amp current noise flowing in the resistors. Voltage noise is unaffected. The 5532 does not have negligible input bias currents.

The focus of this article is on the optimization of the stage by reducing its noise, and this means doing more thinking rather than spending more money on fancy parts. All noise measurements are 22Hz to 22kHz RMS-sensing, unweighted.

Noise calculations are wonderful things, but they get rather complicated in circuits like this with a mixture of positive and negative feedback. It is much quicker to build the circuit and measure it, so I did. There are many settings possible, but I only tested the extremes of boost/cut and frequency, as otherwise we'd be here all week. See the results in **Table 1**.

Step 1: Obviously the first thing to do is replace the TL072 with a 5532 section, and **Table 2** shows an immediate improvement of 2dB to 4dB at the various settings.

Step 2: I suggest the next thing to look at is the relatively high value of R1 and R2. Using a 5532 means we can reduce these considerably. From here on in, to reduce the amount of measuring, we are looking at just minimum frequency (which gives

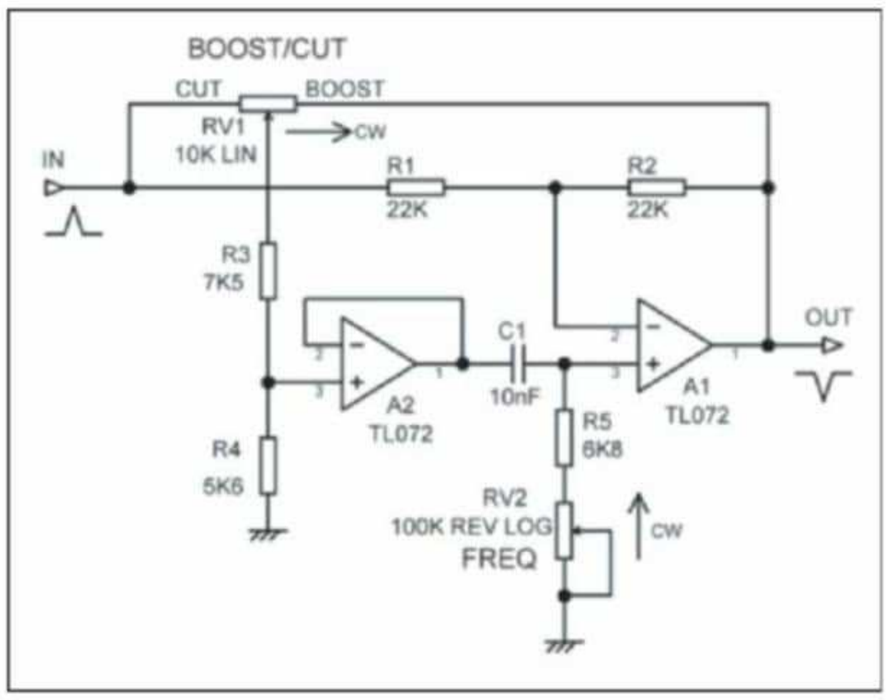


Figure 1: The variable-frequency high-frequency shelving circuit as in SSAD3 p412. The ±1dB break frequency range is 250Hz to 4kHz.

A1	A2	Freq	Boost/cut	R1=R2	Noise out
TL072	TL072	Min	Boost	22k	-84.1 dBu
TL072	TL072	Min	Flat	22k	**-94.5 dBu**
TL072	TL072	Min	Cut	22k	-97.6 dBu
TL072	TL072	Max	Boost	22k	-95.3 dBu
TL072	TL072	Max	Flat	22k	-97.8 dBu
TL072	TL072	Max	Cut	22k	-99.5 dBu

Table 1: Noise measurements for the Figure 1 circuit with TL072 op-amps

A1	A2	Freq	Boost/cut	R1=R2	Noise out
5532	5532	Min	Boost	22k	-88.8 dBu
5532	5532	Min	Flat	22k	**-98.1 dBu**
5532	5532	Min	Cut	22k	-100.2 dBu
5532	5532	Max	Boost	22k	-99.5 dBu
5532	5532	Max	Flat	22k	-101.8 dBu
5532	5532	Max	Cut	22k	-103.3 dBu

Table 2: Noise measurements for the Figure 1 circuit with NE5532 op-amps

A1	A2	Freq	Boost/cut	R1=R2	Noise out
5532	5532	Min	Flat	22k	**-98.1 dBu**
5532	5532	Min	Flat	4k7	**-100.8 dBu**
5532	5532	Min	Flat	2k2	**-101.4 dBu**

Table 3: Noise measurements for the Figure 1 circuit with NE5532 op-amps and reduced values of R1 and R2

A1	A2	Freq	Boost/cut	R1=R2	Noise out
5532	5532	Min	Boost	2k2	-96.4 dBu
5532	5532	Min	Flat	2k2	**-101.9 dBu**
5532	5532	Min	Cut	2k2	-103.4 dBu

Table 4: As in Table 3 but with control range reduced to ±12dB

the biggest variation in noise with boost and cut, because the amount of the audio spectrum affected is larger) and settings of maximum boost, flat, and maximum cut. The results for the flat condition are bolded in the tables. We get a very handy 3.3dB reduction in noise at zero cost, as in **Table 3**.

Step 3: The stage in Figure 1 gives ±15dB of boost/cut, which is standard for mixing consoles. It is however a bit excessive for hi-fi use, so the range was reduced to ±12dB by reducing R4 to 4k7. That affects the noise, so it is re-measured in **Table 4**.

A1	A2	Freq	Boost/cut	R1=R2	Noise out
5532	5532	Min	Boost	2k2	-98.3 dBu
5532	5532	Min	Flat	2k2	**-104.7 dBu**
5532	5532	Min	Cut	2k2	-106.6 dBu

Table 5: As in Table 4 but with R3 and R4 reduced in value

A1	A2	Freq	C1	R5	VR2	Boost/cut	R1=R2	Noise out
5532	5532	Min	4n7	3k3	47k	Boost	2k2	-99.2 dBu
5532	5532	Min	4n7	3k3	47k	Flat	2k2	**-105.8 dBu**
5532	5532	Min	4n7	3k3	47k	Cut	2k2	-107.6 dBu

Table 6: As in Table 5 but with the impedance of C1 impedance doubled, R5, VR2 halved

A1	A2	Freq	C1	R5	VR2	Boost/cut	R1=R2	Noise out
5532	5532	Min	10nF	1k5	22k	Boost	2k2	-99.7 dBu
5532	5532	Min	10nF	1k5	22k	Flat	2k2	**-106.7 dBu**
5532	5532	Min	10nF	1k5	22k	Cut	2k2	-108.8 dBu

Table 7: As in Table 6 but with C1 impedance doubled, R5, VR2 impedances halved yet again

A1	A2	Freq	C1	R5	VR2	Boost/cut	R1=R2	Noise out
5532	5532	Min	15nF	1k	15k	Boost	2k2	-99.8 dBu
5532	5532	Min	15nF	1k	15k	Flat	2k2	**-107.0 dBu**
5532	5532	Min	15nF	1k	15k	Cut	2k2	-109.2 dBu

Table 8: Again C1 impedance doubled, R5, VR2 impedances halved, taken to its extreme

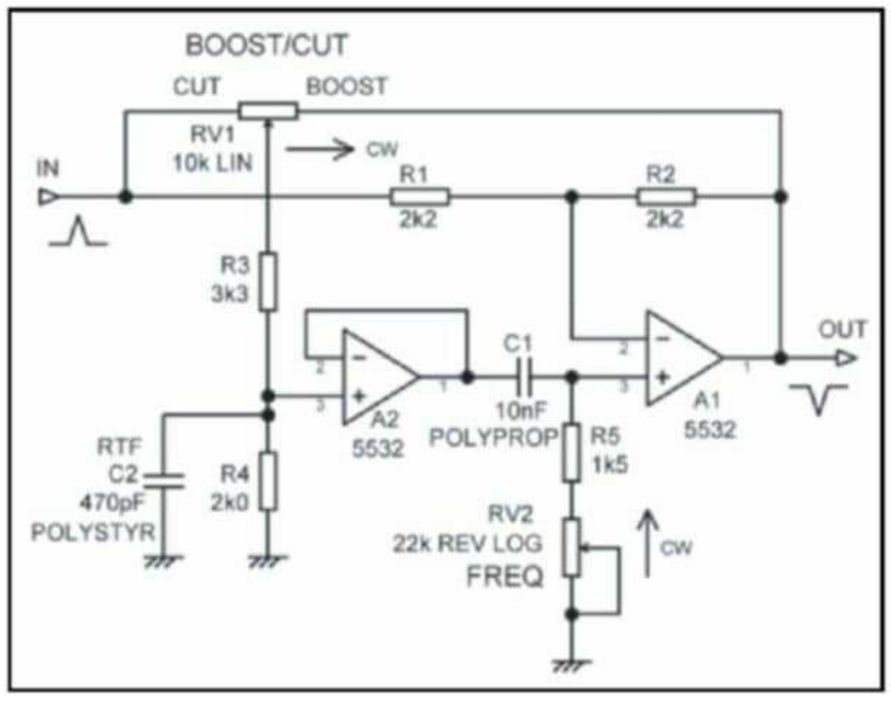


Figure 2: The variable-frequency high-frequency shelving circuit after optimization

If you compare Table 4 with the top three rows of Table 2, we have reduced the noise by from 2dB to 7dB, depending on the boost/flat/cut setting.

Step 4: Now what? Could R3 and R4 be reduced in value, keeping their ratio the same so that the maximum boost/cut is unaltered? Back in the day, the conventional wisdom was that R3 and R4 could not be much lower than shown in Figure 1, because the loading effect on the boost/cut pot would cramp the control to the middle of the pot rotation. I did not then question it. However, now I do question it, because recent simulations showed that reducing R3 to 3k3, and R4 to 2k0 (less than half the original values) only modified the intermediate response curves by 1.4 dB or less. But how much will it reduce noise? See **Table 5**. Comparing Table 5 with Table 4, we have reduced the noise by 2dB at minimum. We move on and see what else we can tamper with.

Step 5: Reducing the impedance of the frequency selective network C1, R5, VR2 looks promising; obviously both resistors and the capacitor need to be scaled to keep the frequency response the same. In **Table 6** we halve the impedance, gaining about 1dB, and in **Table 7** we halve it again, gaining another 0.8dB. **Table 8** takes it about as far as it can be pushed, due to the loading effect of R5 on A2 at minimum frequency, and the further noise reduction is only some 0.3 dB. Given that 15kΩ pots are not going to be easy to find, Table 7 looks like the best result. The optimized circuit is shown in **Figure 2**. Capacitor C1 should be polystyrene or polypropylene to avoid distortion. Just by changing component values and op-amps, the noise set flat at minimum frequency (worst case) is reduced from -94.5dBu to -107dBu, an improvement of no less than 12.5dB without extra cost.

The only component not tampered with so far is the boost/cut pot RV1. Reducing its effective value to 5kΩ and then 3kΩ gave improvements of only a fraction of a decibel and introduced concerns about the load placed on the op-amp and the previous stage. The pot was therefore left at 10kΩ.

Another feature provided by the *Linear Audio* preamp was the RTF outside the audio band. To implement this, we need to stop the tone control action at very high frequencies, and the simplest way of doing that is to put a small capacitor across R4. The result of adding a 470pF capacitor across R4 is demonstrated in **Figure 3**. The capacitor should be polystyrene to avoid distortion.

The typical tone-cancel switch is a changeover that gives a brief break in the audio as it moves between tone-control and bypass. I hold that the resulting transients make critical judgements harder, and a no-break switch that just turns on

and off the tone-control action is superior. No-break high-frequency EQ cancel can be implemented by a short to ground at either the non-inverting input to A2 or the non-inverting input to A1. The latter is preferred as the noise from A2 does not reach A1 in cancel mode.

The final re-designed stage is shown in Figure 2, and the response at the frequency extremes is shown in Figure 3. The lower frequency curve shows the start of RTF action at 20kHz. When the equalizer is set to flat with a 3Vrms input, THD remains at 0.0004% from 20kHz down to 10Hz.

The equalizer stage inherently phase-inverts so it needs another inversion before or after it to maintain absolute phase. An extra stage that merely inverts to correct phase has never appeared to me a good use of power and components; here the obvious answer is to cascade it with the equivalent low-frequency equalizer, which also phase-inverts, giving a conventional two-band tone control, and getting the phase straight again.

Optimizing a Low-Frequency Equalizer

The low-frequency shelving equalizer works in the same way as the high-frequency equalizer but uses a low-pass rather than a high-pass sidechain. The classic circuit is shown in **Figure 4**. The noise output of the stage set to maximum frequency (the worst case) is shown in **Table 9**. The noise out with the equalizer set flat and to maximum frequency is -99.7dBu, compared with -94.5dBu for the high-

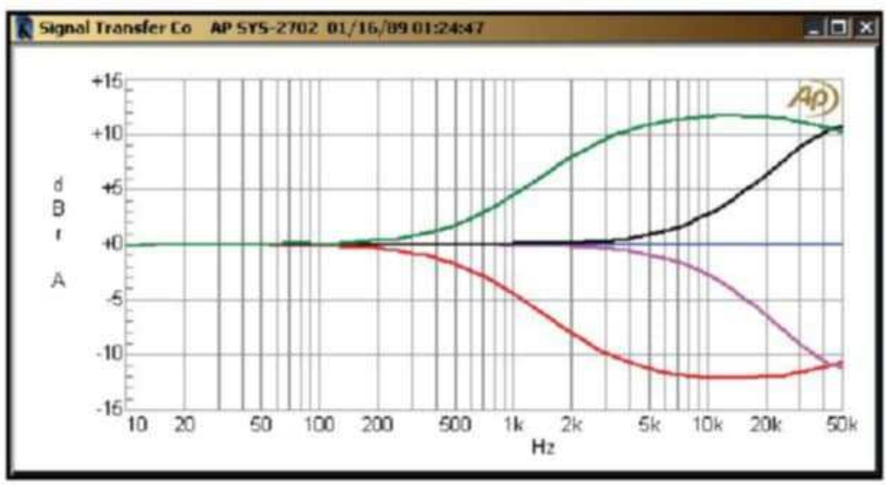


Figure 3: Frequency response of variable-frequency high-frequency shelving circuit after optimization. The 1dB break frequencies at the extremes of the frequency control are 250Hz and 4kHz.

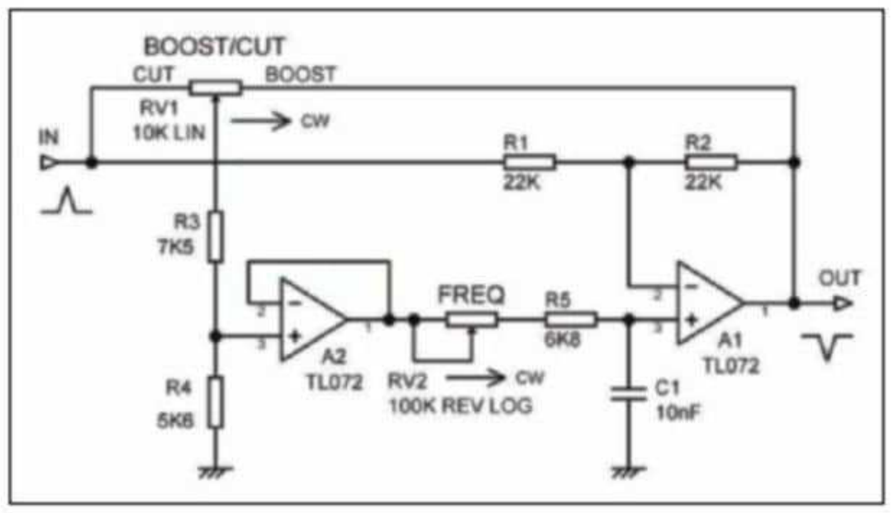


Figure 4: The variable-frequency low-frequency shelving circuit as in SSAD3 p414. The ±1dB break frequency range is 250Hz to 4kHz.



A1	A2	Freq	Boost/cut	R1=R2	Noise out
TL072	TL072	Max	Boost	22k	-97.4 dBu
TL072	TL072	Max	Flat	22k	**-99.7 dBu**
TL072	TL072	Max	Cut	22k	-100.4 dBu

Table 9: Noise output of Figure 4 circuit at maximum frequency with TL072 op-amps

A1	A2	Freq	Boost/cut	R1=R2	Noise out
5532	5532	Max	Boost	22k	-102.0 dBu
5532	5532	Max	Flat	22k	**-103.8 dBu**
5532	5532	Max	Cut	22k	-104.5 dBu

Table 10: Figure 4 circuit with NE5532 op-amps

A1	A2	Freq	Boost/cut	R1=R2	Noise out
5532	5532	Min	Flat	22k	**-103.8 dBu**
5532	5532	Min	Flat	4k7	**-108.2 dBu**
5532	5532	Min	Flat	2k2	**-109.3 dBu**

Table 11: Figure 4 noise with R1 and R2 reduced in value

A1	A2	Freq	Boost/cut	R1=R2	Noise out
5532	5532	Min	Boost	2k2	-107.5 dBu
5532	5532	Min	Flat	2k2	**-109.8 dBu**
5532	5532	Min	Cut	2k2	-110.4 dBu

Table 12: Figure 4 noise with control range reduced to ±12dB and reducing R3 and R4 in value

A1	A2	Freq	C1	R5	VR2	Boost/cut	R1=R2	Noise out
5532	5532	Min	22nF	3k3	47k	Boost	2k2	-107.9 dBu
5532	5532	Min	22nF	3k3	47k	Flat	2k2	**-110.8 dBu**
5532	5532	Min	22nF	3k3	47k	Cut	2k2	-111.1dBu

Table 13: Further reducing the frequency selective network C1, R5, and RV2 by half

A1	A2	Freq	C1	R5	VR2	Boost/cut	R1=R2	Noise out
5532	5532	Min	47nF	1k5	22k	Boost	2k2	-110.0 dBu
5532	5532	Min	47nF	1k5	22k	Flat	2k2	**-111.4 dBu**
5532	5532	Min	47nF	1k5	22k	Cut	2k2	-111.3 dBu

Table 14: Reducing the frequency selective network C1, R5, and RV2 taken to its extreme, with only limited advantage

About the Author



Douglas Self has a world-wide reputation as a leading authority on audio power amplifier design, but it is perhaps less well-known that he has devoted a good deal of study to preamplifiers and other small-signal circuitry, including many years as the Chief Design Engineer at one of the major mixing console manufacturers, where his achievements included winning a Design Council Award. His rigorous, skeptical, and thoroughly practical approach to design has been applied to the small-signal area as well. Douglas studied engineering at Cambridge University, and then Psychoacoustics at Sussex University. He has spent many years working at the top level of design in both the professional audio and hi-fi industries, and has taken out a number of patents in the field of audio technology.

frequency version. This is because the frequency-selective side chain in the low-frequency equalizer is a low-pass network and so attenuates the noise from R3, R4, and A2. In Tables 9-14 the noise in the flat condition is bolded.

The optimization of the low-frequency equalizer proceeds exactly as for the high-frequency equalizer.

Step 1: First replace the TL072 with a 5532 section, and **Table 10** shows the noise reduced by 4.1 to 4.6 dB in the three boost/cut settings.

Step 2: The relatively high values of R1 and R2.are reduced in stages to 2k2, as shown in **Table 11**.

Step 3 and Step 4 are then carried out together to save time, so the boost /cut range is reduced from ±15dB to ±12dB, and R3 is reduced to 3k3 and R4 reduced to 2k0, giving us 1.6dB less noise in the flat condition. See **Table 12**.

Step 5: Reducing the impedance of the frequency selective network C1, R5, RV2 by making R5 = 3k3, C1 = 22nF gives us a further noise reduction of 1dB. See **Table 13**. Making R5 = 1k5, C1 = 47nF gives us a yet further noise reduction of 0.6dB, and clearly, we are into diminishing returns, and we will stop there, as we did for the high-frequency equalizer. Noise has been reduced by 11.7dB for no added cost, compared with the 12.5dB for the high-frequency equalizer stage. See **Table 14**. Capacitor C1 should be polystyrene or polypropylene to avoid distortion.

As for the high-frequency equalizer, the only component not altered is the boost/cut pot RV1. Reducing it to 5kΩ and then 3kΩ again gave only tiny reductions in noise, and there is concern about increasing the load seen by the op-amp and the previous stage. This pot was also left at 10kΩ.

To implement the RTF function, a high-pass time-constant must be added to the sidechain so its output falls at very low frequencies. This can be simply done by inserting a capacitor C2 in series with R3, but the snag is that it needs to be large because of the relatively low value of R3 and R4; about 22µF. This would be unacceptably big and expensive if it was a non-electrolytic type. I explored using, say, a 470nF capacitor and increasing its effective value by electronic trickery, but the simplicity of the basic circuit is lost and there were some awkward problems with offset voltages. Using an electrolytic capacitor for C2 seems at first ill-advised because of the distortion it would generate if there was a significant signal voltage across it. We might piously hope there are no big signals around 1Hz, which is where this component does its work. Electrolytics also have large tolerances, but are, I think, acceptable here because small variations in response at 1Hz are of little consequence.

THD was measured at full boost and 10Vrms out, just short of clipping, and with C2 as a 22V/63V part there was 0.0008% at 20Hz; without the capacitor you get 0.0006%. With a 22V/35V part you find a rather less appealing 0.0023%. This is an interesting example of how the voltage rating of an electrolytic can affect how much distortion it creates. In other tests at differing impedances, I have found the voltage rating made very little difference. If you think that is just too much distortion, then C2 can be increased to 47V/63V, which gives about 0.0004% at 20Hz; the RTF will now take place at a lower frequency. Above 100Hz, THD level is about 0.0004% up to 20kHz. See **Figure 5** for the final optimized circuit. When the equalizer is set to flat, THD remains at 0.0004% from 20kHz down to 10Hz.

The RTF for the low-frequency equalizer does its stuff almost entirely below 10Hz, and so cannot be explored with your typical audio test gear. Function generators commonly go down to 1Hz at least, but accurate measurement is hard. My solution is to divide the values of all the capacitors by 10 times so tests can be made at 10 times the frequency. **Figure 6** shows the RTF response with frequency set to maximum. This gives an accurate scaled frequency response but will not give reliable results for noise or distortion. The frequency responses for the tone control extremes (with the capacitors scaled back up again) are shown in **Figure 7**.

No-break low-frequency EQ cancel can be implemented by a short to ground at either the non-inverting input to A2 or the non-inverting input to A1. The latter is preferred as it stops the noise from A2 reaching A1 in cancel mode. Since there are separate high-frequency and low-frequency equalizers in each channel, an overall tone-cancel switch would have four sections, making it a bit unwieldy. I think it would be far preferable to have separate switches for high-frequency and low-frequency tone-cancel. This is also useful as corrections made at one frequency extreme are likely to have little in common with corrections at the other extreme.

When the low-frequency and high-frequency equalizers are cascaded, the noise adds in the usual RMS fashion. **Table 15** gives the output noise for the flat setting at the frequency extremes of the two stages. It does not matter in which order the low-frequency and high-frequency equalizers are connected.

Conclusion

I think this article shows that our noise-reduction project has been successful. We now have a very flexible and very quiet tone-control. The break frequency ranges can be changed to suit requirements, without changing anything else, by altering the capacitor C1 in the sidechain for both high-frequency and low-frequency cases. The break frequency ranges can be reduced if required by increasing the frequency-endstop resistor R5. Do not reduce it as this may lead to excess loading on A2.

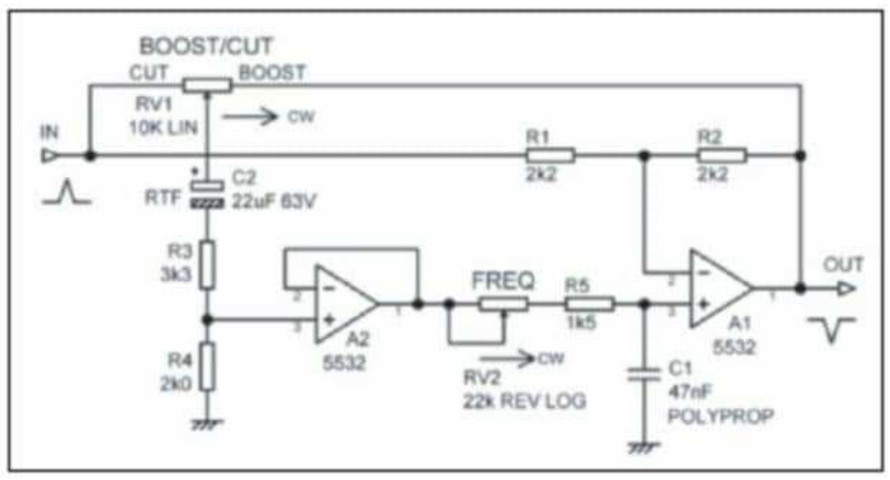


Figure 5: The variable-frequency low-frequency shelving circuit after optimization

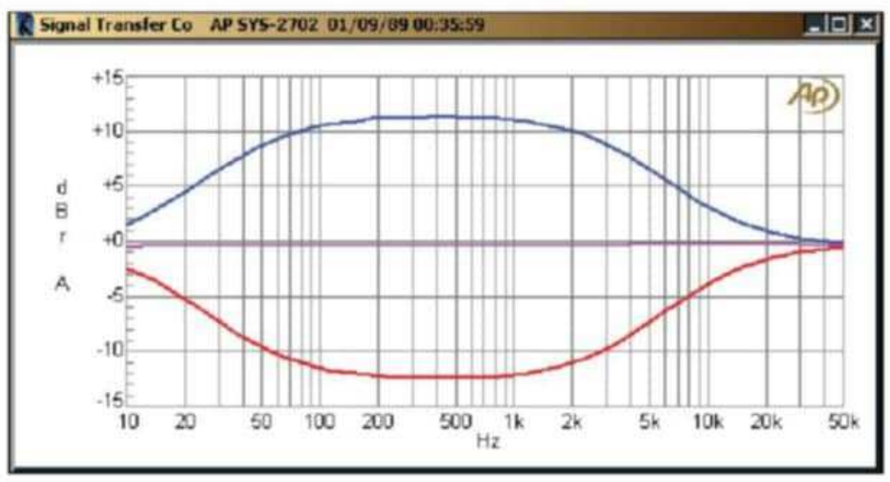


Figure 6: Demonstrating the low-frequency RTF feature with capacitor values divided by 10 times. Maximum frequency.

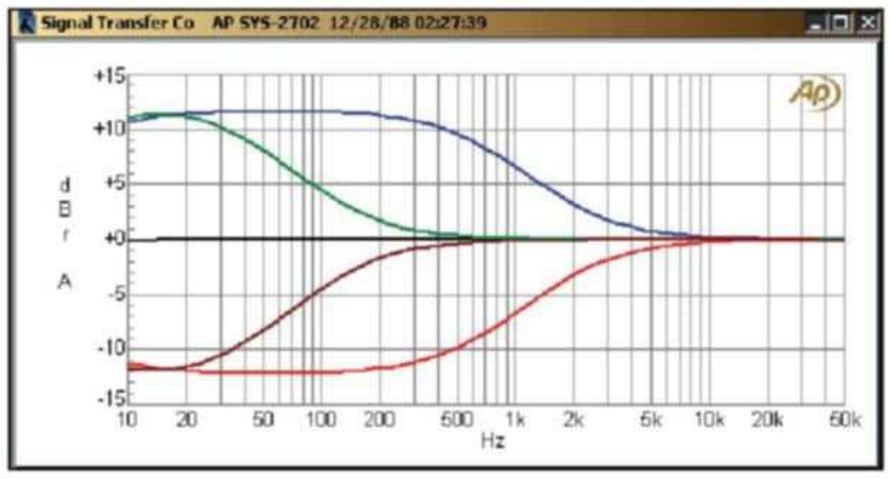


Figure 7: Frequency response of the low-frequency shelving circuit at extreme frequency settings. The ±1dB break frequencies at the extremes of the frequency control are 250Hz and 4kHz.

A1	A2	LF freq	HF freq	Boost/cut	R1=R2	Noise out
5532	5532	Min	Min	Flat	2k2	**-105.1 dBu**
5532	5532	Min	Max	Flat	2k2	**-106.9 dBu**
5532	5532	Max	Min	Flat	2k2	**-105.2 dBu**
5532	5532	Max	Max	Flat	2k2	**-107.2 dBu**

Table 15: Overall noise output with a low-frequency and a high-frequency equalizer cascaded at the frequency extremes

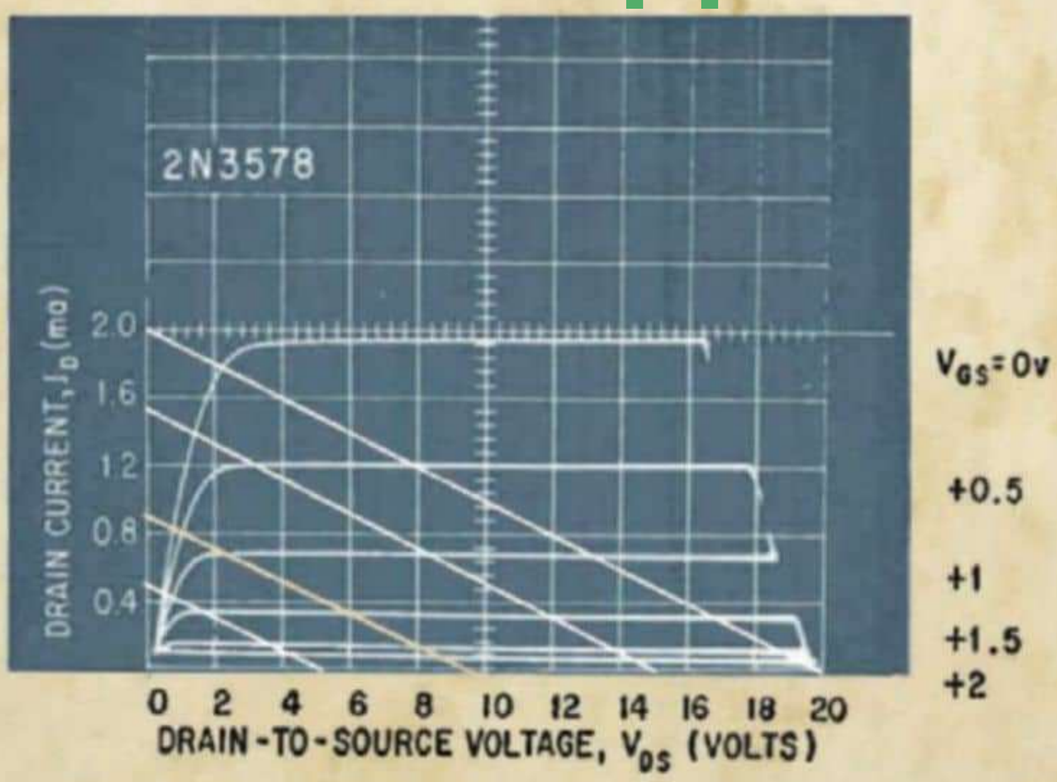
References

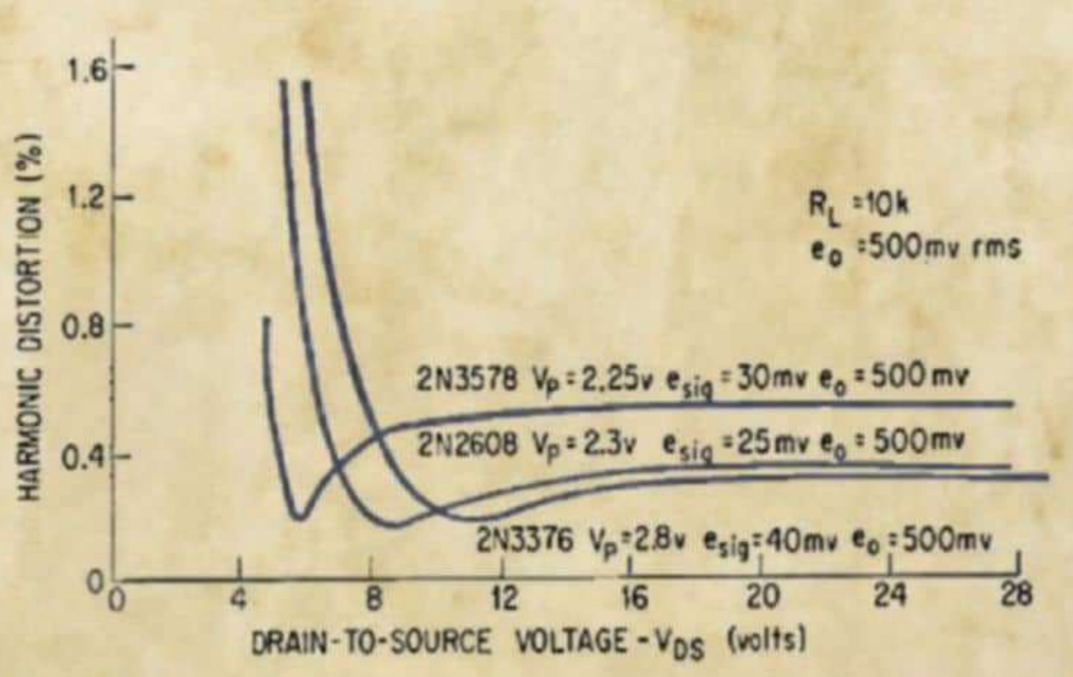
[1] D. Self, "Precision Preamplifier 96," *Electronics World*, July/August and September 1996.

[2] DIYaudio bulletin board, www.diyaudio.com/forums/solid-state/209004-new-doug-self-pre-amp-design-22.html

[3] D. Self, "A Low Noise Preamplifier with Variable-Frequency Tone Controls," *Linear Audio*, Volume 5, p. 141.

JFETs in the Variable-Resistance Mode and their Application in Audio Circuits





By
Dimitri Danyuk

This article explores the variable-resistance mode of JFET operation and describes a practical circuit, built around JFETs in the variable-resistance mode with a drain load resistor. The circuit is applied in a complete phono preamp design in transconductance mode.

The junction-gate field-effect transistor (JFET) has two distinct modes of operation: the variable-resistance mode (resistive mode, Ohmic region, or triode region) and the saturation (pinch-off or pentode) mode. In the resistive mode the JFET behaves like a resistor whose value is controlled by the voltage between gate and source V_{gs}. In the saturation mode an increase in V_{ds} does not result in a correspondent increase in I_d, and the output characteristic flattens out. The channel is heavily constricted with most of the drain-source voltage drop occurring along the narrow and therefore high-resistance part of the channel. Let's start with the variable-resistance mode of operation.

JFETs in the Variable-Resistance Mode

Twelve years ago, I designed an impedance converter for condenser microphones for a high-end microphone manufacturer. Condenser microphone capsules have low capacitance (10pf to 80pF typically) and should be loaded at a very high impedance. An impedance converter (or buffer) is used to match the impedance of the microphone capsule to the input impedance of the recording device together with the long cable.

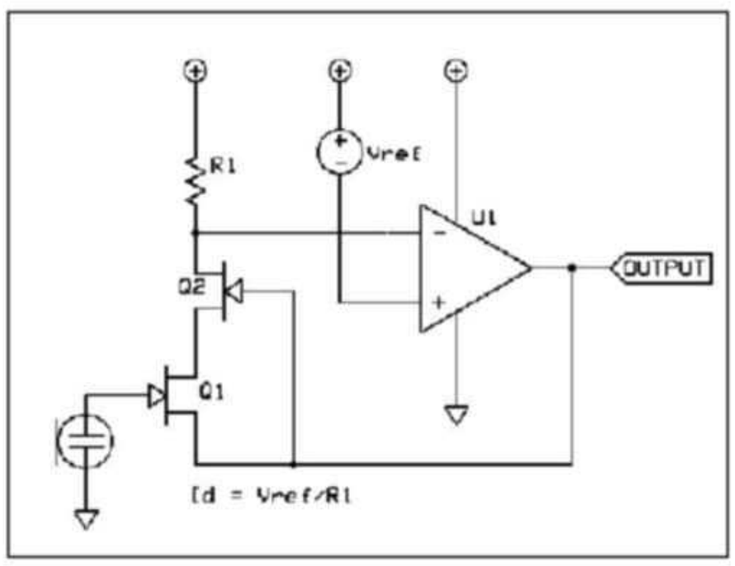


Figure 1: Impedance converter for a condenser microphone

The JFET source follower is the simplest form of the impedance converter. It can be improved by adding a cascode and the output stages to the JFET follower, as shown in **Figure 1**. The cascode device reduces the input capacitance while the output stage improves the output current capability of the JFET follower. According to the handbooks, the pinch-off voltage for the upper device should be high enough to keep the lower JFET in the saturation region.

The cascode JFET Q2 works as a load for a lower JFET Q1, and voltage at the drain of Q1 follows the voltage on the source of Q1. This means that Q2 provides a low input resistance to Q1 and the gate to drain capacitance of Q1 is bootstrapped.

I tested the circuit, shown in Figure 1 with different cascode devices. The typical graph of my measured total harmonic distortion plus noise (THD+N) is shown in **Figure 2**.

Distortion in the circuit shown in Figure 1 becomes lower with lower pinch-off voltage of the cascode JFET and lower voltage between source and drain of the input JFET. That was counterintuitive, but I decided to stop and use the cascode device that gave me the lowest distortion. Later I noticed the app note by Alex Nikitin on similar cascode behavior and I allowed myself to spend some time to satisfy my curiosity.

James Sherwin investigated distortion in simple common source stages. The gain equation of the common source stage is $A=G_{fs}(R_L+1/G_{os})$, where G_{fs} is the JFET forward transconductance, R_L is the load and G_{os} is the output conductance. The output

conductance G_{os} is the ratio of change in drain current I_d to a change in voltage between drain and source V_{ds}.

Transconductance G_{fs} increases with drain current I_d, and the output transconductance G_{os} also increases with I_d. Moreover, the output transconductance G_{os} increases with reduced V_{ds}. At some value of the drain current I_d and drain to source voltage V_{ds} the change in the G_{fs} can be partially offset by the change in G_{os} (or together in R_L and G_{os}). In effect, the distortion produced by the nonlinear G_{fs} can be compensated by distortion produced by the nonlinear G_{os}. This cancellation of two distortion sources produces a point with minimum distortion.

The cascode device works as a common gate stage. The input resistance of the common gate stage is equal to $1/G_{fs}$ and it acts as a load resistor R_L for the input JFET. Let us have a look at the cascode characteristics shown in **Figure 3**.

The small signal transfer function for the JFET at a given DC conditions may be written as a Taylor series power expansion:

$$I_d = \frac{1}{1!}G_{fs}\nu_{sig}^2 + \frac{1}{2!}\frac{\partial G_{fs}}{\partial V_{gs}}\nu_{sig}^2 + \frac{1}{3!}\frac{\partial^2 G_{fs}}{\partial V_{ds}}\nu_{sig}^3 + \ldots\frac{1}{n!}\frac{\partial^{n-1} G_{fs}}{\partial V_{ds}^{n-1}}\nu_{sig}^n$$

$$\nu_{sig} = A_{sig}\sin\omega t$$

The first term $\frac{1}{1!}G_{fs}\nu_{sig}$ represents the fundamental, the second term $\frac{1}{2!}\frac{\partial G_{fs}}{\partial V_{gs}}\nu_{sig}^2$ represents DC offset plus the second harmonic of fundamental $\left(A_{sig}\sin^2\omega t = \frac{A_{sig}}{2}(1-\cos\omega t)\right)$.

Having a look at the cascode characteristics (Figure 3), it is easy to observe an inflection point on the cascode transfer curve (I_d is about 1.8mA and V_{gs} is about -0.9V). At this point the curvature of the smooth curve changes sign, the curve shape changes from convex to concave. For a smooth curve an inflection point is a point on the graph at which the second derivative has an isolated zero and changes sign.

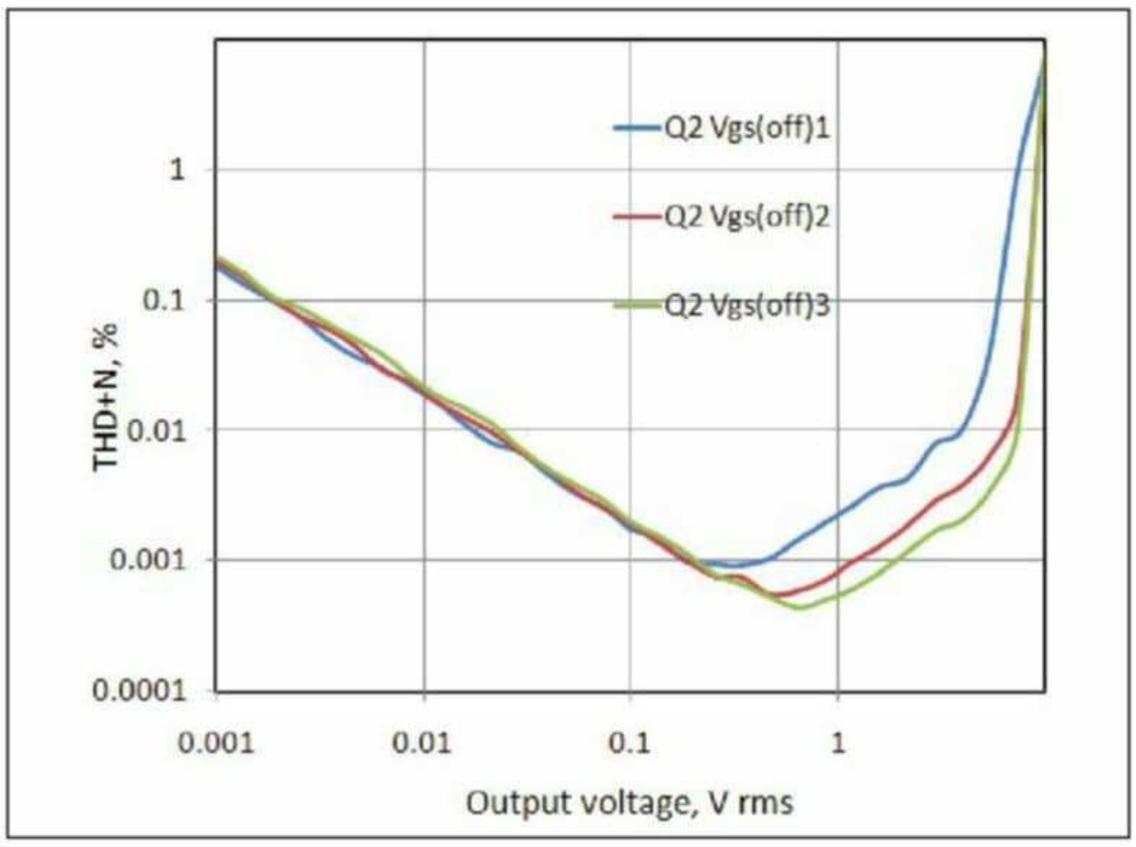


Figure 2: Total harmonic distortion plus noise for the circuit shown in Figure 1 with different cascode JFETs. V_{gs}(off) is the pinch-off voltage of the cascode JFET Q2, Q2 V_{gs}(off)1 > Q2 V_{gs}(off)2 > Q2 V_{gs}(off)3.

As $G_{fs} = \frac{\partial I_d}{\partial V_{gs}}$ the second term can be rewritten as $\frac{1}{2!} = \frac{\partial^2 I_d}{\partial V_{ds}^2}\nu_{sig}^2$.

At the inflection point $\frac{\partial^2 I_d}{\partial V_{ds}^2}$ is equal to zero, the second harmonic distortion, associated with the second term is zero and we can expect that the THD will reach minimum value in the vicinity of the inflection point, by operating on the most linear portion of the transfer curve.

To check the distortion behavior of the different JFETs in cascode configuration I assembled the simple circuit shown in **Figure 4**. Current source *I* sets the operating current for the input JFET Q1 and for cascode JFET Q2; a floating DC voltage source sets V_{ds} for the input device.

I measured dozens of JFETs and MOSFETs with this and the following circuit but will only share data for CPH3910 JFET (**Figure 5**) to save space. Results obtained with other devices are similar. There is a distortion minimum for each value of drain current. The distortion minimum becomes deeper and wider with drain current.

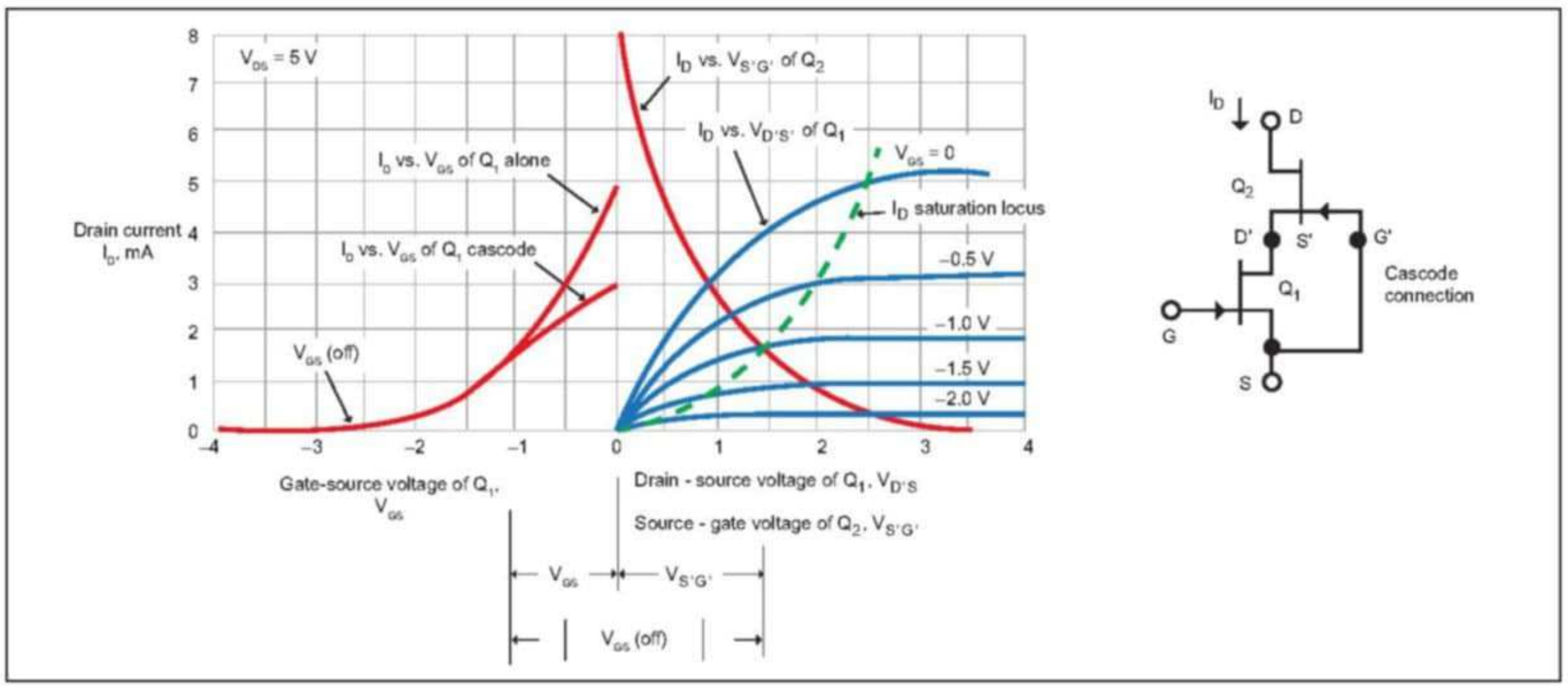


Figure 3: Cascode characteristics (from *Designing with Field-Effect Transistors*, ed. by A.D. Evans, McGraw-Hill, New York, 1981)

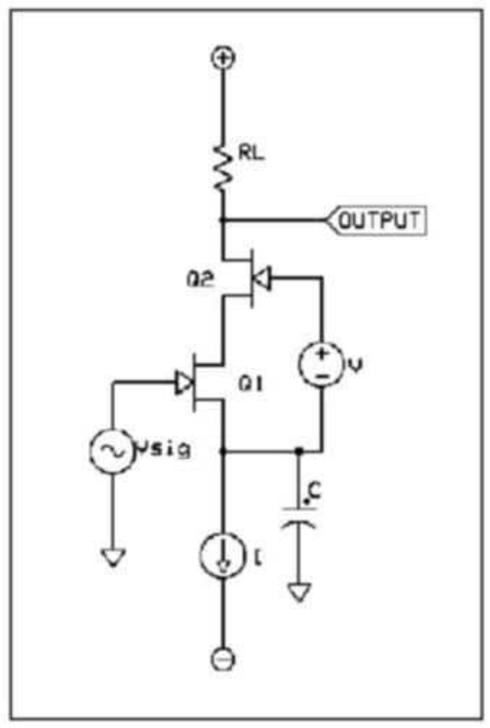


Figure 4: Circuit for cascode distortion measurement with variable V_{ds} of the input JFET

My next step was to determine the role of the cascode device. Does the nonlinearity of its input resistance play the leading role in changing the curvature of the transfer function and corresponding distortion reduction? To answer this question, I assembled the circuit shown in **Figure 6**. The cascode device is replaced with resistor connected in series with the drain terminal of the input JFET. The op-amp together with MOSFET Q2 maintains the stable voltage on the other side of the drain resistor being equal in value to floating DC voltage source V. The input JFET drain current passes through MOSFET Q2 to load resistor R_L.

Figure 7 shows the distortion for the JFET with drain resistor. Again, you can notice distortion minimums for each value of drain current. The minimums look a little bit narrower compared to the JFET cascode (Figure 5). At low drain to source voltages the nonlinearity in G_{os} of the device itself, reduced by Rd can compensate the quadratic nonlinearity of the JFET transfer for a certain range of the input signal v_{sig}. So, it is not necessary to use another JFET in common gate configuration to

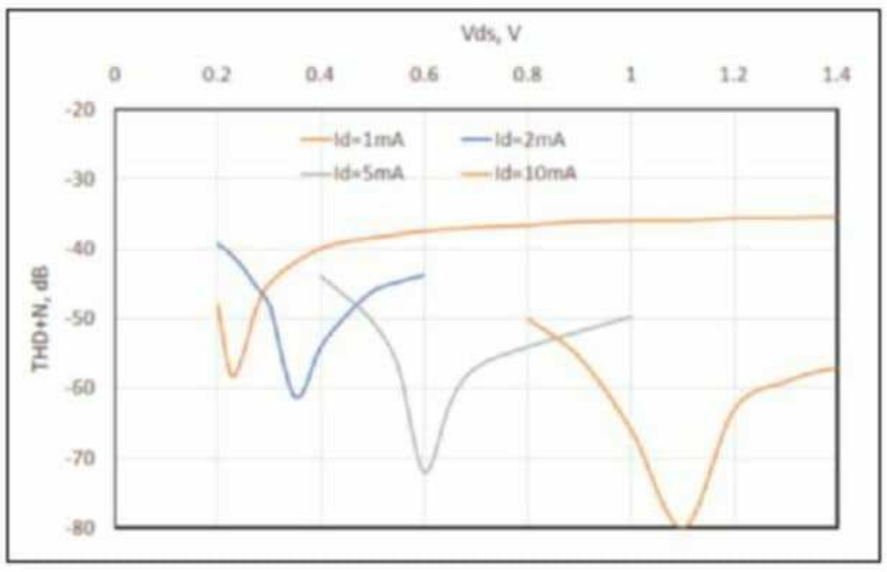


Figure 5: THD+N for CPH3910 JFET with variable V_{ds} of the input JFET. Cascode JFET is also CPH3910. The input signal is 10mV rms.

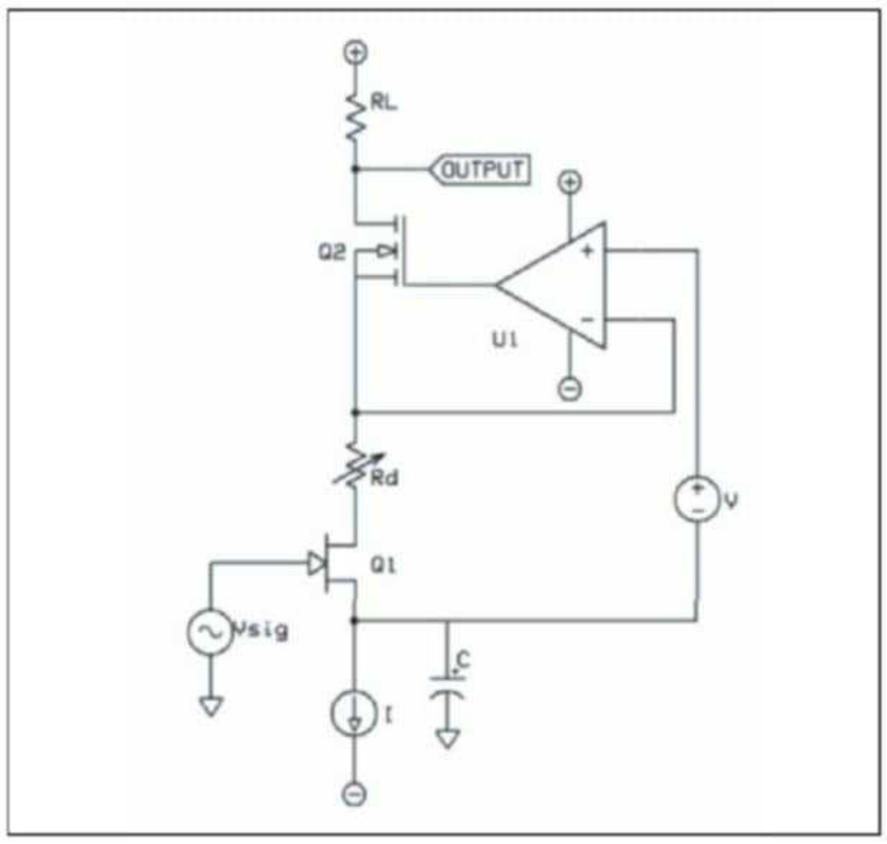


Figure 6: Circuit for JFET distortion measurement with variable V_{ds}

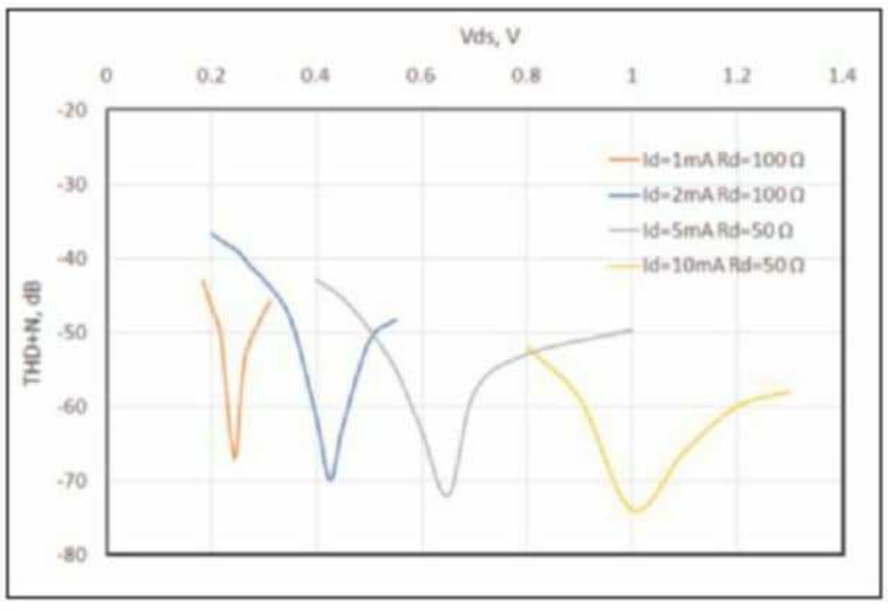


Figure 7: THD+N for CPH3910 JFET with variable V_{ds} and drain resistor. Input signal 10mV rms

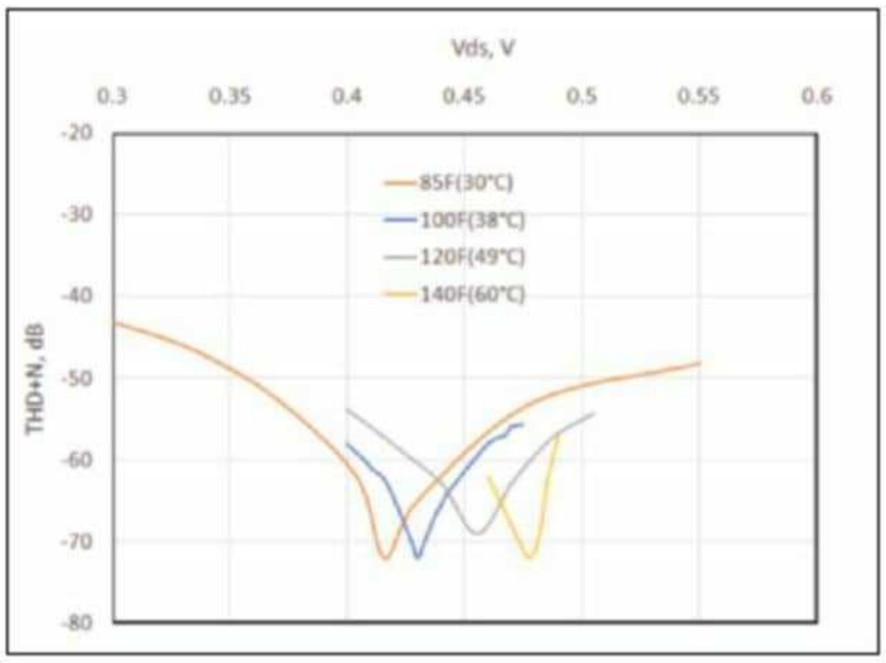


Figure 8: THD+N for CPH3910 JFET with variable V_{ds} with drain resistor Rd. Rd=100Ω. Input signal 10mV rms.

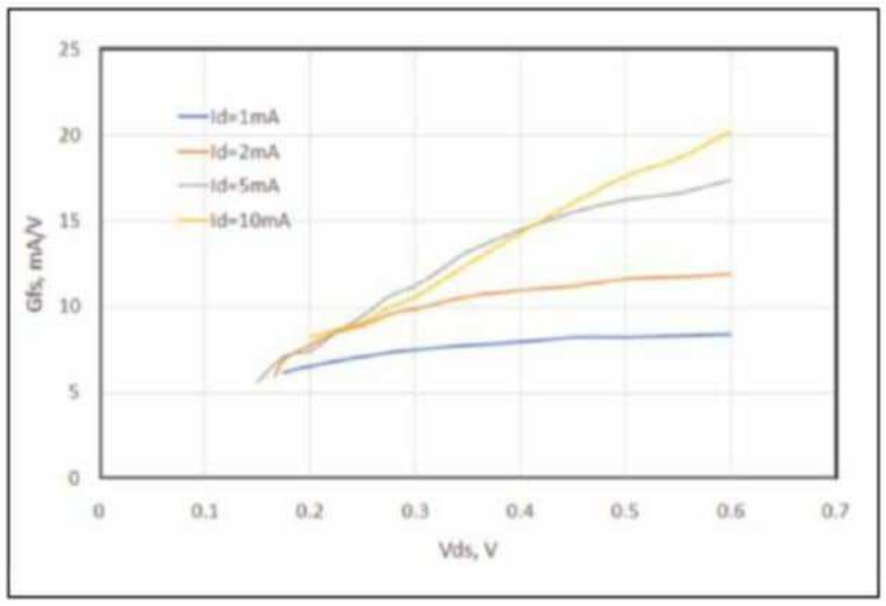


Figure 9: Transconductance curves for CPH3910 at low V_{ds}

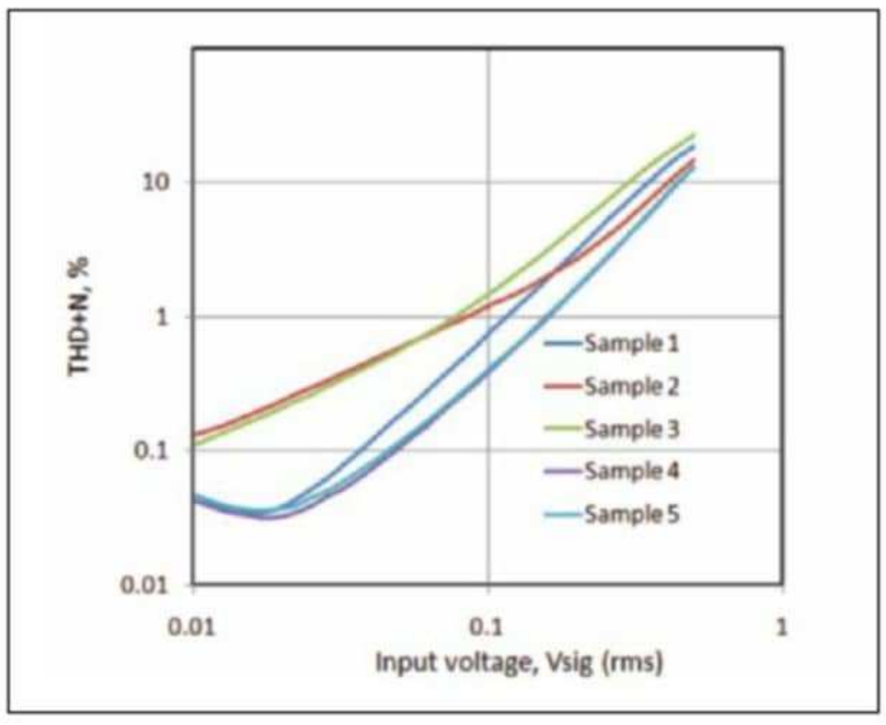


Figure 10: THD+N curves for different devices in the variable-resistance mode with a drain resistor and optimal V_{ds} (samples 1, 4, 5) and in saturation mode with high V_{ds} above the locus point (samples 2 and 3).

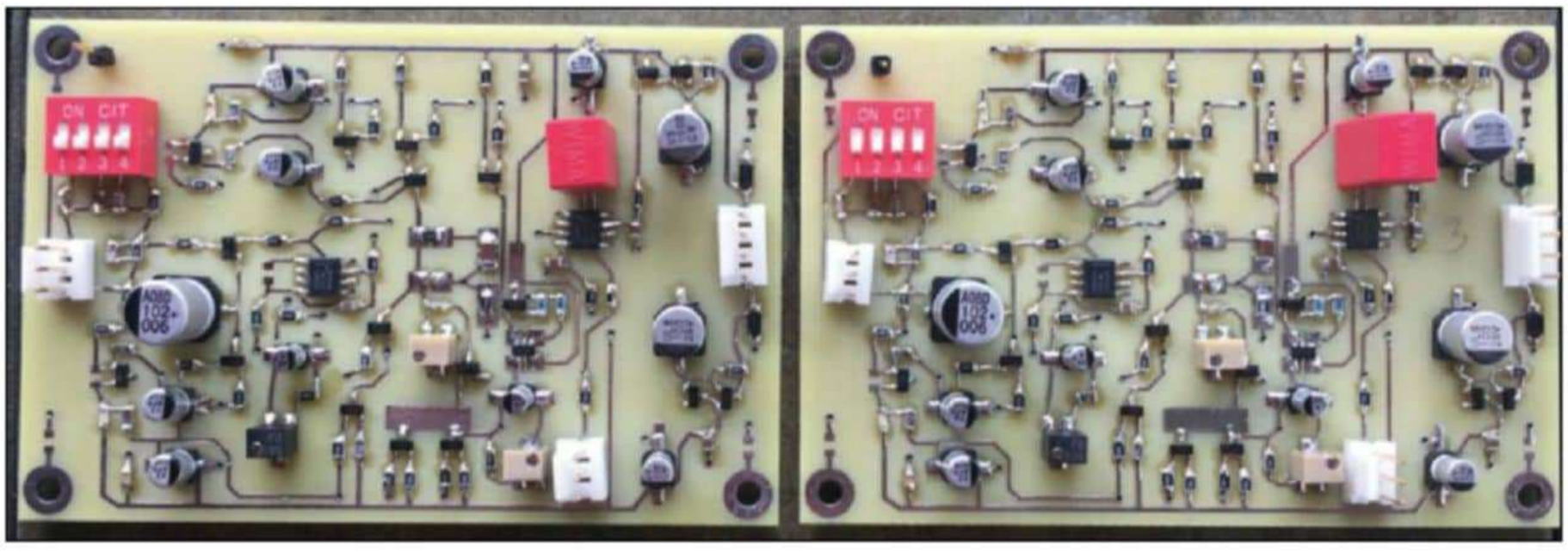


Photo 1: For the transconductance phono preamplifier, I built a a practical circuit around JFETs in the variable resistance-mode with a drain load resistor.

achieve distortion compensation. The N nonlinear resistor also can be used in place of Rd (for example, JFET with V_{gs}=0).

Now let me address the practical aspects of the JFET amplifying stage with a drain load resistor. V_{ds} should be maintained within a narrow range (say ±20mV) to keep good distortion reduction (Figure 5 and Figure 7). How does this voltage depend on the device temperature? Temperature dependence of V_{ds} setting for minimum distortion is shown in **Figure 8**. The drain-source voltage for minimum distortion varies with temperature, however at is still possible to get substantial distortion reduction over an acceptable temperature range (in this case 30°C...50°C) without V_{ds} trimming.

In the variable-resistance mode, transconductance G_{fs} strongly depends on the drain-source voltage. The thermal noise in the JFET channel can be modeled by a gate series resistance with the value $R_n=2/3G_{fs}$. Its noise voltage can be calculated in the usual way $-E_n=\sqrt{4kTR_n}$. If transconductance drops to half its value with V_{ds}, then the noise increases by 3dB.

Another important question comes to mind. Distortion reduction mentioned above is achieved for small input voltage 10mV rms. What is the distortion at high input signals? **Figure 10** illustrates distortion curves for the different devices with the same part number. Some JFETs have V_{ds} set near optimal value for minimum distortion, some of them have a high V_{ds} voltage. At low input level there is 15dB to 20dB improvement in distortion, and this improvement reduces with level. If we draw a line tangent to each curve and measure its slope, then we get a sense of the steepness. The slope of the THD curve for the devices in saturation region is about 1 (linear) and the slope for the THD curves for devices in the variable resistance-mode is about 2 (square). That means that a JFET in saturation mode produces mainly second-order harmonic while a JFET in the variable-resistance mode with drain resistor and optimum V_{ds} value produces mainly third-order harmonic.

This method for linearization of the JFET (MOSFET) transfer function can be used in the circuits, which are optimized for single-ended amplifying devices and when the application of a differential pair is not straightforward.

Next, I will describe a practical circuit, built around JFETs in the variable resistance-mode with a drain load resistor (**Photo 1**).

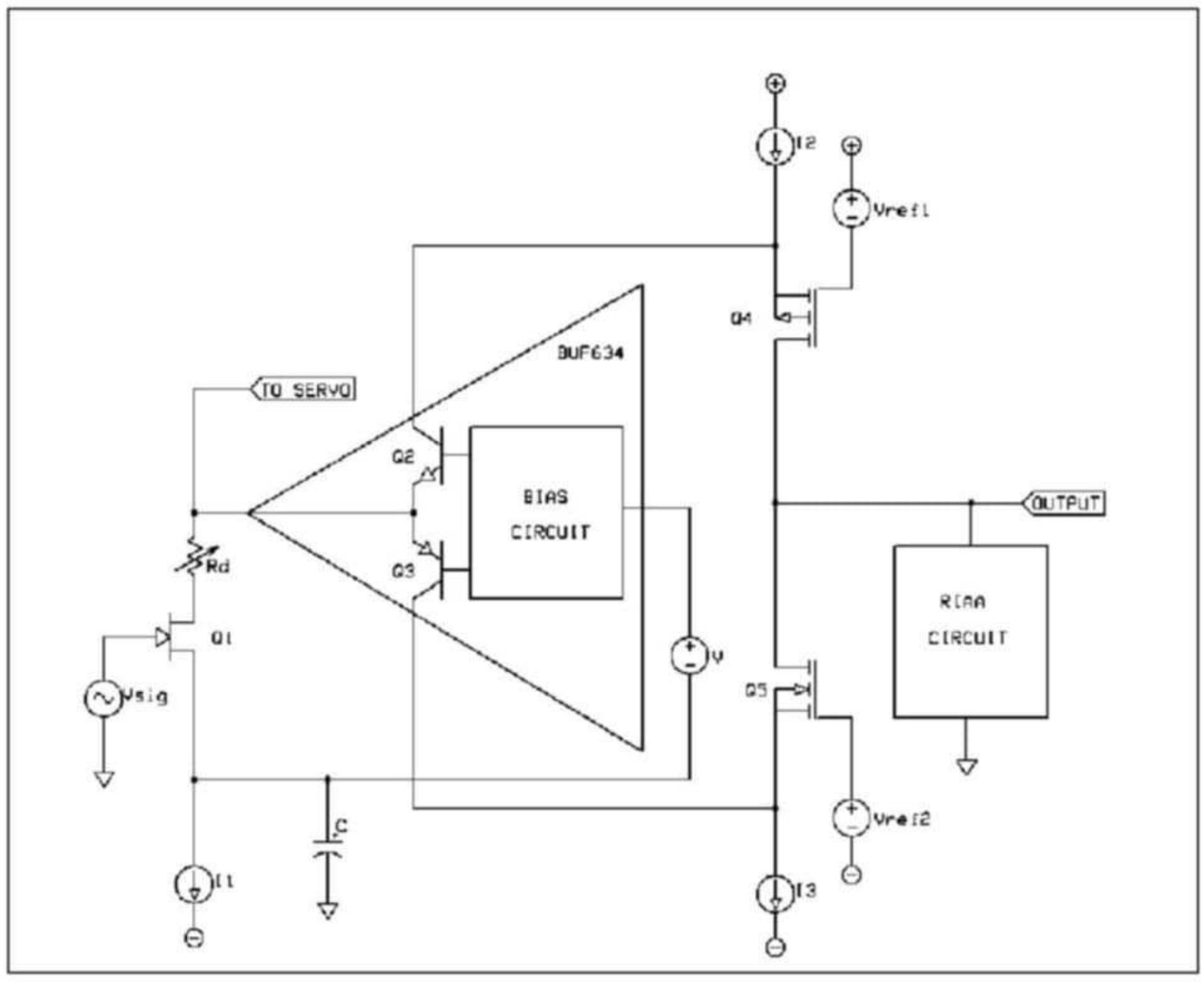


Figure 11: Simplified diagram of the transconductance phono stage

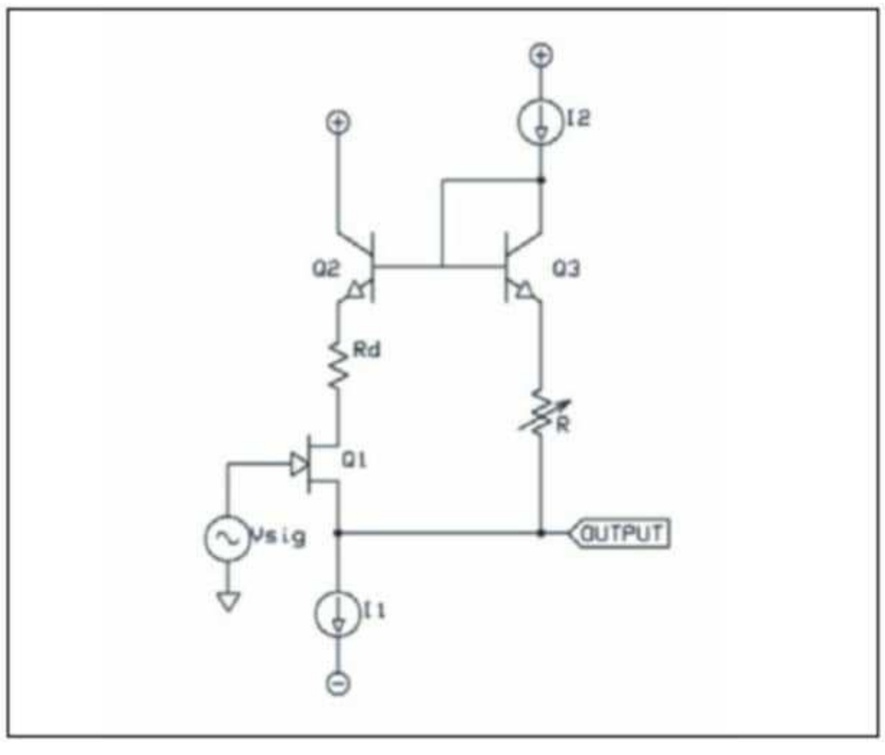


Figure 12: Simplified diagram of the output unity gain buffer

Figure 13: A fast Fourier transform (FFT) of distortion spectrum of standalone output unity gain buffer, shown in Figure 12. Load 100kΩ, output 1V rms, fundamental 1kHz.

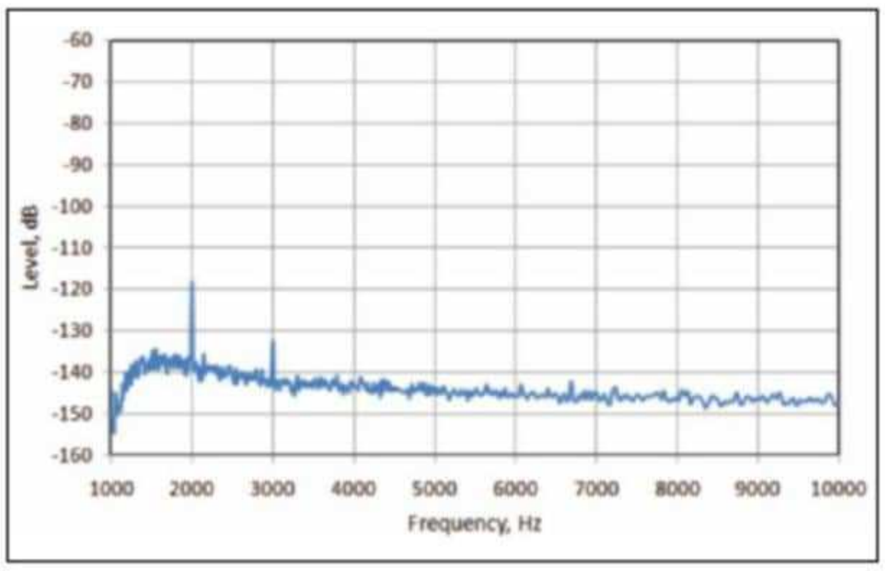
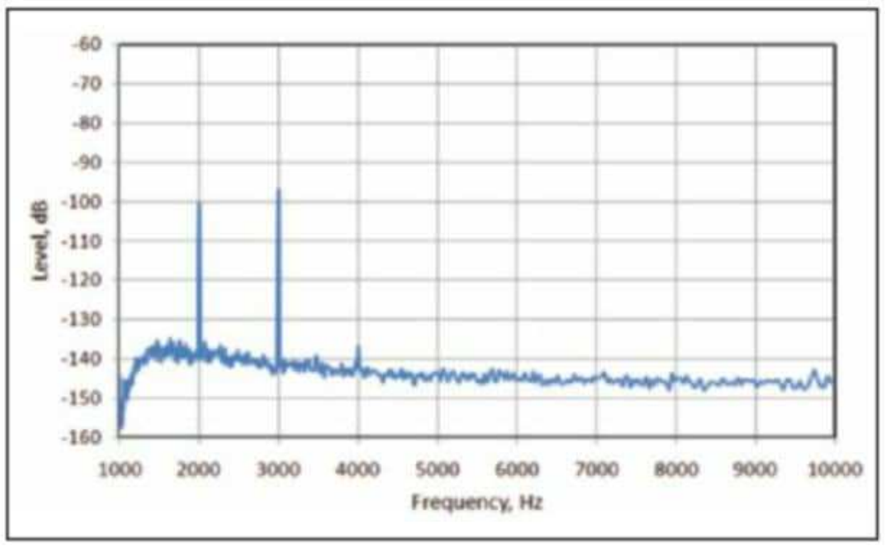


Figure 14: A FFT of distortion spectrum of standalone output unity gain buffer, shown in Figure 12. Load 600Ω, output 1V rms, fundamental 1kHz.



Transconductance Phono Preamplifier

Transconductance phono preamplifiers have been well known for years. I can mention the following designs (listed in random order): Dr Malcom Hawksford's circuits, Pink Triangle PIP phono by Owen Jones, Jonathan Carr's circuits, Creek OBH phono preamp by Alex Nikitin, Audio Interface Resources (AIR) preamplifier by Lars Clausen and John Madsen, and Paradise by Joachim Gerhard. These designs include a high transconductance amplifying stage loaded with a two-terminal RIAA network.

The amplifying stage output current passes through the RIAA network. The voltage across the RIAA network decreases with frequency according to the RIAA equalization curve. This voltage drop is limited only by amplifying stage peak output current. Transconductance phono preamplifiers usually have larger headroom (at least it is not limited by supply rail voltage).

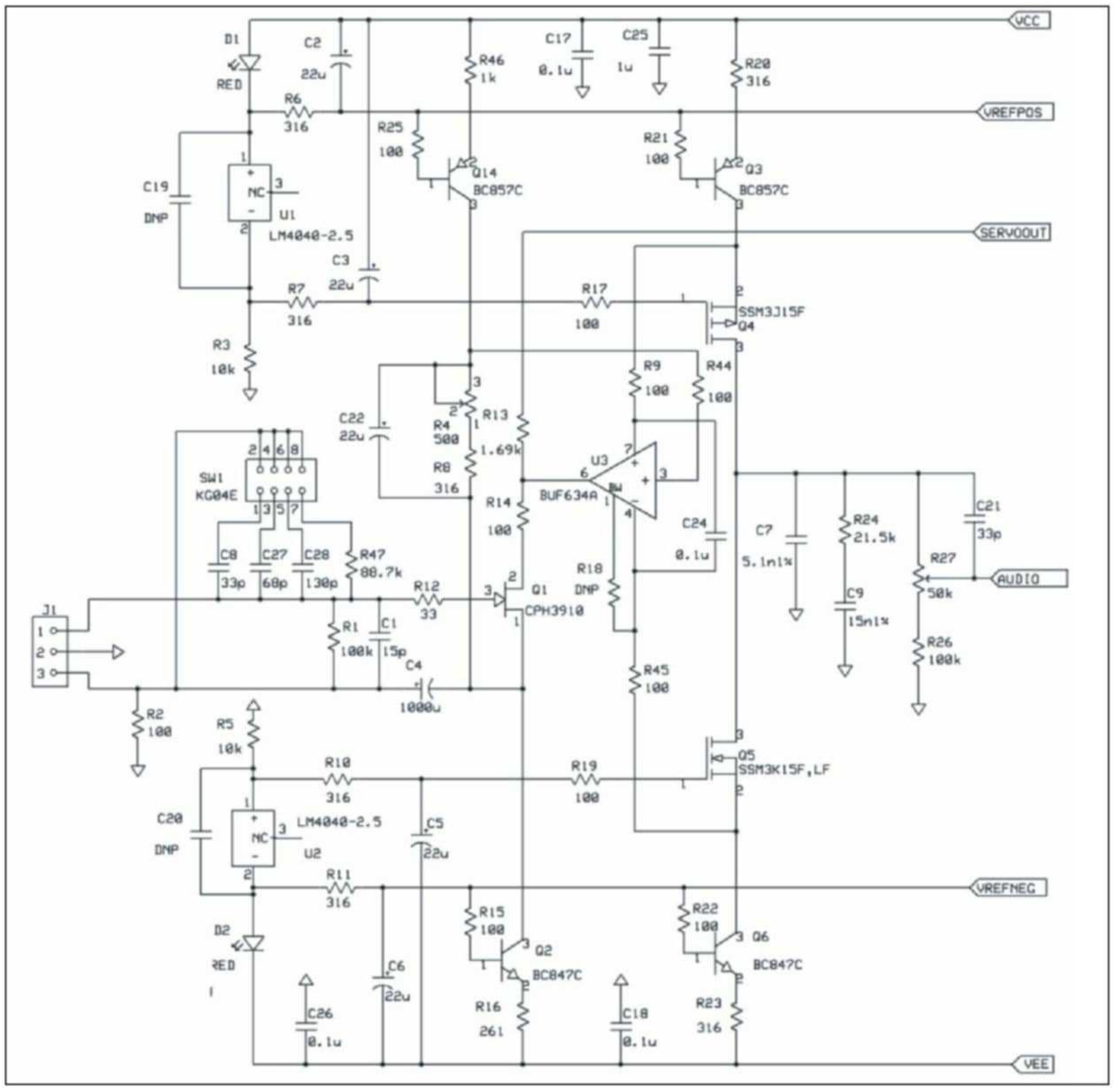


Figure 15: The full schematic of the transconductance phono preamp input stage and RIAA network

The Circuit. The basic topology is a zero global feedback concept based on a single amplifying JFET stage with current driven two terminal RIAA network as shown in **Figure 11**. Constant current source (CCS) I1 sets the operating point of the input common source stage (Q1) for 2mA. The input JFET has series drain resistor Rd. The output current of the input stage feeds the output terminal of the integrated buffer (BUF634A).

The floating voltage source together with the integrated unity gain buffer sets V_{ds} for the lowest distortion in the input stage. Output devices Q2, Q3 of the buffer transfer Q1 output current to the sources of the folded cascode stage (MOSFETs Q4 and Q5).

Current sources I2, I3 set the cascode devices quiescent current. The folded cascode stage has a high output impedance and is loaded with the CRCR RIAA two terminal network. The JFET unity gain buffer shown in **Figure 12** isolates the RIAA network from the receiving equipment.

CCS I1 sets the operating point of the common drain stage (Q1) for I_d=20mA, the voltage across Q1 and Rd is equal to the voltage drop on trimmer R. This voltage drop is set by current source I2 and the value of the trimmer R. Trimmer R is adjusted for lowest distortion.

Transistor Q3 operates as a diode and compensates for the temperature variation of the base-emitter voltage of emitter follower Q2. The distortion spectrum of this unity gain buffer is shown in **Figure 13** and **Figure 14**. You can see that only the second and third harmonic are above the noise. This is remarkable performance for such simple circuit.

Schematics. The schematic of the complete transconductance phono preamplifier is shown in **Figure 15** and **Figure 16**. The preamp can be powered from a bipolar 15…18V supply. Red LEDs

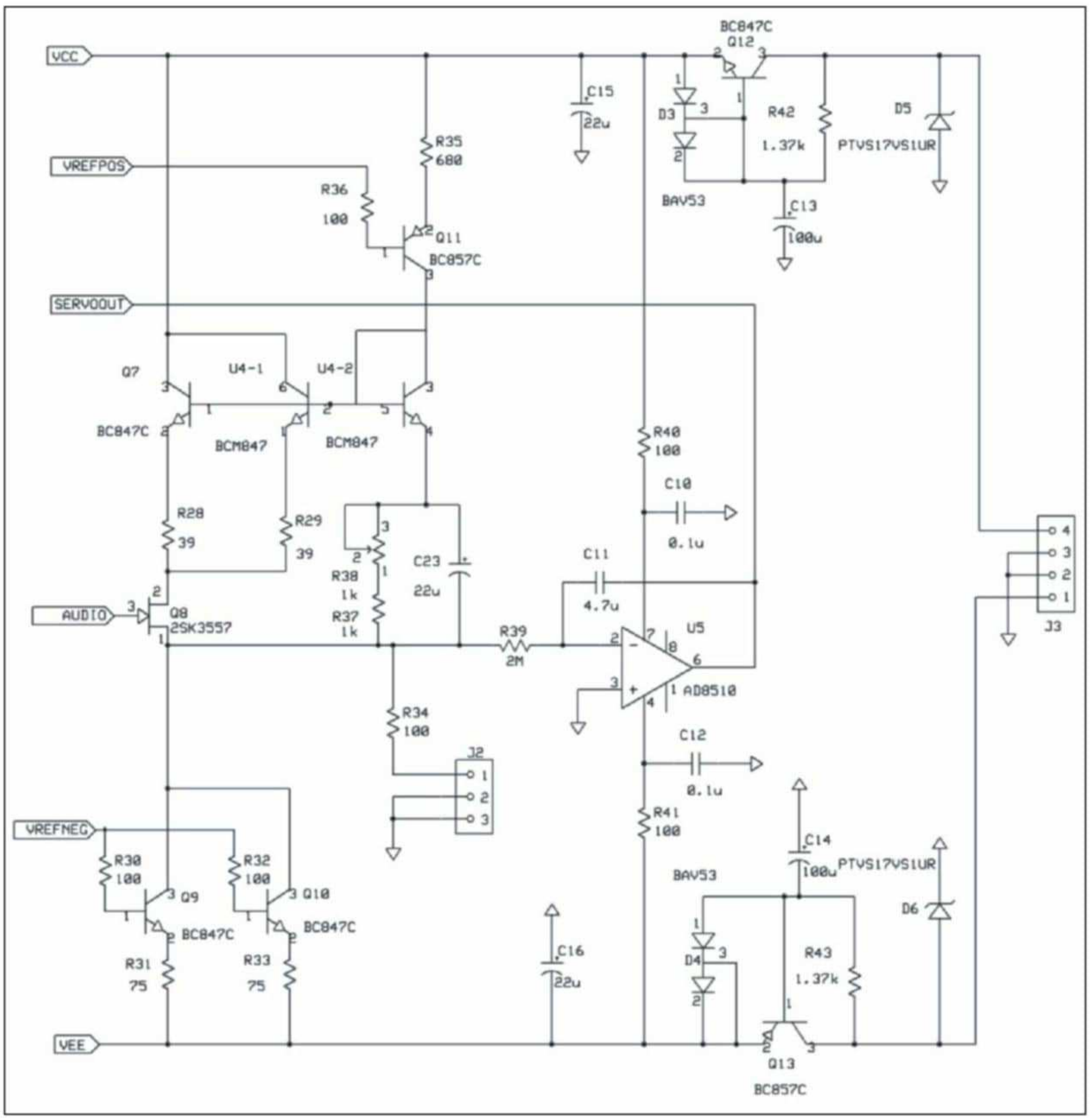


Figure 16: The schematic of the transconductance phono preamp output buffer

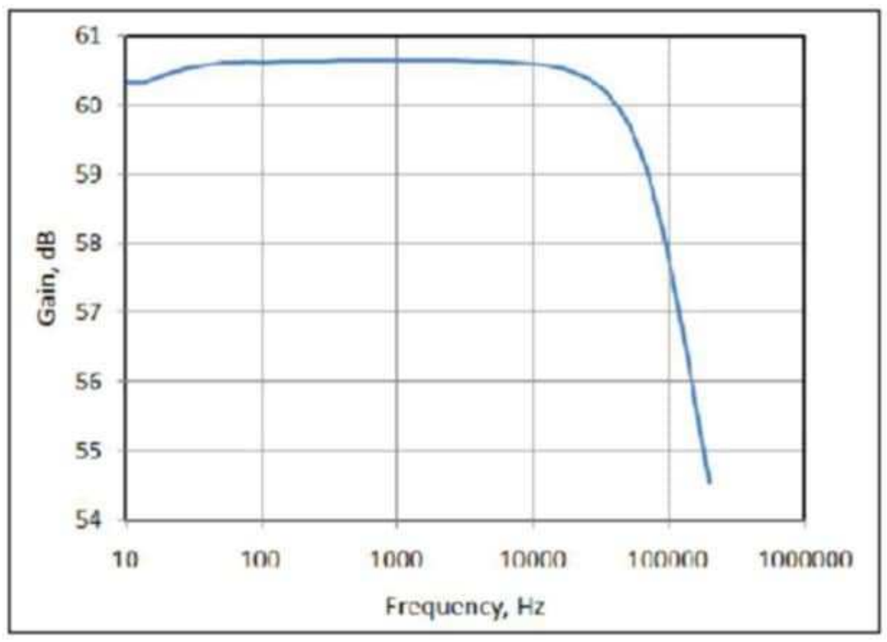


Figure 17: Frequency response of the transconductance phone preamplifier without RIAA equalization network components C7, C9, R24. Input JFET transconductance 6.7mA/V, load 150kΩ (R26+R27).

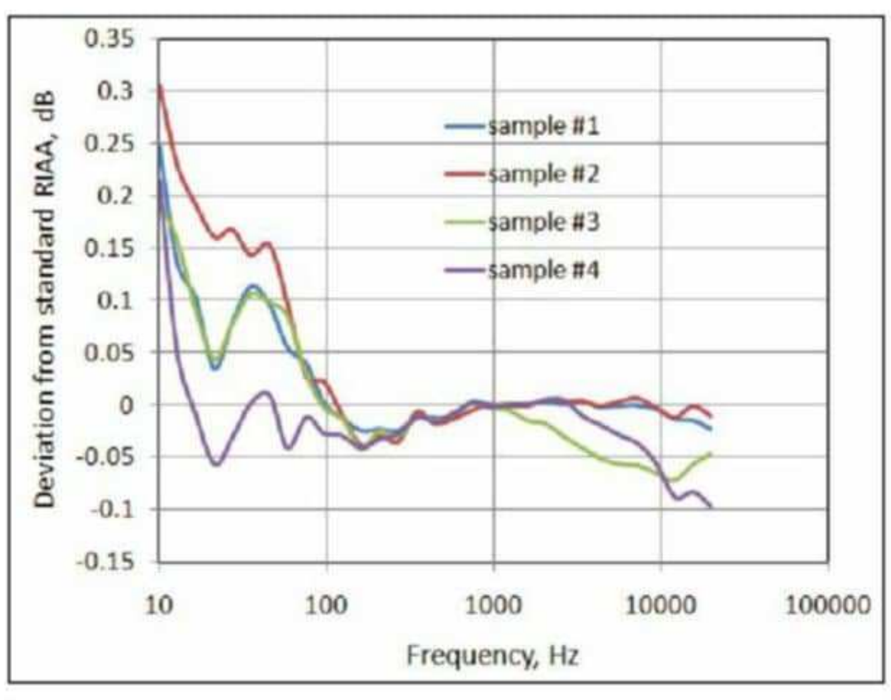


Figure 18: Deviation of the frequency response of the transconductance phone preamplifier from the RIAA curve

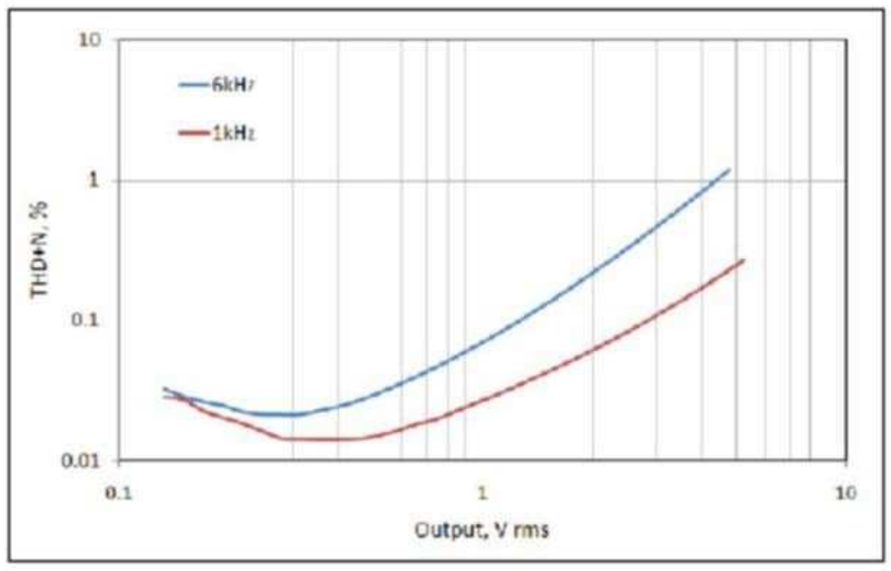


Figure 19: THD+N for the transconductance phono preamplifier for 1kHz and 6kHz sine wave, output voltage in Volts rms.

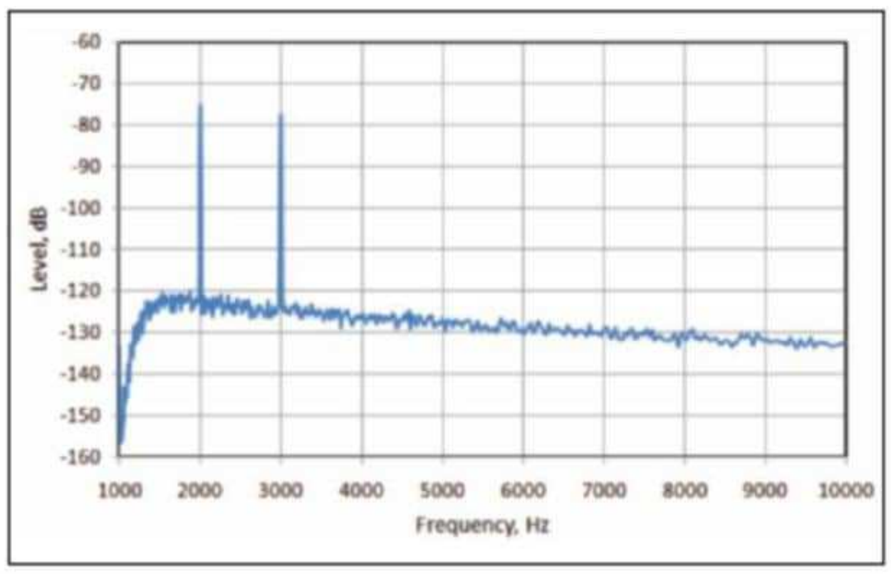


Figure 20. The FFT of the distortion spectrum for the transconductance phono preamplifier, output 1V rms, fundamental 1kHz

About the Author



Dimitri Danyuk has been the principal hardware engineer at Harman luxury audio since 2013. He graduated with honors from Kyiv Polytehnic in 1985. His passion is to help audiophiles in all aspects of audio design. His articles have appeared in a number of magazines including *Journal of the Audio Engineering Society* (JAES), *IEEE Transactions*, *EDN*, *Electronic Design*, and more. Dimitri also provides consulting services for various businesses.

D1, D2 act as voltage references for the current sources Q2, Q6, Q9, Q10, Q14, Q3, Q11. Shunt voltage references U1, U2 provide gate voltage for the folded cascodes Q4, Q5. The reference voltages pass through corresponding low-pass RC filters. The MM pickup is connected between pins 1 and 3 of the input connector J1.

A ground lift switch can be connected to pins 2 and 3 of the same connector. DIP switch SW1 allows choosing different cartridge loads: 47kΩ and 100kΩ; 33pF, 68pF, 100pF, 130pF, 160pF, 200pF, and 230pF. Q1 is the input JFET in common source configuration. Q1 works in variable-resistance mode with a small drain resistor R14. Q1 is the single amplifying device in the whole circuit. Trimmer R4 sets the voltage between drain and source. Integrated open-loop buffer U3 passes Q1's output current to the sources of cascode devices Q4, Q5. The MOSFETs are chosen as cascode devices due to their low output transconductance. The folded cascode stage is loaded with a CRCR RIAA two terminal network. Trimmer R27 allows altering the total gain within ±1.5dB to compensate for Q1 transconductance spread.

A source follower Q8 isolates the load from the high impedance RIAA network. There are two current sources in the source network (Q9, Q10) to keep device dissipation within limits. There are also two paralleled drain devices (Q7, U4-1) for the same reason. Trimmer R38 sets the voltage drop between the source follower source and drain. The load is connected at J2. Inverting integrator U5 provides the necessary DC voltage to the first stage. There is a capacitance multiplier filter in each power rail (Q12, Q13) for removing power supply residual hum. Transient voltage suppressors D5, D6 protect from overvoltage and reverse power. Schematics and PCB files, the BOM and Gerber files for the preamplifier can be found in the Supplementary Materials section of the *audioXpress* website.

Test Results. The measurement results of the complete phono preamp are shown in **Figures 17–21**. The noise level (60dB gain, no RIAA network) is 360µV in a 22kHz BW or 2.5nV/√Hz.

Adjustment. The adjustment is simple—set minimum distortion at low output level (100mV) by trimmer R4, set minimum distortion at higher output level (1V) by trimmer R38, make the output signals equal with trimmer R27.

Conclusion

Further improvements are possible. The distortion is limited by the performance of the input stage. Distortion can be reduced by setting the drain current in the first stage to a higher value.

The distortion minimum becomes wider with higher drain current (less dependent on V_{ds}). The value of the drain resistor and drain to source voltage should be adjusted accordingly.

It is pure fun to build and test a phono amplifier with a single amplifying device. I encourage readers to play with this solution and try it in other circuits. For example, consider a common gate I/V converter for a DAC. The input resistance of a common source stage is not linear (reduced with current). A small resistor in the drain network together with optimal V_{ds} improves linearity of the input resistance. Other components can be used instead of R_g (e.g., a JFET with shorted gate and source). ax

Author's Note: I'd like to express my sincere thanks to Marcel van de Gevel for his valuable feedback, comments, and suggestions.

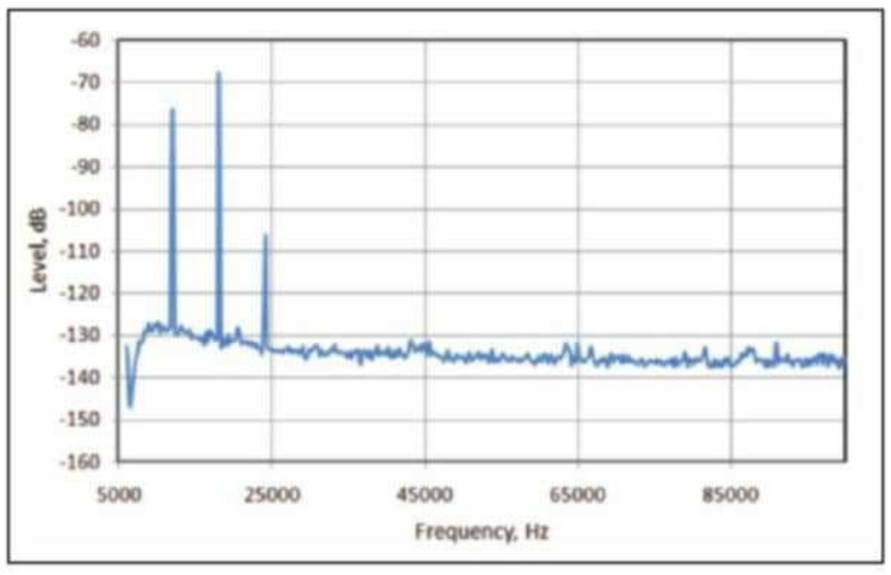


Figure 21: The FFT of the distortion spectrum for the transconductance phono preamplifier, output 1V rms, fundamental 6kHz

Project Files

To download the supplementary files for this project, visit: http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Resources

A. Nikitin, "Parametric distortion compensation in FETs or How to build a Solid-State Triode," www.ant-audio.co.uk/Theory/Parametric_Linearisation.pdf

N. Pass, "The Sweet Spot," *audioXpress*, January 2010, https://audioxpress.com/article/The-Sweet-Spot/1805, https://audioxpress.com/article/The-Sweet-Spot/1806

J. S. Sherwin, "Knowing the cause helps to cure distortion in FET amplifiers," *Electronics*, December 12, 1966, pp.99-105.

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You Can DIY!

eDD — A Direct-Drive Amplifier for Electrostatic Speakers

By
Jan Didden



Electrostatic speakers require drive signals different from electromagnetic speakers. Where electromagnetic speakers require up to 100V RMS at up to 10A RMS drive levels, electrostatic speakers require up to 5kV RMS, balanced, at up to 100mA RMS. This project describes a well-performing high-voltage amplifier to directly drive electrostatic speakers.

Most audio power amplifiers are unsuited to directly drive electrostatic speakers. Therefore, step-up transformers, with step-up ratios from 1:50 to 1:200, are used to couple the electrostatic speaker load to the audio power amplifier. Transformers with such high step-up ratios for the audio bandwidth are specialty items, difficult to design and expensive. The capacitive load of an electrostatic speaker, transformed down to the output of the audio power amplifier, presents a difficult load that not all amplifiers can handle. Filters and impedance networks are used between the speaker input and the step-up transformer to make the amplifier load less reactive and maintain a flat enough impedance. This makes driving an electrostatic speaker complex and expensive, and only a subset of available amplifiers can be used successfully. One needs a high-power amplifier with complex load stability and ample output current reserve.

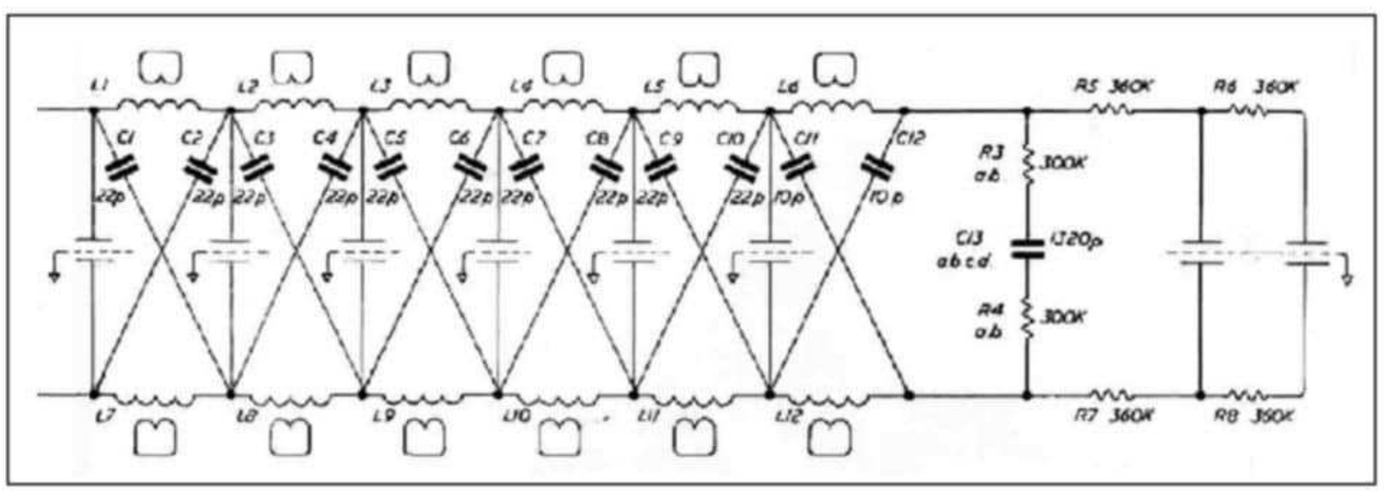


Figure 1: ESL63 transmission line circuit (see text)

But first, we should make the distinction between single panel and segmented panel ESLs. Single panel ESLs as built by MartinLogan and others are basically a diaphragm between two stator plates, a physically large capacitor, with a large capacitance between the stators. Designing a direct drive amp for such a speaker may require up to 100mA rms current or more at full power and high signal frequency.

At the other end of the spectrum are the segmented ESLs, where parts of the diaphragm are segmented in separate areas or strips, either mechanically by clamping or electrically with the segments connected via R-C networks or transmission lines. Which segments are active and emitting sound at any one time depends on the signal frequency. Segmentation is done to control high-frequency beaming of the speaker, but such speakers also present a more benign load to the step-ups, and a high-voltage direct drive would require maybe up to 20mA rms, and that is much easier to design and build. My direct drive amplifier has been designed with the QUAD ESL63 and its derivatives in mind, with a design goal of 1500V rms at 20mA rms over the full audio band.

The obvious question is why there are so few commercial amplifiers to drive electrostatic loudspeakers directly, to circumvent the complexity and cost of the step-up transformers, filtering components and stringent power amplifier requirement. There have been a few developments

in the past but these direct-drive amplifiers were complex to be able to manage the very high voltages, therefore expensive, and often unreliable. Due to limited experience with high voltage amplifiers at service centers, it was hard to get a defective direct-drive amplifier repaired reliably and affordably. Most direct-drive amplifiers offered on the Internet are defective in one or both channels.

Design Considerations

The QUAD ESL63 is an electrically segmented ESL, meaning that the different audio frequency ranges are routed to different (circular) segments of the diaphragm. **Figure 1** shows the circuitry of an ESL63, from the output of the step-up transformer at the left side, to the stator elements. The dotted lines in the middle with the bent arrows are the diaphragm segments. This is a transmission-line like structure, and the segments are circular in shape. **Figure 2** shows the construction.

This setup creates a sound field that appears to originate from a point source behind the speaker [1]. In general, an ESL acoustic output on axis increases with frequency, so the drive signal is often low passed to maintain level output. In the ESL63 the active segments decrease in surface area with frequency, and if done right, the whole setup can be driven with a flat frequency response for a flat on-axis SPL response. This type of load is much less capacitive than an unsegmented speaker and does not require large output currents at high frequencies. **Figure 3** shows the stator input impedance of the ESL63 over the audio frequency range, roughly between 300kΩ and 400kΩ.

I decided to use a bipolar supply for the amplifier. My target was 1.5kV rms out, necessitating a 4.4kV total supply voltage. That is scary to have in an amplifier box, so I decided to go for ±2.2kV. It does complicate the supply but should help with reliability (and still is scary!).

From the beginning I have been weary of stability issues for an amplifier with a dominantly capacitive load. (I only later decided to target the QUADs, which have more benign load characteristics.) It seemed that the best approach would be to design a current output amplifier that would give the amplifier-capacitive load combination a first order low pass response. That means I could wrap feedback around the amp to lower distortion and set the output impedance to whatever I wanted without having to worry about stability. This also appeared to combine well with the fact that an ESL SPL is linearly related, first order, to the stator current, not voltage, as derived by Peter Walker in his famous “Walker equation” [2]

I built some prototypes using high voltage MOSFETs and IGBTs output devices. These worked fine—if the frequency remained below a few kilohertz. These high voltage devices can withstand up to 4.5kV and conduct several amps but are basically designed for switching applications. They have very large and nonlinear inter-electrode capacitances that vary with voltage levels. In switching applications that is no issue, you just drive them as hard as needed. But their internal (nonlinear) capacitances are so large that they totally dominated my amplifier load, being several times as large as the speaker they were intended to drive.

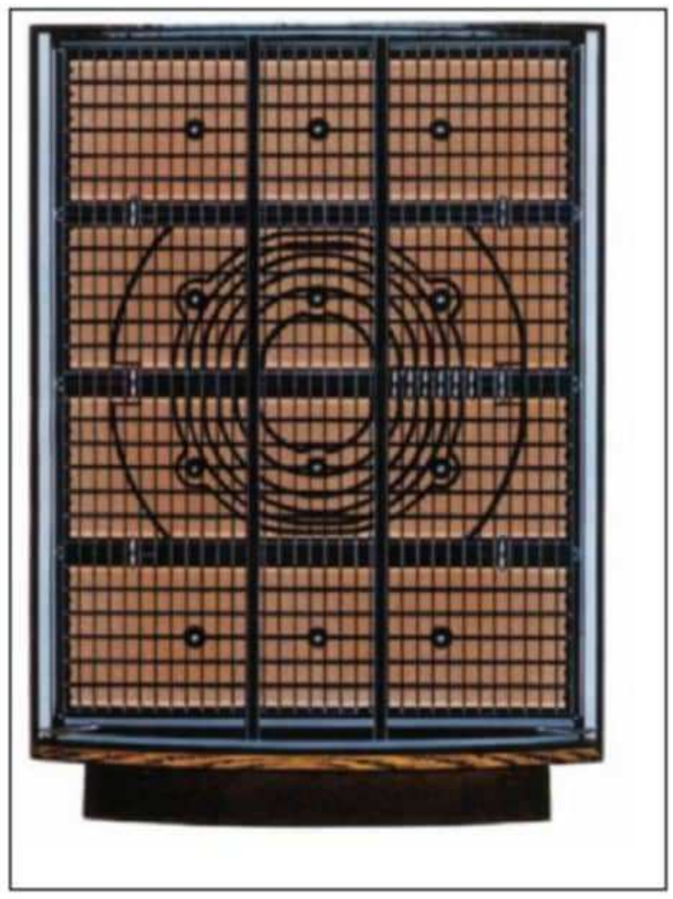


Figure 2: ESL63 construction showing the circular segments

An additional complication was that those switching high-voltage devices are not suited for linear operation due to the Spirito effect, which has a similar destructive effect as secondary breakdown in a bipolar power device [3]. After discussing the problem with some of my audio friends, one asked if I had considered using tubes as output devices. This was a good friend so I forgave him for his impertinence, but he got me thinking. I got myself a couple of 6HS5s, unsoldered the IGBTs in the prototype output stage, connected the tubes with some external heater supplies, switched on the prototype and all my issues were gone! My circuit design was sound, I had just used the wrong output devices. It was all downhill from then on.

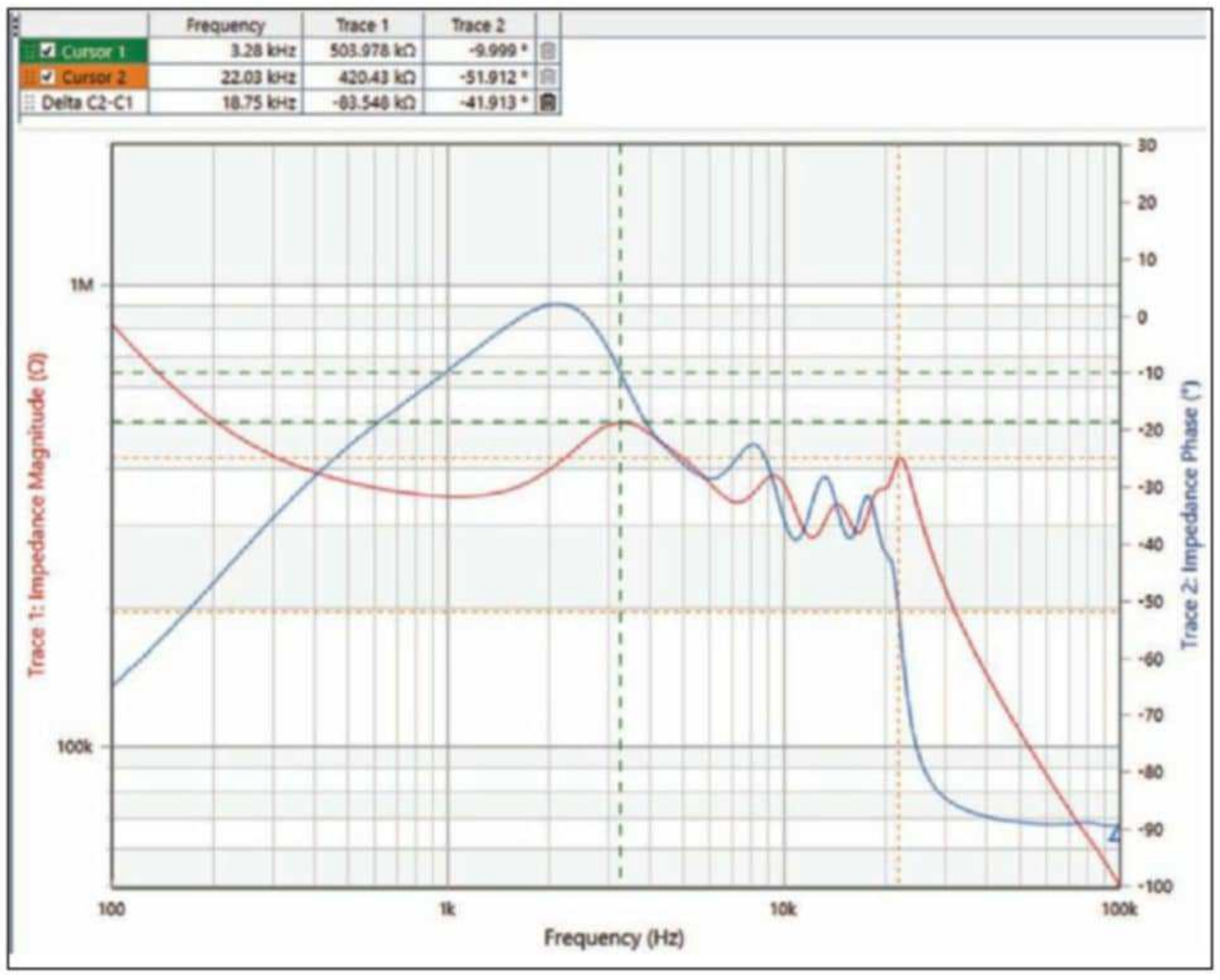
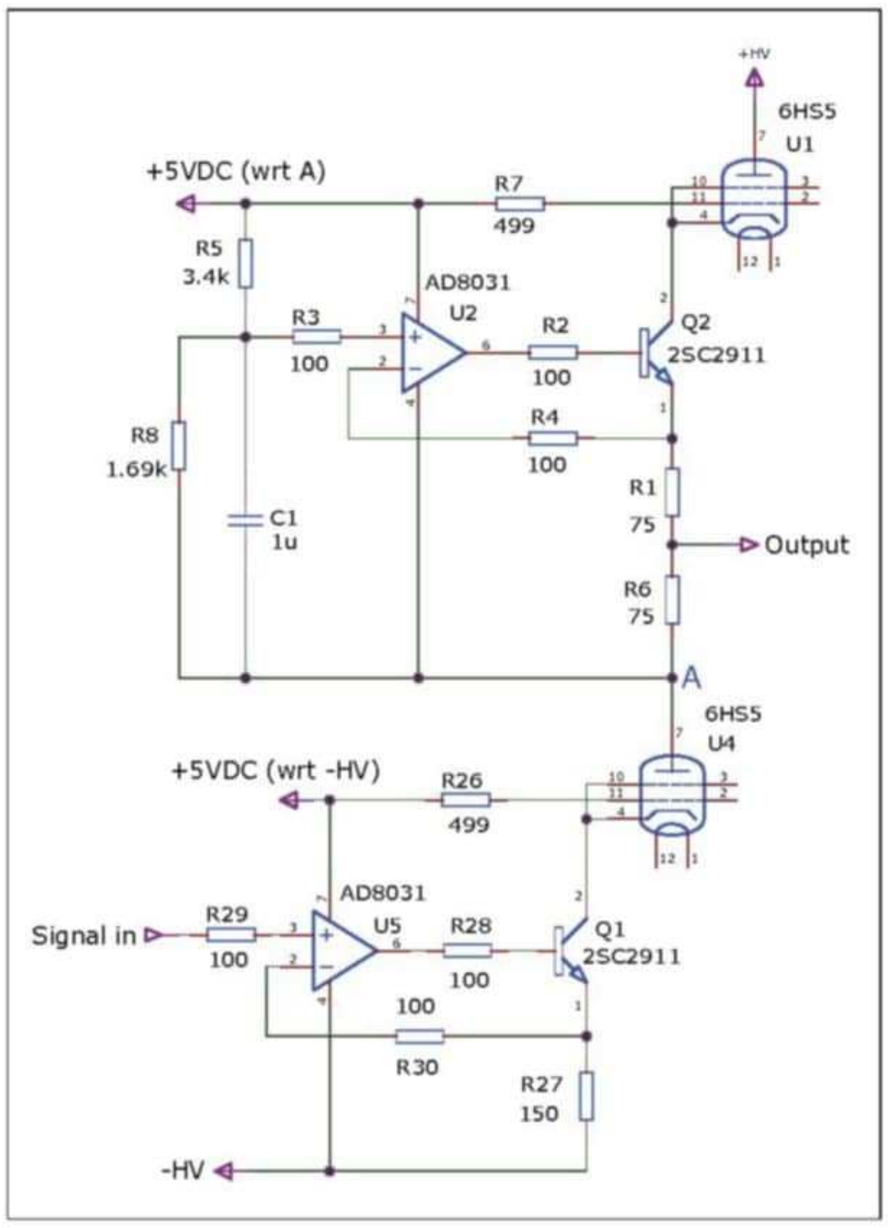


Figure 3: ESL63 load impedance

Figure 4: Basic output stage circuit



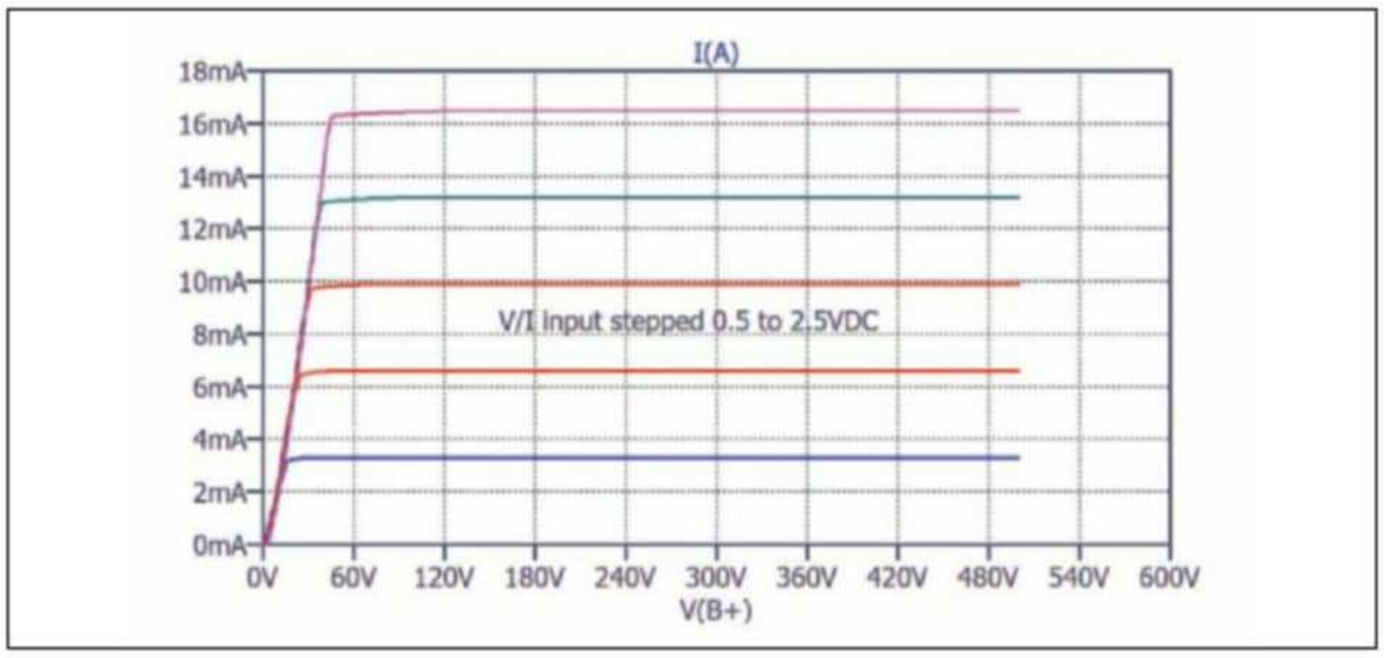


Figure 5: Grounded grid triode output curves (see text)



Photo 2: The high-voltage amp PCB, top side

The Output Stage

The basic output stage design is shown in **Figure 4**. The bottom tube U4 is loaded by a constant current source (U1), for a high impedance current output characteristic. Ideally, I would have selected a pentode to get the high output impedance, but that would require an additional floating screen supply. Tube availability in this class of voltage, current and power levels is limited, and I decided on the 6HS5, which is a beam triode.

To give it pentode-like characteristics, I used the tube in common grid, driving it at the cathode from a voltage to current converter. (Viewed from the voltage to current converter you could call it a cascode, but what's in a name?) **Figure 5** shows the output curves for U4 triode-turned-pentode in the Figure 4 circuit, recorded with "Signal in" stepped from 0.5V DC to 2.5V DC in 0.5V DC increments. The curves are nicely flat indicating a very high output impedance.

Next up is the constant current source (CCS) load, U1, another 6HS5. I used the same trick, driving it in common grid so it functions as an almost ideal CCS. This completes the output stage. It runs at around 10mA bias, so the Class-A output current range would be ±10mA peak, slightly less in practice.

Now we apply a little trick that was pioneered by Nelson Pass and goes by various names (i.e., modulated current source or active current source). We do not pick off the signal from the anode of the driver or from the cathode of the CCS, but from a point halfway (R1–R6 in Figure 4). The effect is that when the driven tube U4 pulls more current from the load, there is more voltage across R6. Since the voltage across R1+R6 is constant (R5–R8 divider from the 5V DC bias), the voltage across R1 must decrease which means that the CCS current decreases. This helps U4 to pull down the output. Conversely, if U4's current decreases, the voltage across R6 decreases, the voltage across R1 increases, which means that U1 drives more current into the load. In theory, the result is a symmetrical output current twice the peak value of the simple CCS loading. In practice, the output current is some ±18mA peak, ample for the QUADs.

The Tube Heater and the Bias Supply

Note that for U4, the 5V DC power supply for the grid bias, the heater supply, and the cathode drive op-amp are referred to the -2.2kV supply. For the CCS, the heater, bias, and op-amp supply, which are referred to point A in Figure 4 are swinging with the output signal. So, these auxiliary supplies should have good isolation from the high voltages involved. The swinging supplies for the U1 CCS should also have low capacitances to the rest of the universe to avoid losing load current into those parasitics. The solution

is a pair of medical-grade isolated DC-DC converters. The Traco THM 15-1211 accepts a nominal 12V DC input and transforms that to a 5V at 3A output. They can withstand up to 5kV, and, better yet, have just 20pF between input and output (**Photo 2**).

That 5V DC output is too low for the 6HS5 heater (nominal 6.3V), but the Traco can be trimmed up to 5.7V, and I accepted that the 6HS5 would run at the bottom of its heater voltage range. The tubes do not run anywhere to their specified maximum current, so I felt comfortable with that. For the 5V DC supplies for the grid bias and op-amp supply I used a low dropout low noise linear regulator (ADP7118-5.0) to get that 5.7V DC heater back to 5V DC with very low ripple and noise. **Figure 6** shows that part of the system.

A 12V DC AC-DC converter in the system provides 12V DC to X1, the medical grade DC-DC converter THM 15-1211. The output is trimmed up to 5.7V DC by RU1 and is used as input to the ADP7118-5.0, which provides a regulated low noise 5V DC for the grid bias as well as to the op-amp that drives the voltage to current converter at the cathode. The heater voltage for the tube has a soft start system required to prevent overloading X1 when the heater is cold. Initially, an NTC N1 limits the heater inrush current. After a short while, C21 gets charged to 5.7V DC through R35 and MOSFET Q2 is fully on, with just 8mΩ resistance shorting out N1 so the full 5.7V DC is applied to the heater.

The Transconductance Drive Circuit

This is a straightforward application of an op-amp configured as a voltage-controlled current source, as depicted in **Figure 7**. U5 is a wideband op-amp with rail-to-rail input and output range and able to work on just 5V supply. Its output will drive the base of Q1 until the voltage across R27, fed back to its inverting input, matches the input voltage at its non-inverting input. That makes the current through R27—which is also the cathode current of U4—the input voltage/150Ω. The DC component of the input is about 1.6V setting the DC bias of U4 to approximately 10.7mA.

Output Stage Audio Input

You probably noted that the input to the voltage to current converter driving U4's cathode is also referred to the -2.2kV supply. So, I need to bring the audio signal down from the ground referenced input to the negative supply reference. **Figure 8** shows the circuit. U1 is an analog optocoupler with some unique characteristics. There is a single LED that is optically coupled to two matched photodiodes. Therefore, the two diodes will both conduct the same current from the LED output flux. The two diodes are matched to 0.1%, which is, in distortion terms, better than -60dB.

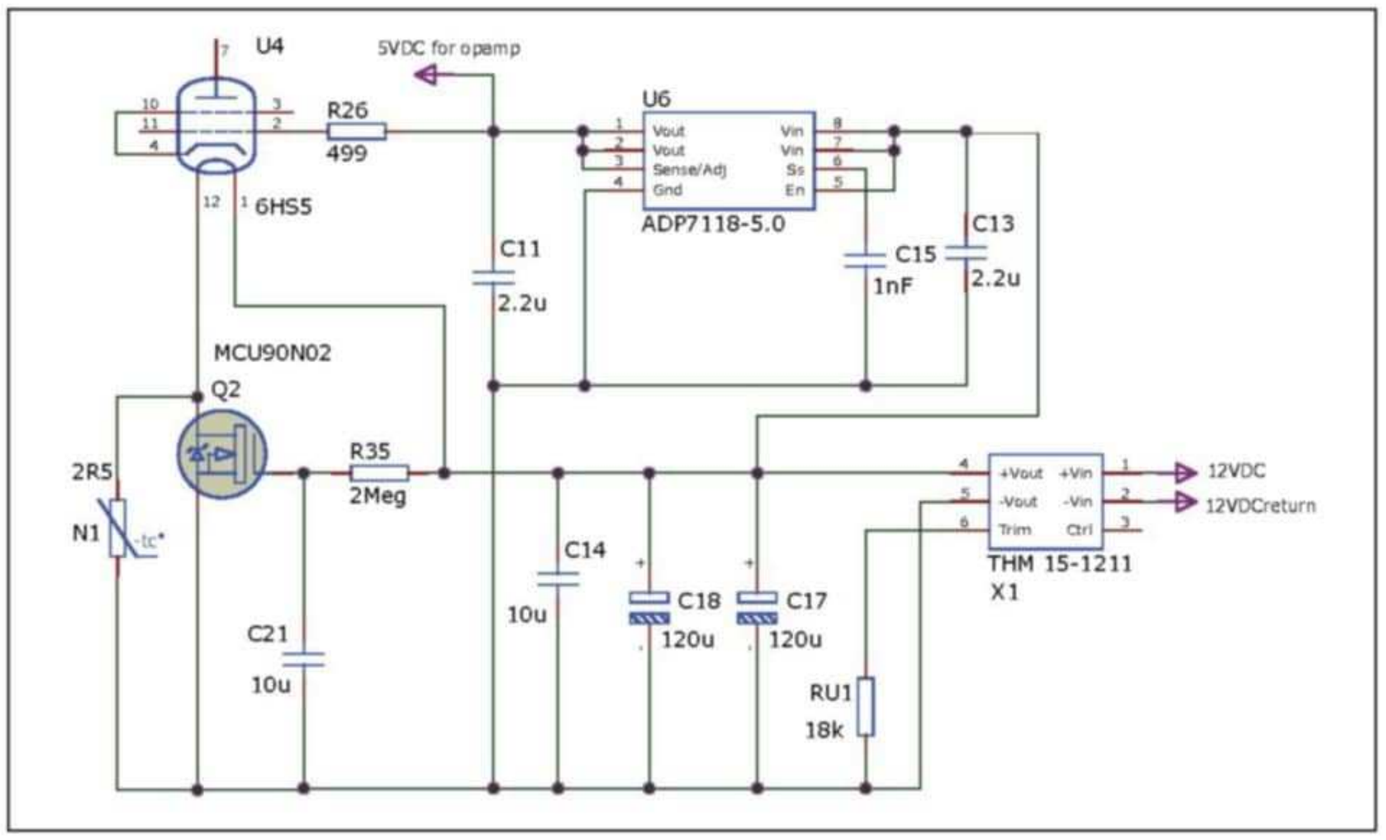


Figure 6: The tube floating bias and heater supplies

Assume that the voltage at pin 5 of U2:B is zero. U2:B then drives the LED in U1 such that the current through R16 is the same as the audio current through R6 but in opposite phase, which would put the inverting input of the op-amp at zero volts as well.

Because of the photo diode matching in U1, the currents through R16 and R3 are identical, so the voltage across R3 will be a faithful representation of the audio input voltage but isolated from the input circuit and ground and referred to the negative high voltage. We assumed that the input at the non-inverting input of U2:B was zero volt, but in reality, it has about 1.6V DC bias from the R12-R13 voltage divider. The upshot is that across R3 you

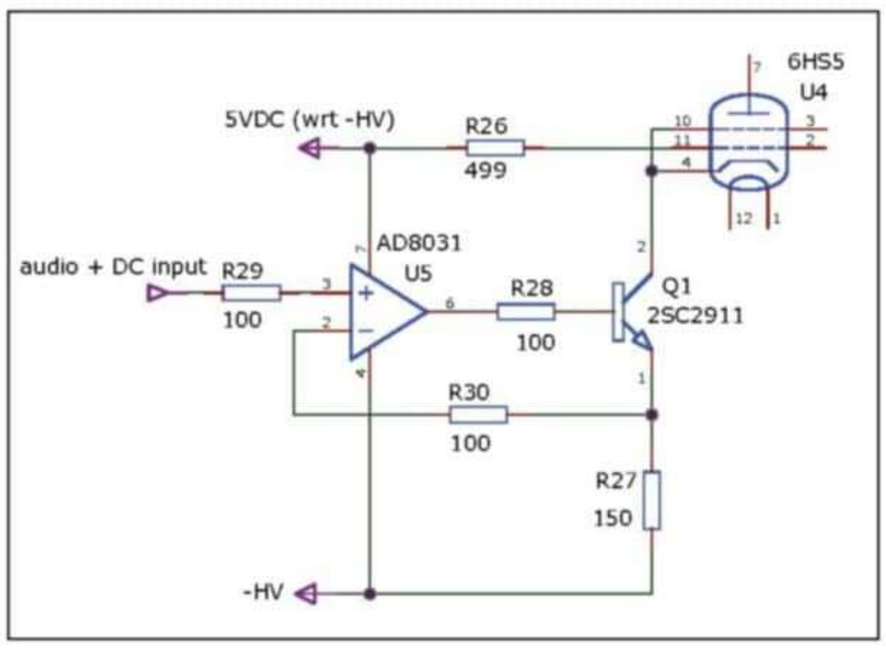


Figure 7: Voltage to current converter cathode driven circuit

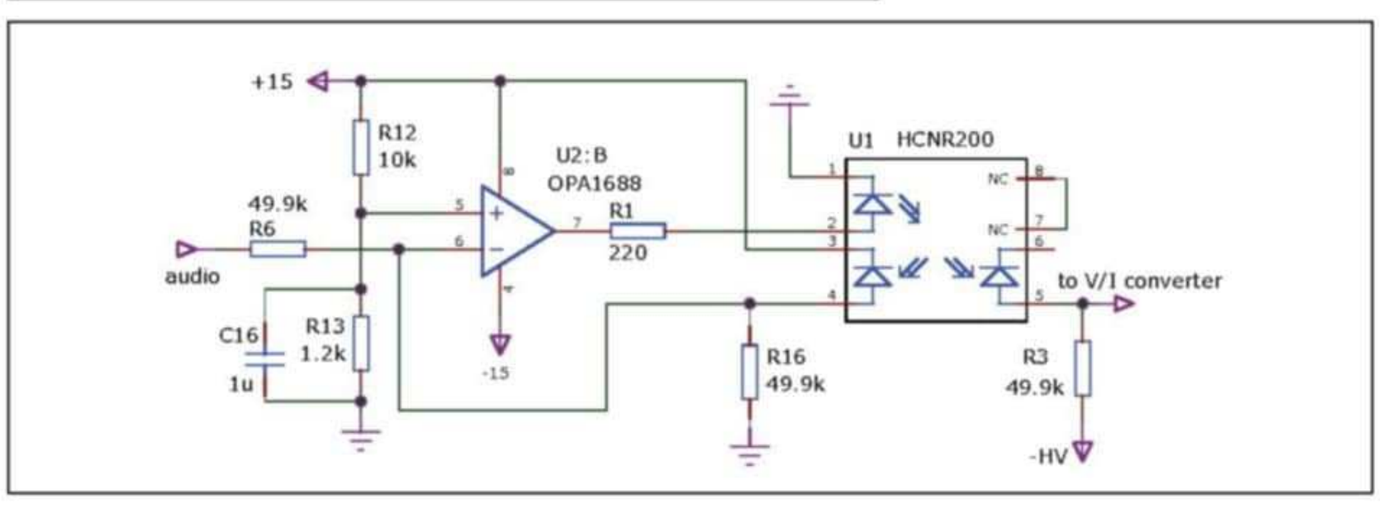


Figure 8: Optical isolator audio translation circuit

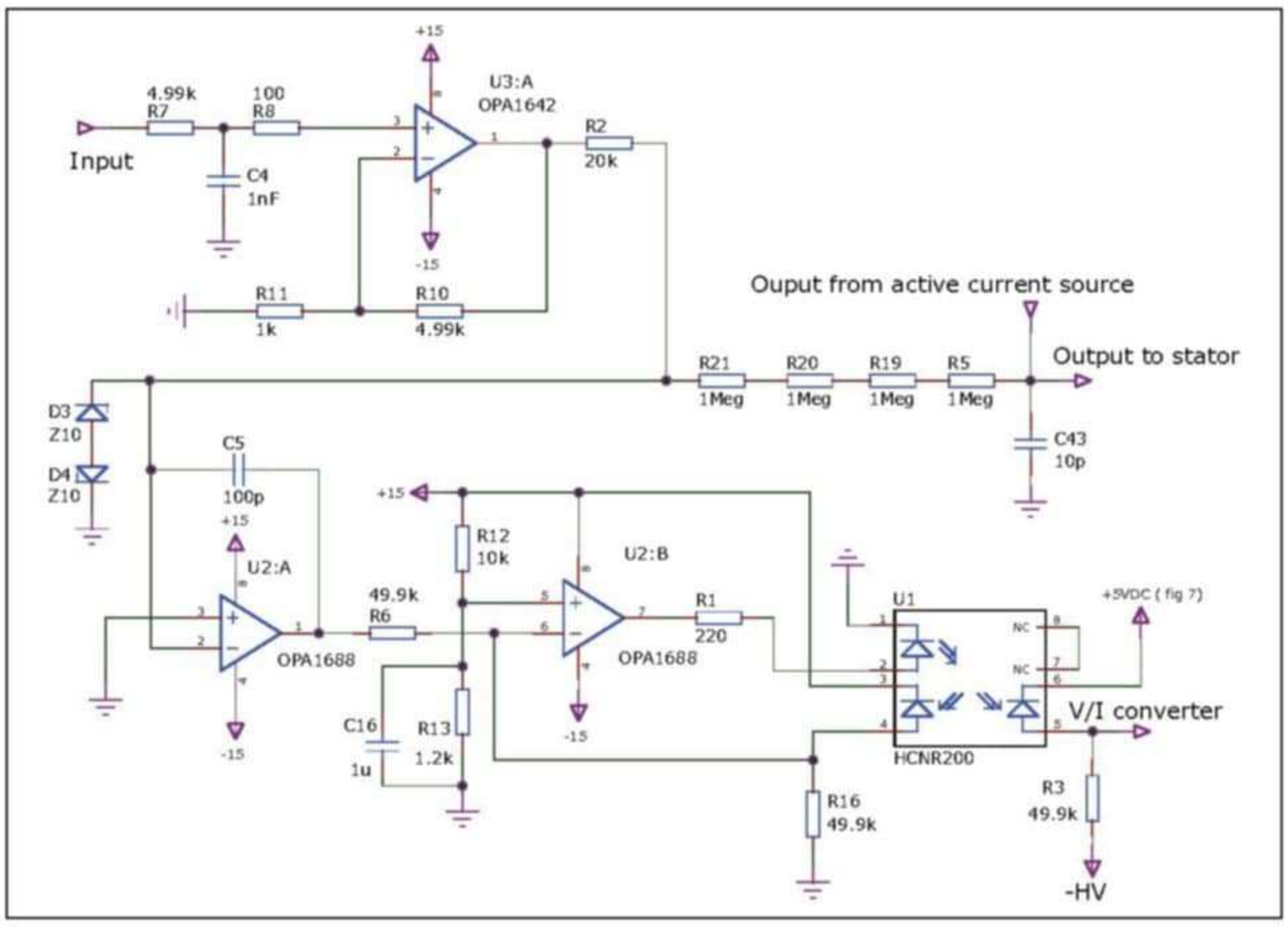


Figure 9: Preamp, feedback, and isolation circuit

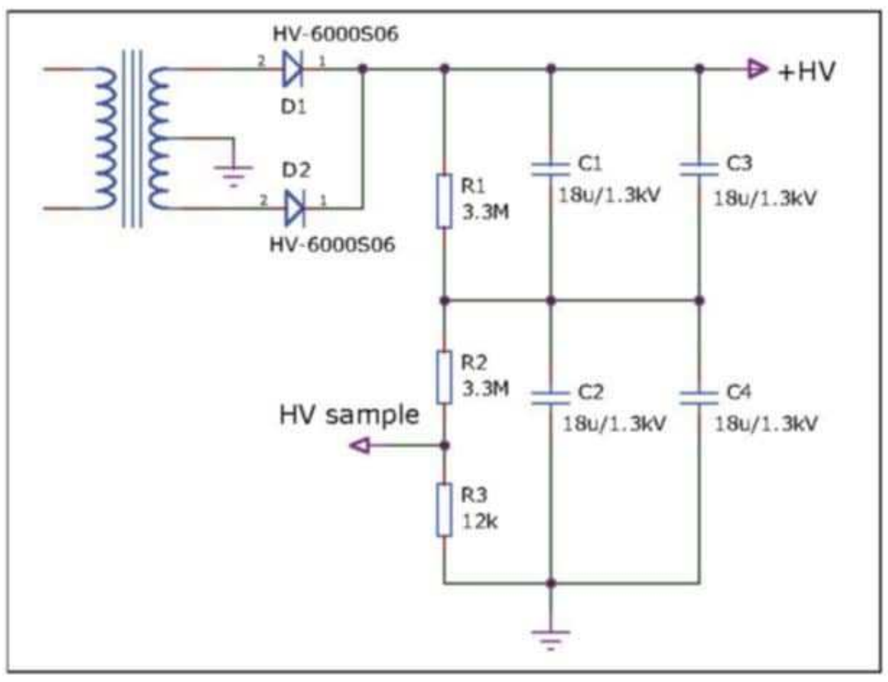


Figure 10: High voltage power supply (pos polarity shown)

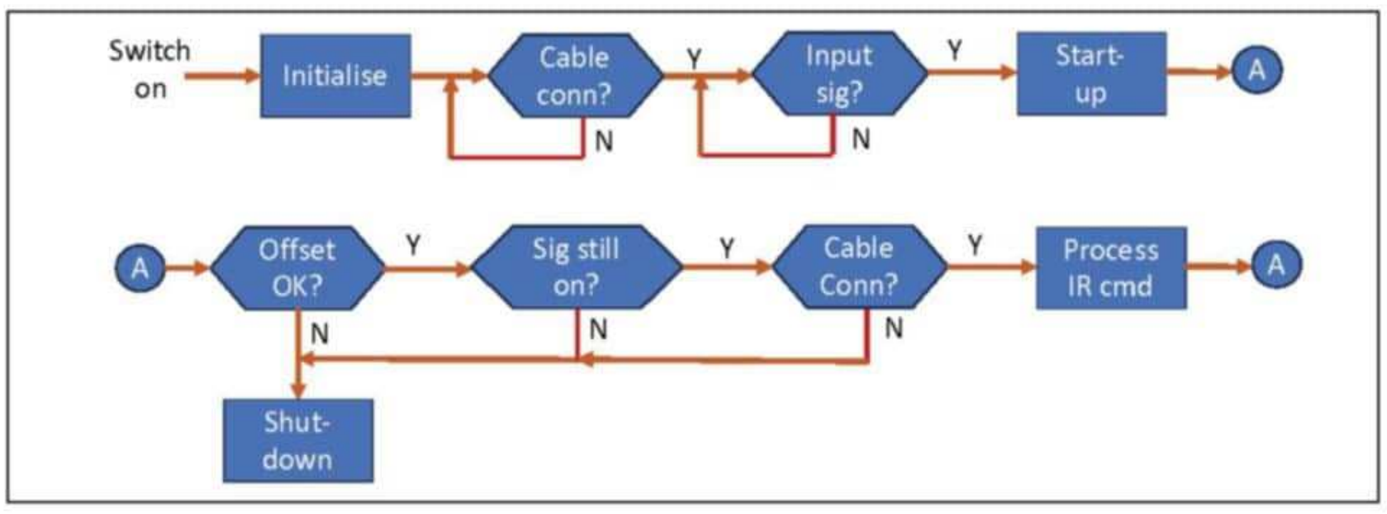


Figure 11: Flowchart 1 Basic control logic flowchart

find not only the audio input signal but also 1.6V DC bias which, as described, sets the bias of U4 to nominally 10.7mA.

Overall Circuit Diagram

Figure 9 shows the amplifier input stage. The forward path contains a gain of six stage, U3:A. The output of this stage drives summing stage U2:A through R2. Summing stage U2:A sums the amplified signal with the feedback signal coming from the output through the high impedance string with the 1MΩ resistors R5-R19-R20-R21. The gain of the output stage is thus 200, set by the ratio of this string to R2. D3 and D4 are there to protect U2:A and U3:A. The output of the summer drives the isolation amplifier U2:B, described earlier, which drives the voltage to current converter driving U4. Since the feedback is DC coupled, this automatically nulls the DC offset at the output. The amplifier output impedance is around 4kΩ over the audio band.

DC output offset in this type of amplifier is a different issue than in regular power amplifiers, as there is no low-impedance load to draw offset current. Rather, an output DC offset pulls the diaphragm off center, and in that manner does decrease the symmetrical output signal available. But to put it in perspective: my amplifier has about 1V DC output offset for a symmetrical output level of ±2.1kV. I do not think it will audibly limit the output.

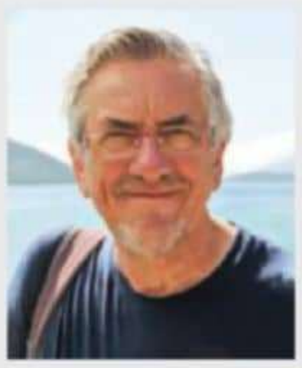
Power Supply System

The amplifier requires several power supplies. I already mentioned the floating tube heater and bias supplies, provided by the isolated DC-DC converters. The isolators are supplied by an AC to DC converter with 12V output at several amps. The control system with the controller (described later) requires a 5V supply. The various op-amps in the preamp and driver stages require ±15V DC. The 12DC, +5V DC and ±15V AC-DC converters are universal mains input (90V to 240V) converters. Where I live, mains voltage is nominally 230V AC. That means that the amp will work fine at half mains input, which allows operation at reduced high voltage through a variac. This was great for my peace of mind during development.

The 5V standby for the controller is always on, but the 12V DC to feed the heater converters is switched on through a solid-state relay, and the DC-DC converter for the ±15V low level circuits supply has a switching input which can be directly controlled from 5V logic. Both are switched on by the controller (described later).

The final stage high voltage DC is provided through a custom high voltage toroid. Menno van

About the Author



Jan Didden has written for *audioXpress* since the 1970s and he is the magazine's Technical Editor. He is retired following a career with the Netherlands Air Force and NATO. He worked in logistics, air defense, and information technology. Retirement has provided him with the time to finish all the audio projects that have piled up for decades. He sometimes writes about them on his website linearaudio.nl. Didden is also known as the publisher and managing editor of the twice-yearly bookzine *Linear Audio*, which he published until 2018. He is currently teaching audio electronics for enthusiast hobbyists and serious amateurs in the Belgian city of Hasselt. More information is available at linearaudio.net

der Veen helped to specify that transformer and I had it manufactured by his TRAFCO partner in Belgrade, Serbia. Output at nominal mains is 3.2kV center tapped for 50mA continuous DC load and it performed exactly as specified. It is a large transformer due to extensive use of isolation material. **Figure 10** shows the positive 2.2kV supply, and the negative supply is similar (with opposite polarity diodes), with resistive capacitor balancing and a tap of the output DC allowing the controller (described later) to sense and display the high voltage DC for measurement purposes. The caps are polyester foil. The high voltage is switched on by the controller through a solid-state relay in its primary feed.

System Sequencing, Control, and Setup

From the beginning of my experiments, I have been cautious when working on the amp. One hand in the pocket, no wristwatch, rubber-soled shoes, the works. I would switch on the next supply only after the previous state was verified as working correctly. At an early stage I decided to include a microcontroller in the system for sequential power-up and health checking. Having previous experience with Microchip controllers I selected a PIC16F15356. I have been using Flowcode [4] (now

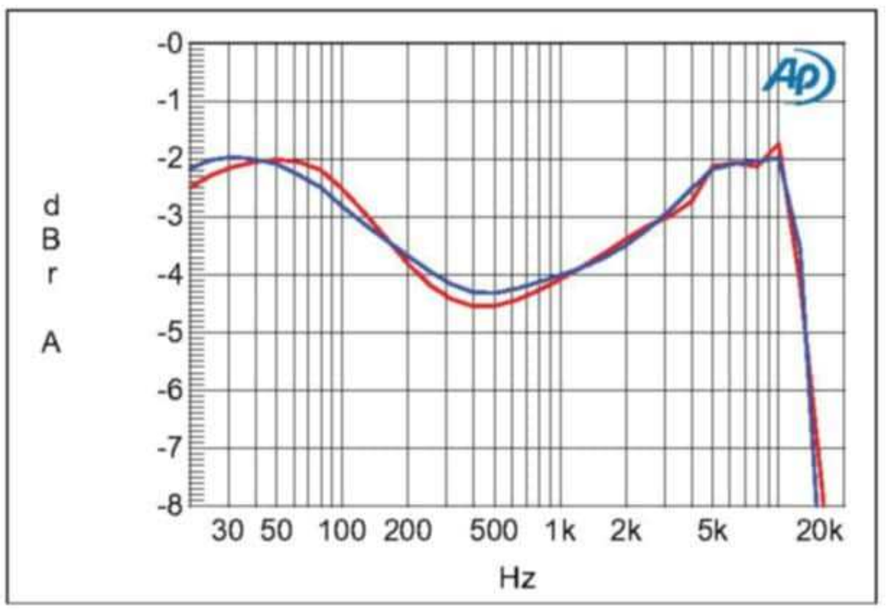


Figure 12: Frequency response at the stators, from the step-up transformer (red) and equalized response from the direct drive amp (blue)



Photo 3: The finished Direct-Drive amplifier (top left) and step-up drive power amplifier



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on version 10) as my programming environment as it is graphically oriented and avoids having to worry about low level routines and libraries. **Figure 11** shows the main logical flow.

There is also an IR receiver in the amp, through which I can make the controller send a command to the speaker through an extra wire in the amplifier-speaker umbilical. There are four high-voltage relays inside the speaker that switch the stators between the direct-drive input signal and the output of the step-up transformers. With that facility, an instant remote-controlled listening comparison is possible. The direct-drive amplifier and the step-up amplifier are each driven from a stereo DAC in an RME ADI-2 Pro FS R. The signal to the direct-drive amp is equalized to match the frequency response of the step-up at the stators and both are level matched. **Figure 12** shows the frequency responses. I spent quite some effort on a special precision, high-voltage, flat response 1000:1 attenuator so I could measure directly on the stators with the Audio Precision, without burning it out with the high voltage.

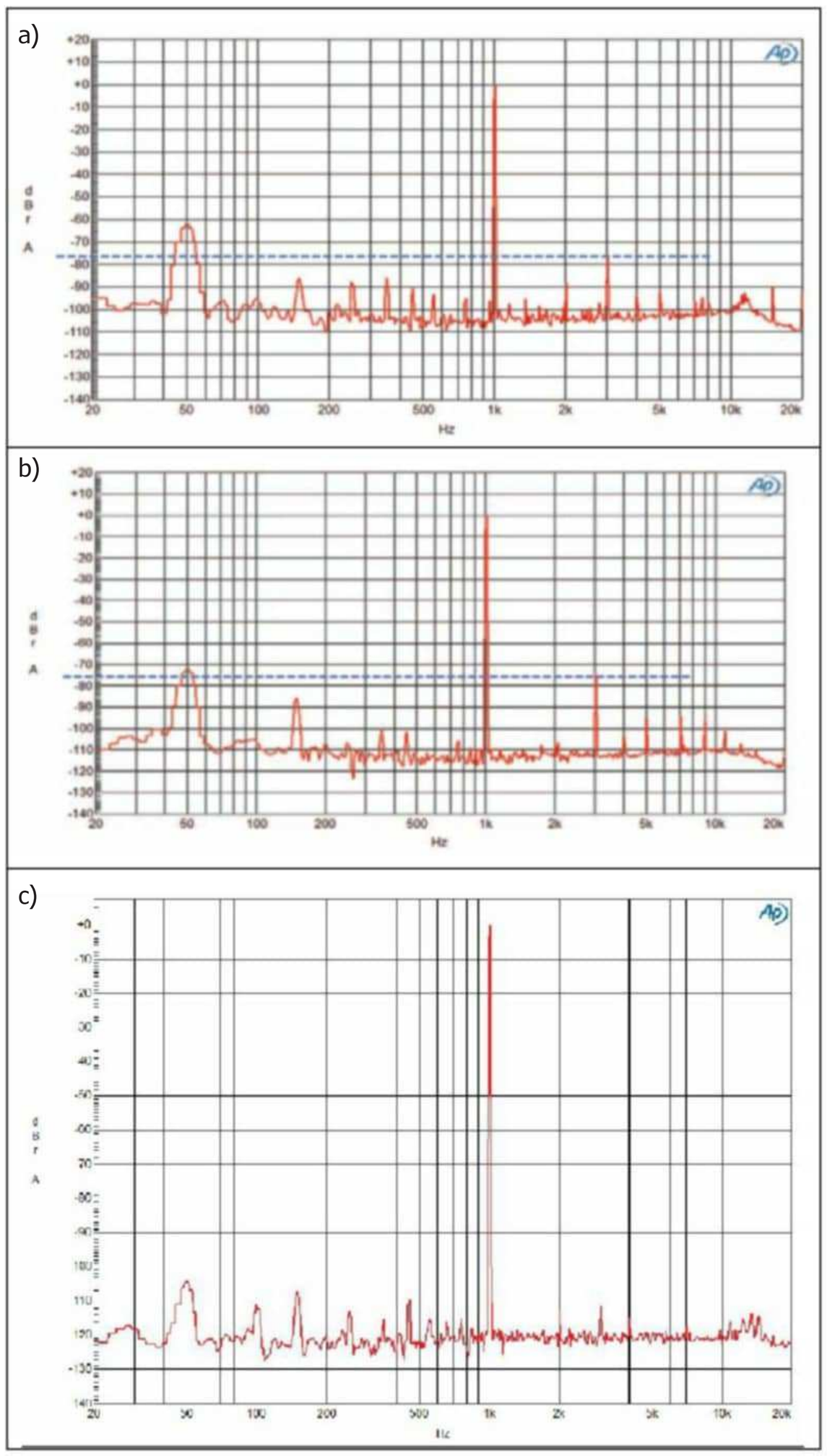


Figure 13: On-stator 1kHz distortion comparison, (a) paX, (b) Purifi, (c) Direct Drive, 1kHz, 500V RMS

The Bottom Line

This project, while not a simple or quick one, proves that it is perfectly possible to design a well-performing and reliable high voltage amplifier to directly drive electrostatic speakers of the QUAD ESL 63 family (**Photo 3**). The amp does not reach the drive levels of a traditional drive through a step-up transformer, but it produces very loud SPL levels in a living room setting. It just took me a long time as I had to design and work through unfamiliar and potentially lethal circuitry. Except for the high-voltage transformer, no special or custom parts were needed, everything is basically available from the usual distributors albeit sometimes with a backorder.

Listeners at several informal listening sessions unanimously stated that direct drive provides a more natural and detailed, high-resolution reproduction compared to traditional step-up drive. This difference is supported by **Figure 13**, which shows that, measured at the stators, the direct drive amp has substantially lower distortion compared to step-up drive. The Purifi and paX amplifiers both have less distortion than what is shown in the graph, and I contribute the difference to the step-up transformer.

Using high-voltage solid state output devices was not successful, although, at the time of this writing, 2.5kV MOSFETs for linear operation are becoming available. But the large interelectrode capacitances remain a challenge. Readers seriously considering a similar project can contact me for further information and details.

References

[1] P. Walker, et al., UK Patent 815.978, 1955

[2] P. J. Baxandall, "Electrostatic Loudspeakers," Chapter 3 in *Loudspeaker and Headphone Handbook*, John Borwick, Editor, Taylor & Francis, 2001

[3] The Spirito Effect, www.youtube.com/watch?v=oogXjjYKDyk

[4] Matrix Flowcode, www.matrixtsl.com/flowcode

Resources

D. Hermeyer, "An Electrostatic Speaker," *The Audio Amateur*, Issue 3/1972, p. 11.

D. Hermeyer, "Electrostatic direct drive amplifier," *The Audio Amateur*, Issue 4/1972, p. 9.

Steen Frank Jansen website, http://steenfrankjensen.com/projects/electrostatic-loudspeaker

Jazzman's DIY Electrostatic Loudspeaker Page, by Charlie Mimbs (Jazzman) on segmented ESLs, http://jazzman-esl-page.blogspot.com

R. Sanders, "Electrostatic speaker," *The Audio Amateur*, Issue 4/1975, p. 18.

R. Sanders, "Electrostatic direct drive amplifier," *The Audio Amateur*, Issue 1/76, p. 12.

F. Verwaal, "The Design of Electrostatic Loudspeakers," 2017, Rev. 2, www.scribd.com/document/63688628/The-Design-of-Electrostatic-Loudspeakers-Frank-Verwaal-2011.

Tubes, BJTs, JFETs, MOSFETs, or ICs?

By
Richard Honeycutt

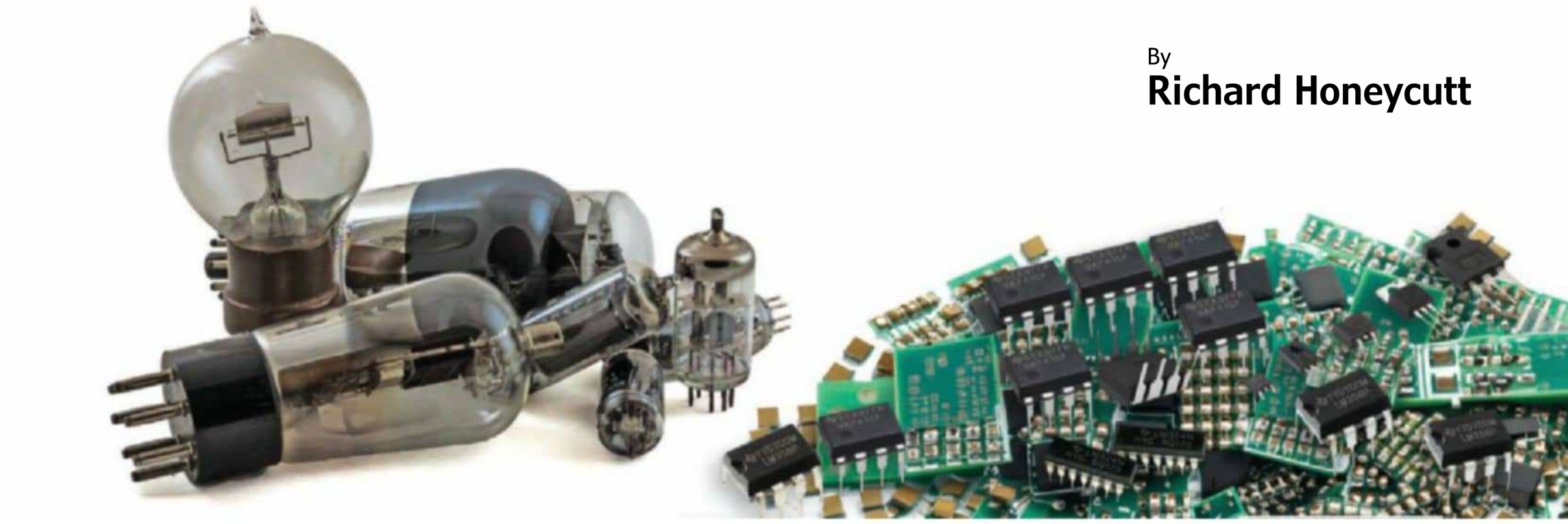


Photo 1: Most people have a simplistic concept of the “Ascent of Electronics.”

The history of semiconductor amplifiers, from the junction field-effect transistor (JFET) to the Bell Labs transistor, or Texas Instruments silicon-transistor, runs parallel to the ascension and twilight of vacuum tubes. Likewise, solid-state tube replacements remain a valid idea, gladly offered by some manufacturers.

The imminent demise of vacuum tubes was announced in the press soon after the invention of the transistor in the late 1940s. In actuality, we can rephrase Mark Twain and say rumors of the death of hollow-state technology have been greatly exaggerated. Let’s see what has actually happened (**Photo 1**).

In popular opinion, the history of electronics can be summarized in three steps. Vacuum tubes were invented about 100 years ago; they were large and had limited lifetimes. Transistors were invented about 76 years ago; they were smaller and longer-lived. Integrated circuits were invented about 60 years ago; they combined many transistors and other components into a single unit, and furthered miniaturization and reliability.

However, the reality is much more involved. Within a decade of the introduction of vacuum tubes, high-reliability ones were available for those who could afford them (i.e., the telephone companies and the military.

Transistors

The concept of the junction field-effect transistor (JFET) was patented by Julius Lilienfeld in the 1920s and 1930s, although 20+ years of development in material science and fabrication technology were required before German physicist Heinrich Welker patented the first working JFET in 1945. In 1947, William Shockley, Walter Brattain, and John Bardeen of Bell Telephone Laboratories (Bell Labs) were experimenting in an attempt to make the JFET manufacturable and practical when they invented the point-contact transistor (**Photo 2**). They announced their invention in June 1948—the same month that Welker announced his independent invention of the same device, which he called the “transistron.” Point-contact transistors being inherently noisy, Bell Labs experimented on ways to improve the transistor, resulting in Shockley patenting the bipolar junction transistor (BJT), as shown in **Photo 3**, on June 26, 1948. Although fabrication technology has advanced tremendously since 1948, for many years BJTs were the most common type of transistor.

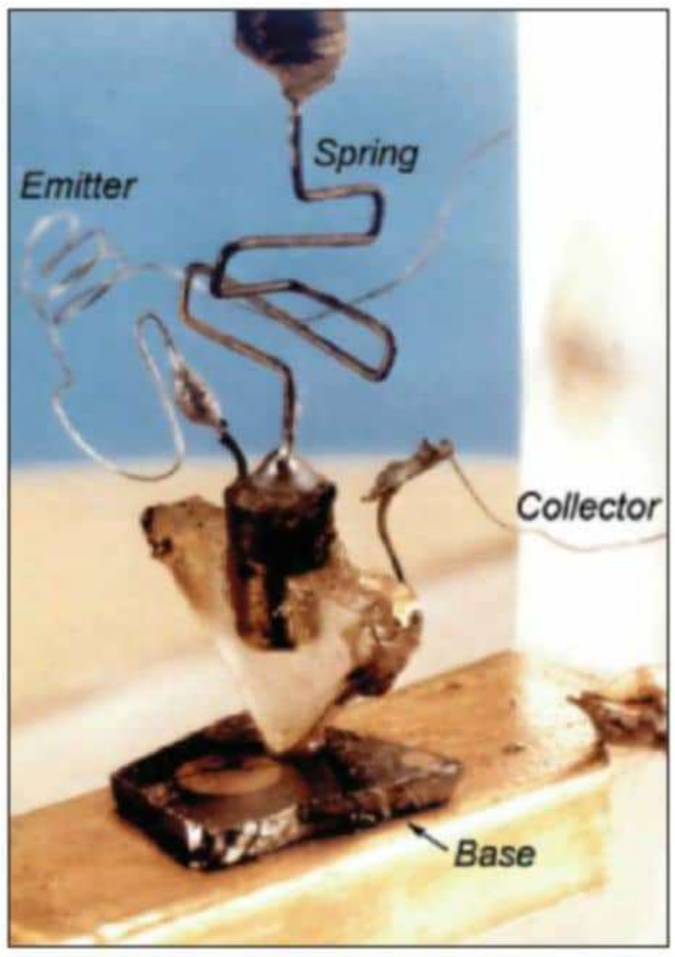


Photo 2: While smaller than the original vacuum tube, the first point-contact transistor was hardly miniature.

Now it gets complicated, because during World War II, a sort of analog computer called the M9 artillery director was used to aim big guns. Mathematical operations in the device were performed using

Hollow-State Electronics

Photo 3: The BJT was much quieter and more reliable than the point-contact transistor, and was capable of tremendous miniaturization.

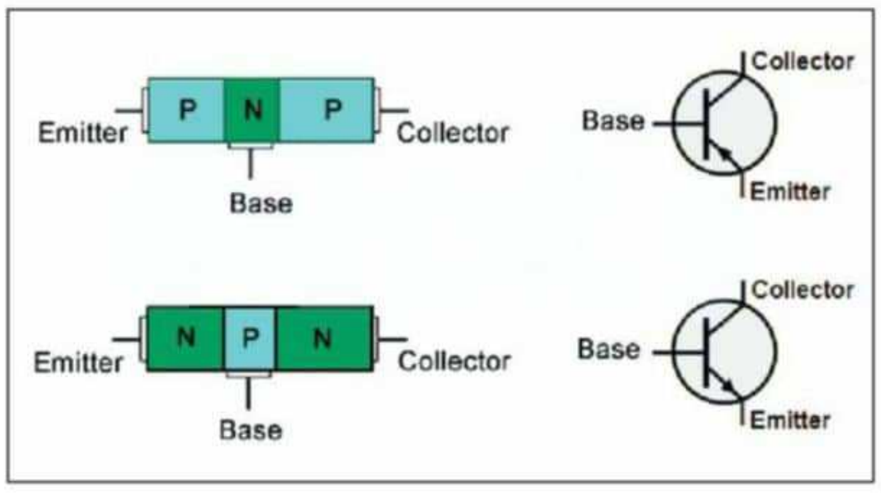


Photo 4: The Philbrick K2-W hollow-state op-amp comprised two 12AX7 tubes with supporting circuitry, connected to an octal plug.

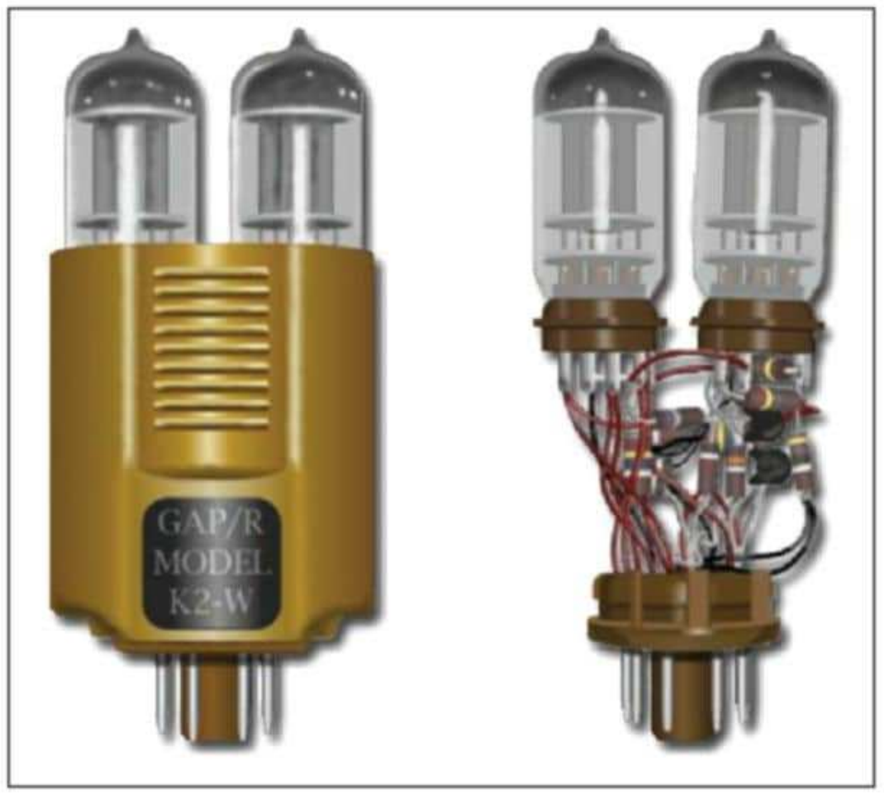


Photo 5: Although ICs have been made in many shapes and sizes, this TI model NE5534P in its Dual Inline Package (DIP) was one of the most popular high-performance audio op-amps for many years.

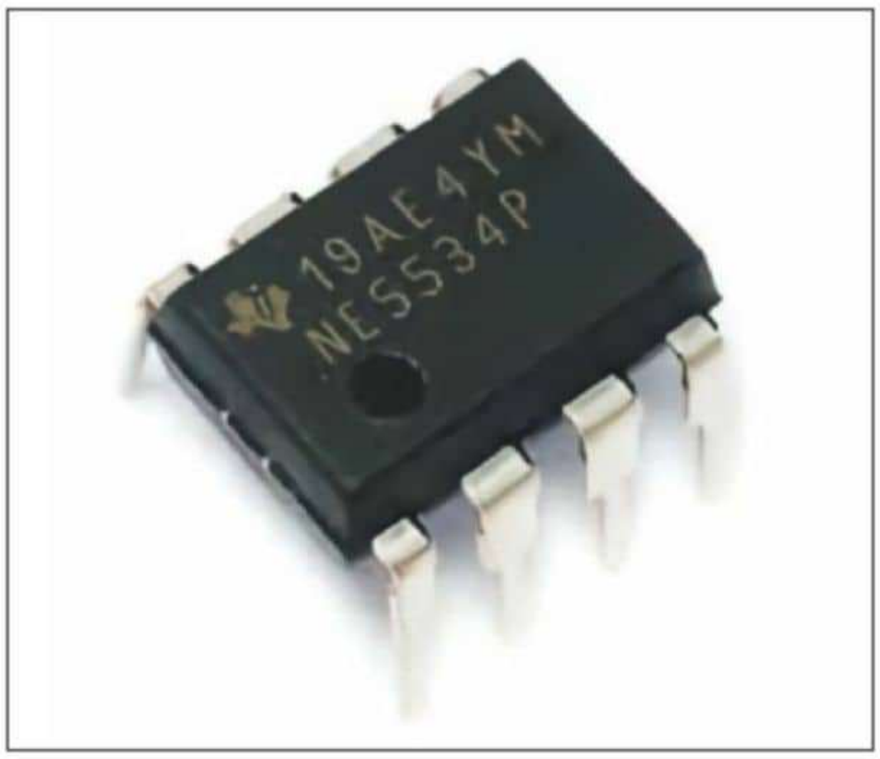
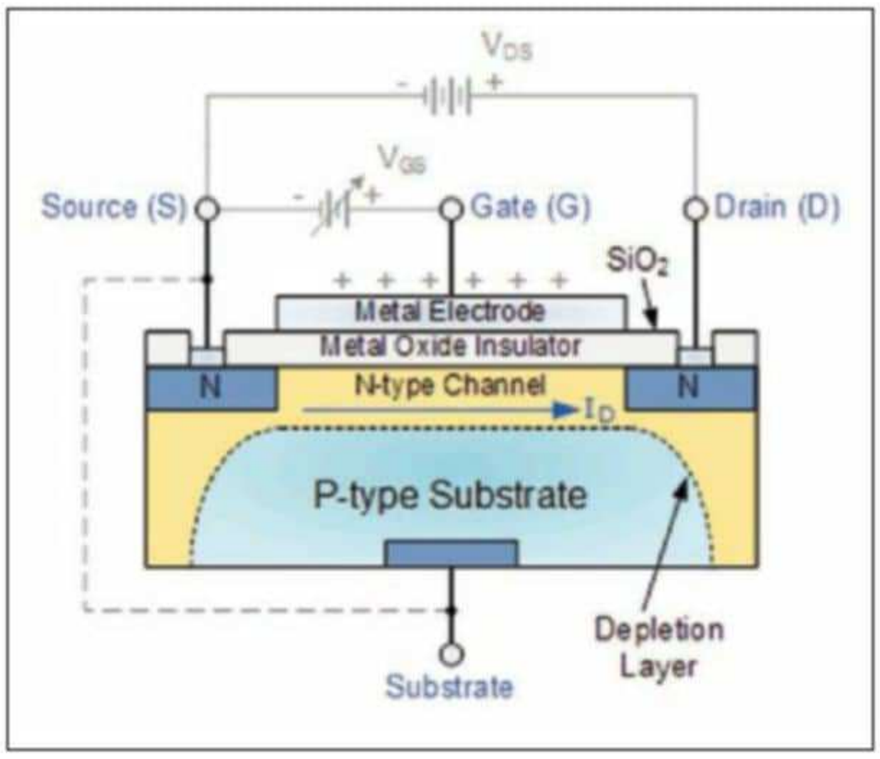


Photo 6: This drawing illustrates the internal construction of a MOSFET.



operational amplifiers (op-amps), first patented by Karl D. Swartzel, Jr. of Bell Labs in 1941. These were amplifiers having very high gains so that their electrical performance could be controlled by external feedback networks. By 1953, companies such as George A. Philbrick were manufacturing op-amps based upon vacuum tubes (**Photo 4**).

Jack Kilby, who had recently gone to work for Texas Instruments (TI) in 1958, conceived the idea of making both analog and digital building-block circuits on single blocks of semiconductor material. This concept led to the manufacture of solid-state op-amps and other types of integrated circuits (ICs). Kilby's ICs rapidly took over functions previously performed by discrete devices (**Photo 5**).

From the late 1940s into the 1960s, research continued on ways to improve field-effect transistors (FETs), leading to the development of metal oxide semiconductor field-effect transistors (MOSFETs) as shown in **Photo 6**. These devices had an almost infinite input resistance, and thus could be made to behave more like vacuum tubes, an important point since by the late 1960s, hollow-state amplifiers—recently proclaimed dead—were just beginning to develop a commercially significant following, especially for musical instrument amps and high-end home music amps.

Prompted by the large number of installed vacuum tubes in telephone amplifiers ("repeaters"), in 1967 Teledyne began marketing the Fetron, JFET-based plug-in replacement for vacuum tubes (**Photo 7**). The object of the Fetron was to provide solid-state reliability for existing equipment without requiring rewiring. Available for replacement of a variety of tube models, Fetrons were primarily sold to replace tubes in telephone repeaters, where their higher cost versus tubes was offset by the reduced cost of maintenance labor and lower air conditioning costs. A Fetron generally comprised a cascode pair of JFETs, resistor-capacitor networks, and a fuse. The metal tube in which the Fetron was mounted, and the base/pinout, was chosen to match those of the device that it replaced.

Now let's look at the performance of tubes, BJTs, JFETs, and MOSFETs.

Behavior of Tubes, BJTs, JFETs, and MOSFETs

In audio amplification, the principal types of tubes used are triodes and pentodes. These types share two characteristics:

- They are normally-on devices—a negative control-grid bias is required to reduce

the plate current to a “quiescent” value, which produces a voltage drop across the internal plate resistance r_p, producing an instantaneous plate voltage equivalent to the zero-crossing value of the output AC signal.
- They act as voltage-controlled current sources.

The gain of triodes and pentodes can be stated as a transconductance, symbolized as g_m, which stands for “mutual conductance,” the old name for transconductance. Conductance G is the inverse of resistance: R=V/I, so G=I/V. In this case, V is the grid-to-cathode voltage, and I is the plate current. It is calculated as $g_m=\Delta i_p/\Delta v_{g\text{-}k}$, or the change in instantaneous plate current divided by the change in grid-to-cathode voltage that caused it.

The gain of a triode is often stated as a voltage gain μ, which is calculated as $\mu=\Delta v_p/\Delta v_{g\text{-}k}$. It can be derived from the transconductance and plate resistance: $\mu=g_m r_p$. Because of the structure of a triode or pentode, both types have a very high input resistance when used in a common-cathode circuit.

As you can see from **Figure 1** and **Figure 2**, plate current in a triode is affected by the plate voltage; whereas in a pentode, not so much. Triodes



Photo 7: The Teledyne Fetron was a generic name for a JFET-based vacuum tube replacement.

About the Author



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

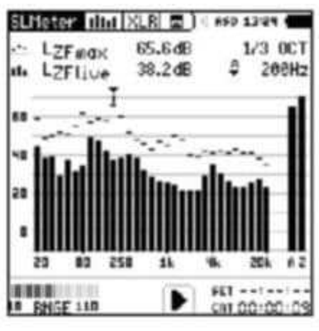
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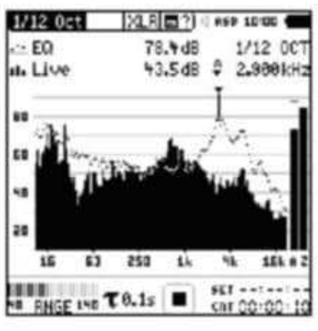
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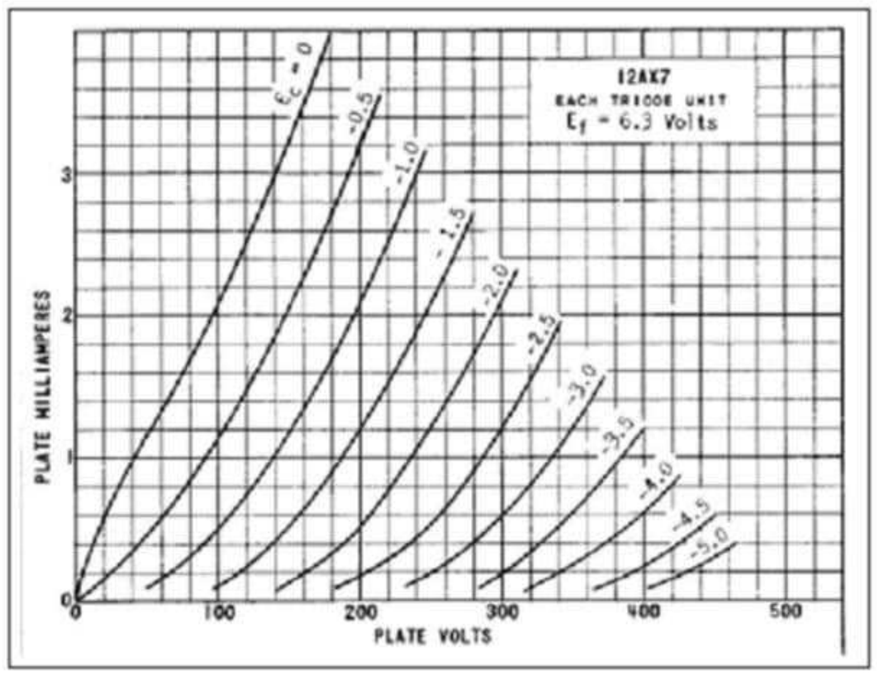


Figure 1: The plate characteristic of a triode shows the dependence of plate current on both plate voltage and grid-to-cathode voltage.

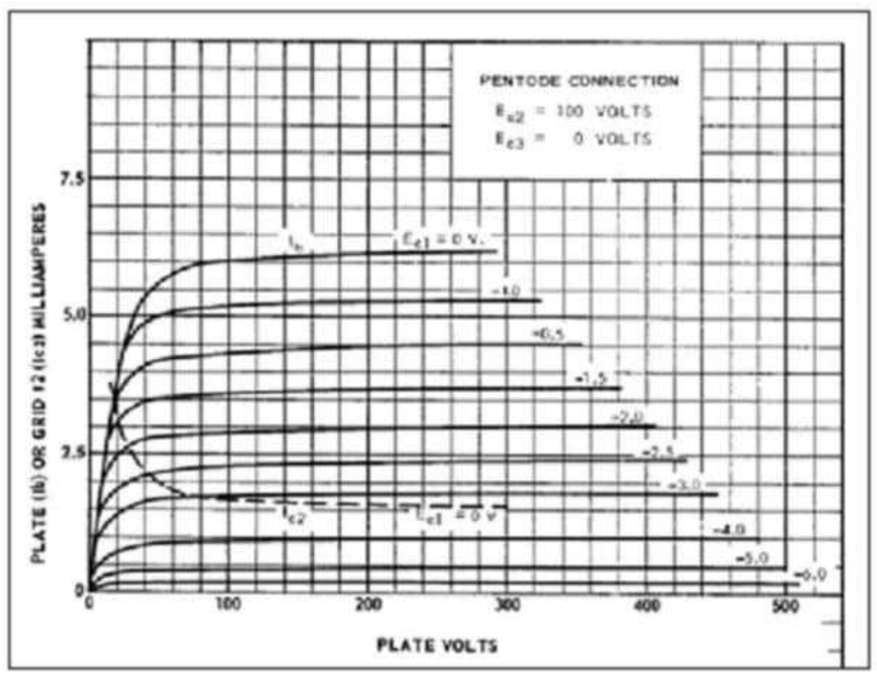


Figure 2: The plate characteristic of a pentode shows much less dependence of plate current upon plate voltage, although it does depend upon control-grid-to-cathode voltage. (Screen-grid voltage was held constant at 100V for this measurement, and the suppressor grid was grounded.)

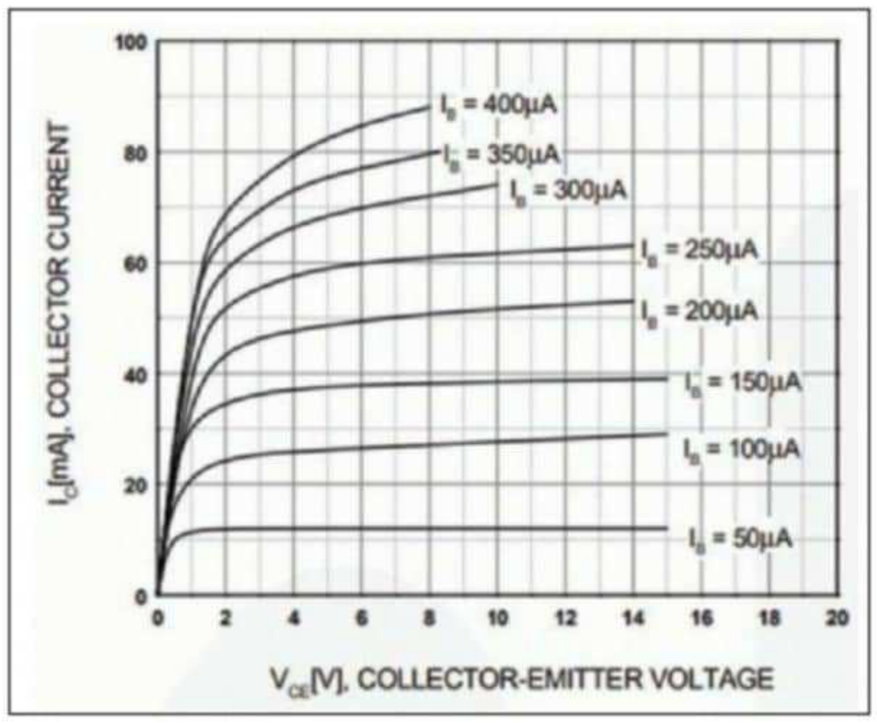


Figure 3: The collector characteristic curve of an NPN BJT resembles the plate characteristic curve of a pentode.

Resources

Freya+ Balanced Tube Preamp with Remote Control, Schiit Audio, www.schiit.com/products/freya-n

D. Laws, "Who Invented the Transistor?" Computer History Museum, December 2013, https://computerhistory.org/blog/who-invented-the-transistor

Saga+ Passive or Active Remote Tube Hybrid Preamp, Schiit Audio, www.schiit.com/products/saga-2

Valhalla Triode OTL Headphone Amp and Preamp, Schiit Audio, www.schiit.com/products/valhalla-1

Vali Tube Hybrid Headphone Amp and Preamp, Schiit Audio, www.schiit.com/products/vali-3

tend to produce more even-order than odd-order distortion; pentodes, more odd-order.

A BJT is a normally off device—a base bias voltage (positive for an NPN transistor, negative for PNP) must be applied to turn the base-emitter diode on and cause collector current to flow. Thus, it acts as a current-controlled current source, having a much lower input resistance than a tube does. **Figure 3** shows that the collector characteristic more closely resembles a pentode's plate characteristic than a triode's. Like pentodes, BJT's produce more odd-order distortion than even-order.

Op-amps are purposely designed with an extremely high voltage gain, so that their performance in a practical amplifier circuit is controlled by a large amount of negative feedback. This gives them the ability to have very low distortion as long as they are not overdriven, but once overdriven, they produce high amounts of predominantly odd-order distortion extending to far higher harmonics than that of tube amplifiers.

A JFET is normally on, and its gate is normally negatively biased like a triode's control grid. Available in N-channel and P-channel versions (analogous to NPN and PNP BJTs), the JFET is a voltage-controlled current source whose gain is specified as a transconductance. **Figure 4** shows the drain characteristic of a common JFET. It has a high input resistance, but not as high as a vacuum tube.

MOSFETs are available in both N-channel and P-channel versions and are also voltage-controlled current sources having input resistances essentially as high as those of tubes. Normal biasing depends upon the mode of the MOSFET: As you can see from **Figure 5**, enhancement-mode devices are normally off, while depletion-mode ones are normally on. Beyond the initial curve, a MOSFET's drain current is a very linear function of the gate-to-source voltage. This can be seen from the drain characteristic of a depletion-mode MOSFET shown in **Figure 6**.

The linearity of an active electronic device is related to the evenness of spacing of the output characteristic curves as the control bias (grid/base/gate) is changed in 1V increments. As you can see, a MOSFET is the most linear of the devices we have discussed. Some listeners prefer to hear a slightly distorted output signal—most often with primarily low even-order distortion. This preference has been suggested as a reason that some prefer Class-A triode power amplifiers over other designs. As discussed in Hollow-State Electronics articles published in September, October, and November 2012, the most rigorous listening tests so far performed have failed to demonstrate humans'

ability to distinguish among well-designed amplifiers of different designs, as long as the amps were not driven into significant distortion. A given listener's preference for either hollow-state of solid-state technology is therefore likely caused by the difference in harmonic distortion signatures among tubes, BJTs, JFETs, and MOSFETs when amplifiers are overdriven, even during brief transients.

LISST And Other Schiit

Schiit Audio, a company founded in 2010 by Jason Stoddard and Mike Moffat, designs and manufactures an interesting line of audio equipment—all built in Texas and California—including solid-state, hybrid, and tube preamps and headphone amps.

Rather than adopting a "hollow-state purist" approach to audio design, Schiit chose to focus on sound quality, making the choice of technology a secondary consideration. In addition, Schiit Audio manufactures a solid-state, plug-in tube replacement, the Linear, Integrated Solid-State Tube (LISST), a MOSFET-based plug-in replacement for a 6SN7 octal twin vacuum triode (**Photo 8**). Designed specifically for their audio products, the LISST is not intended as a general-purpose tube replacement, as was the Fetron.

At one point in time, the Schiit website was quoted as saying, "What's inside are two depletion-mode MOSFETs, together with minimal support circuitry to protect them against static, and to set the gain necessary for our products. There's no "triodization" or other tricks applied to make them act like tubes or have curves like tubes. They're just good, linear, solid-state devices…in a can."

Schiit does not market the LISST devices as multipurpose tube replacements: they are optimized for use in Schiit products. Customers are welcome to buy and try them in other devices, but since they are not designed or optimized for such use, there are no guarantees for such applications. Schiit also makes no claim that LISST devices sound like tubes: again, users are invited to investigate for themselves."

However, Schiit does manufacture four products incorporating hollow-state technology: two preamps and two headphone amps. Although they also manufacture other audio products, staying within the focus of the Hollow-State Electronics column, we'll limit our discussion to hybrid and tube models.

Preamps

The Saga+ preamp is a combination remote-controlled source selector and relay-switched volume control. It can be operated in passive or active modes. In the active mode, it uses a 6N8S dual triode and a pair of BJTs and can drive long cable runs. The tube filaments run on DC.

The Freya+ is similar in function to the Saga+. However, in addition to the passive mode, it

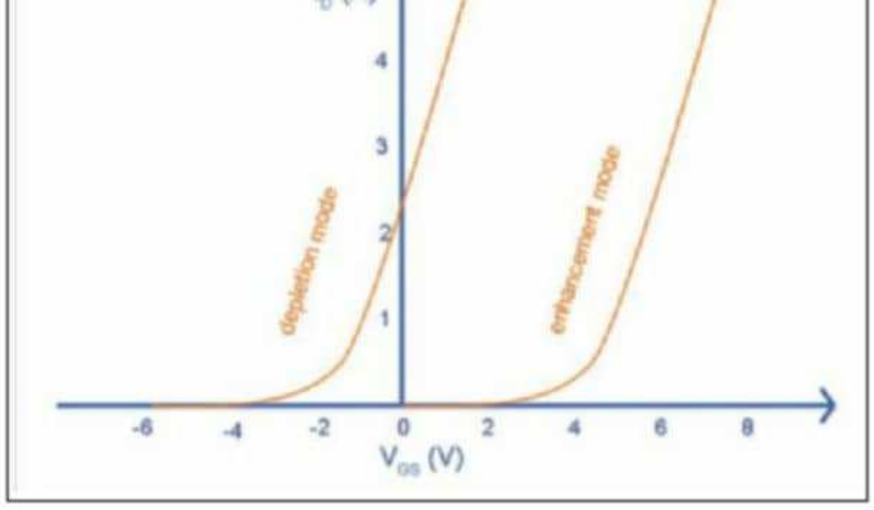


Figure 4: The drain characteristic of a JFET can be divided into three regions: to the left of the dashed blue vertical line is the "ohmic' region; the region used for amplification in in the center; and the breakdown region is usually avoided in amplifiers.

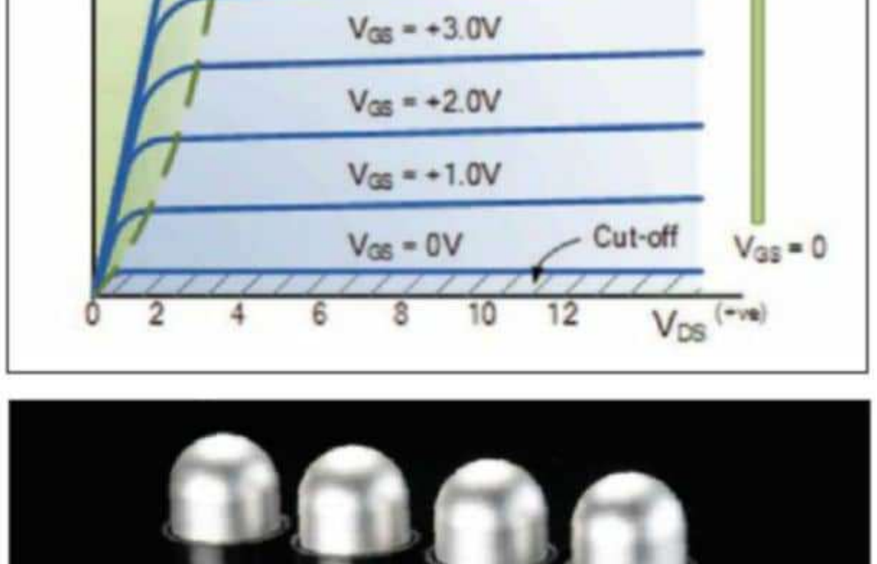


Figure 5: The relationship between drain current and gate-to-source voltage is similar for a depletion-mode and an enhancement-mode device, but the gate-to-source cutoff voltages are different.

Figure 6: The equal spacing between drain-current lines as gate-to-source voltage is varied in 1V increments and indicates linear amplification.



Photo 8: The Schiit LISST replaces an octal-based dual triode.

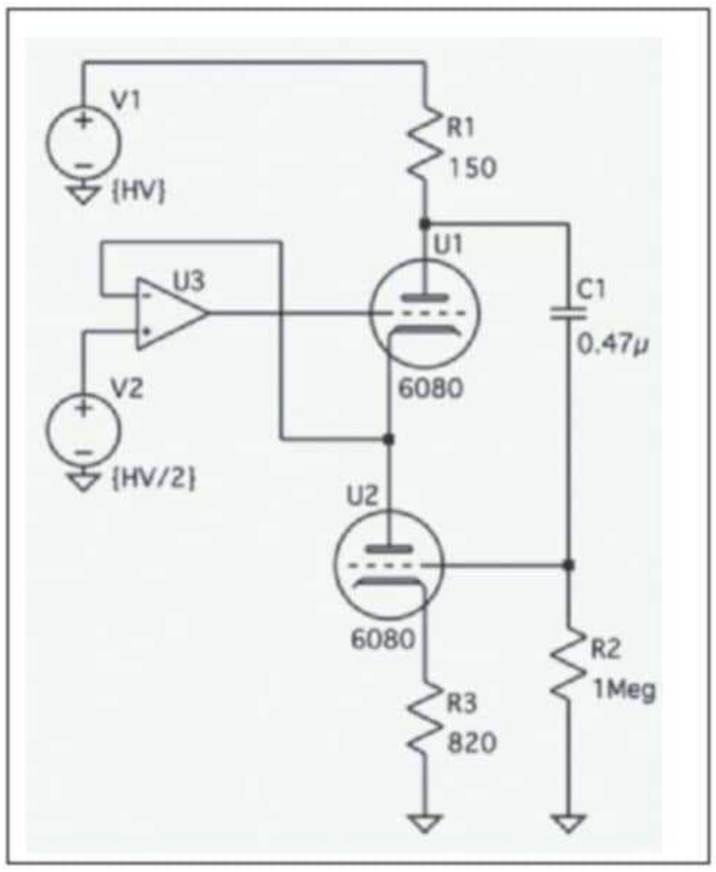


Figure 7: The White cathode follower consists of two triode cathode followers stacked.

offers a choice of unity-gain JFET compound differential buffer and 12dB gain tube gain topologies. In the tube mode, the entire signal path is hollow state. LISST devices can be plugged in instead of the 68NS or 6SN7 tubes, giving a 14dB gain. The tube gain topology employs a differential triode input and a "semi-circlotron" output stage. It is based upon Broskie's single-power-supply circuit described in the sidebar titled "Tube Gain Technology," but the tubes are biased in Class A.

Headphone Amps

The Vali headphone amplifier is a discrete current-mode non-inverting hybrid unit using a single 6N3P dual triode. Other types of tubes can be used as well: the 5670, 2C51, ECC42, and even the WE396A. Maximum output power at 10% THD ranges from 200mW per channel continuous power into a 600Ω load to 1.5 WPC into 32Ω.

The Valhalla is an output transformerless headphone amplifier designed to drive either 300/600Ω or sensitive lower-impedance phones. This flexibility is made possible by switchable gain: 16.9dB (High) or 3.5dB (Low). Each channel uses a 6N1P dual triode connected as an input differential amplifier, and a 6N6P dual triode connected as a White cathode follower (**Figure 7**).

In all the products listed, Schiit Audio defines LISST options as having "a solid-state amplifier, with zero concerns about tube life and tube variability," that "sound very good." In addition to the hollow-state and hybrid products we've discussed in this article, Schiit manufactures a wide variety of solid-state analog and digital products, as you can see by visiting www.Schiit.com. Full performance specs for all Schiit products are also available on the website.

Tube Gain Technology

The Circlotron output stage was discussed in the October 2017 Hollow-State Electronics article. Its schematic is shown in **Figure A**. Cecil T. Hall filed a patent for this circuit in 1951. Tapio Köykka filed a patent (Finnish Patent 27332) for a similar circuit in 1952. He later filed a second patent (Finnish Patent 29642) based on improvements he made to it in 1955. Mr. Sitoshi Shimada of Japan invented a similar circuit in 1952, which he called the Cross Shunt Push Pull (CSPP) topology.

A prominent commercial Circlotron amplifier was the Electro-Voice A100, which boasted 100W rated output power (200W on peaks), a frequency response flat +0.5dB from 20Hz to 50,000Hz, THD was less than 0.5%, and IMD was less than 1.2% at rated output.

Advantages of the Circlotron include the fact that there is no output tube DC in the transformer, a direct result of the balanced bridge circuit. Also hum and buzz are canceled, and all switching transients are eliminated through unity coupling of output tubes. Other advantages are that the primary impedance of the output transformer is one-quarter of that found in conventional amplifier output circuits, affording a vast increase in power output at extremes of the frequency spectrum.

All these circuits used dual floating power supplies, which are awkward to implement. John Broskie devised a Circlotron stage using a single power supply (**Figure B**).

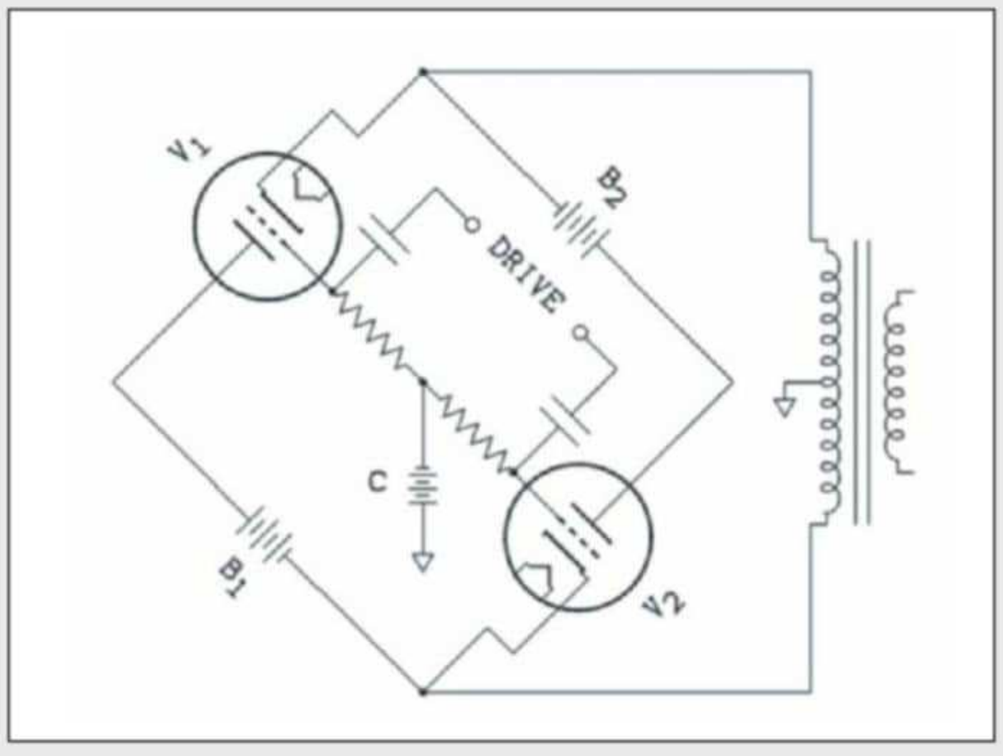


Figure A: This variant of a push-pull stage was introduced in 1951. (Schematic courtesy of Max Robinson)

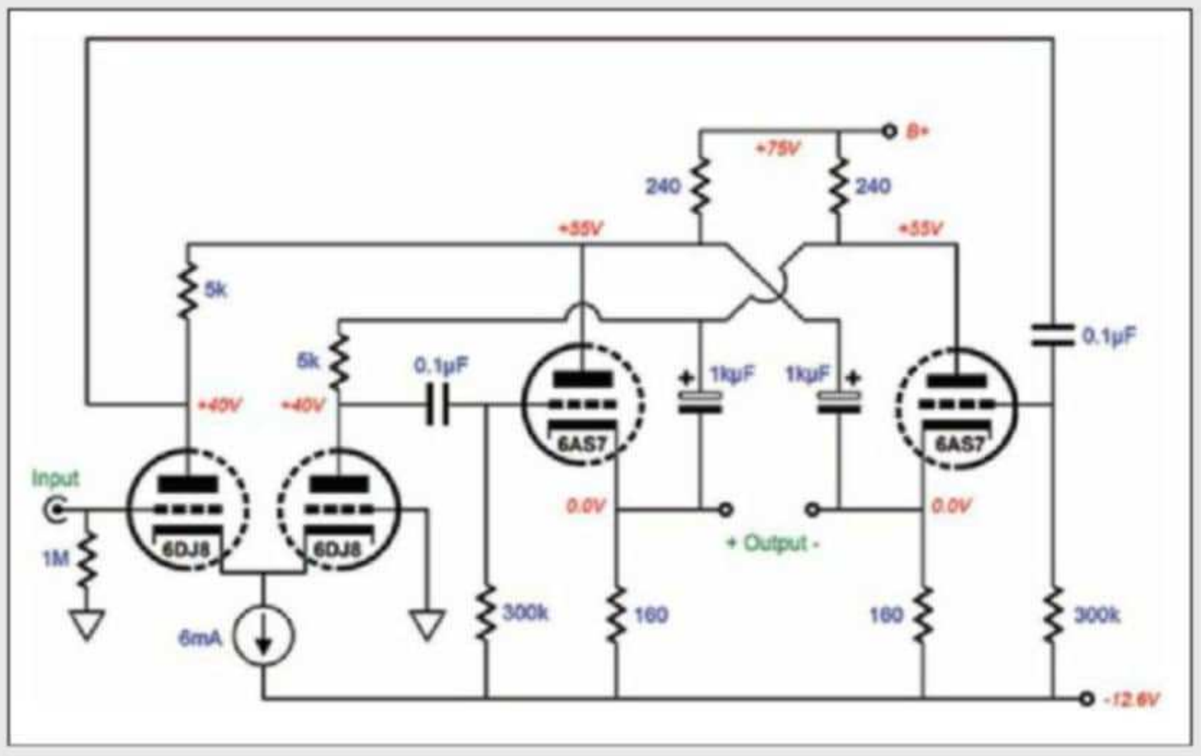


Figure B: John Broskie developed this variant of the circlotron using only a single power supply. (Circlotron and ES Headphone Amplifiers, tubecad.com)

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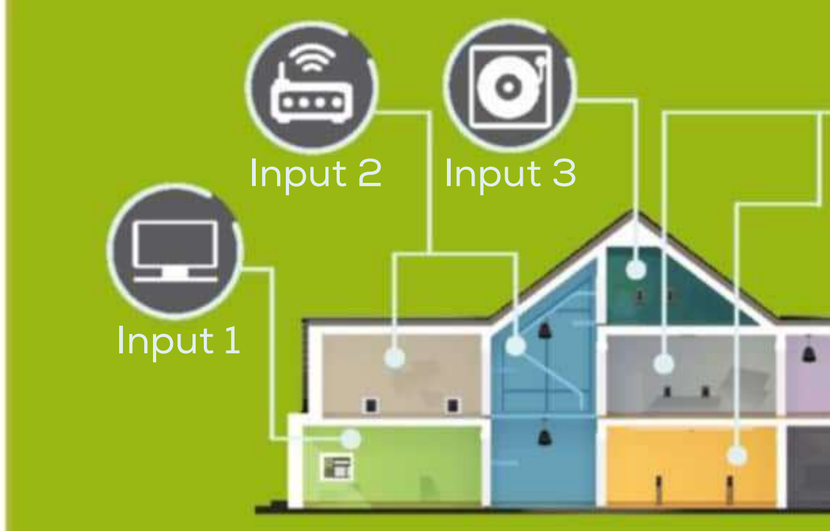
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