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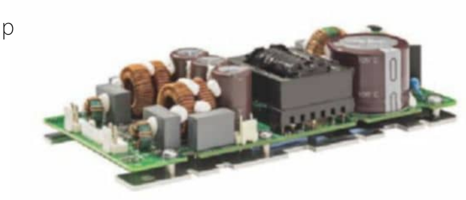

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October 2020 ISSN 1548-0628

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

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

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

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

Advertising:
**Strategic Media Marketing, Inc.**
**2 Main Street**
**Gloucester, MA 01930, US**
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## Spending Time Online

Now more than ever, our activities as magazine publishers only make sense when combined with daily news and article updates on our website—audioxpress.com—plus our email newsletter *The Audio Voice*, which continues to gain a growing audience every week. Given the difficult circumstances of a global pandemic that have caused our digital activities to grow exponentially, our website is visited by more than 10,000 unique visitors every day, while our newsletter now reaches nearly 20,000 mailboxes and probably twice as many readers who follow us through our social media channels. In fact, in many regions of the globe, that's how many readers follow and know about *audioXpress* (and its sister publication, *Voice Coil*).

So here's my appeal for those of you who are not necessarily consumers of online/digital publications much less social media (and I am not going to be the one to endorse one over another…), please visit audioXpress.com, access our massive article archive, and sign up there to receive our newsletter *The Audio Voice*.

A word of warning, *The Audio Voice* content topics reflect all the application areas addressed in the magazine, and change every week, and there are frequent guest editorials by top experts in the audio industry. So, if you see a particular article one week discussing voice recognition or tube amplifiers and you don't feel that's relevant for you, bear with us, because you don't want to miss the next one!

This also serves to highlight that, when we send a new edition of our magazine to the printers every month, we basically wrote another (larger) one every week on a whole different range of topics. This current October 2020 issue, was just a small part of all our work throughout the month of July and August, normally the quietest time of the year for the audio industry.

And because we cover multiple industries and we focus on innovation and industry news, we have managed to keep or even increase the pace of news stories, even during the slowest season of the year for product announcements.

Of course, things are different this year. As a quick survey of industry contacts confirms, R&D activity has accelerated, and marketing is refocusing on a steady promotional calendar. Many new products are being announced by manufacturers at a steady pace— even if they are not, in some cases, necessarily entering production, which has been suffering delays in different regions of the globe.

The month of August saw no shortage of new product reveals. In fact, it was extremely refreshing to learn about Sony unveiling its newest WH-1000XM4 next-generation adaptive noise canceling headphones, almost a month before the IFA show in Berlin, where traditionally that unveiling would take place. And from a different product segment, in home audio, it was just as inspiring to receive a succession of product announcements from NAD Electronics, directly appealing to all the people staying at home, who might want to upgrade their entertainment solutions to high-quality integrated systems that offer all the latest cutting-edge technologies—from hi-res wireless audio to the whole home to AirPlay 2, and even Dirac Live Room Correction.

All things you can read about first-hand on our website.

In the residential installation space, the topic featured in this month's Market Update, things are also evolving. Even as the CEDIA 2020 show was forced to go virtual this year, there's a lot to discuss, as the entire industry is quickly adapting to serve an expanded range of needs for homeowners, including their new workspaces, where connectivity and acoustics are critical. And audio technology companies in the CEDIA space, which typically sell whole home audio and entertainment products, such as home theater, are no longer focusing on the most "installer-friendly" solutions, but on reinforcing the emotional connection with the end user. Last year, the challenges for the home technology integration association were already pointing in this direction. A global virus in 2020 made those changes an imperative.

That's why, before you dive into this month's Market Update article, I think you will find it useful to read my CEDIA 2019 report available online here:
https://audioxpress.com/article/cedia-2019-show-report-residential-integration-in-the-age-of-the-connected-home

J. Martins
Editor-in-Chief

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## COLUMNISTS

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

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

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

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

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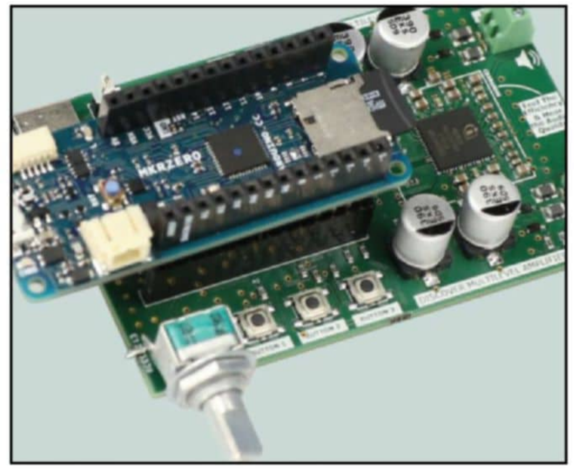

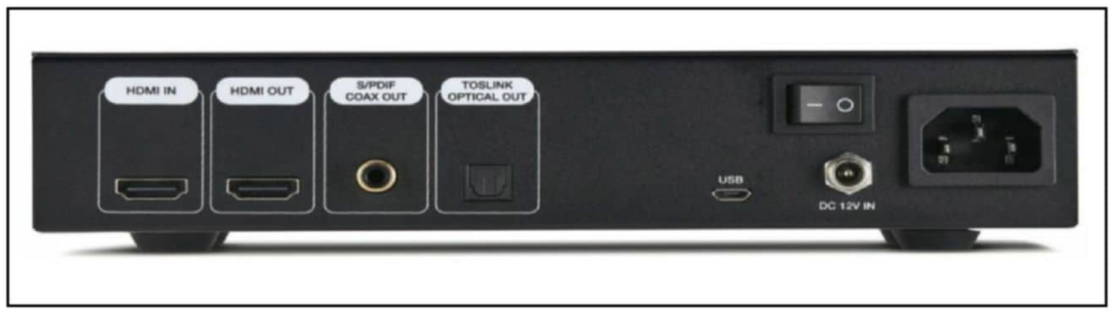




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# Evolving Residential Installation

## Home Entertainment and Workspaces

By
**J. Martins**
(Editor-in-Chief)

Entertainment installation at home. (Image courtesy of Mikodam Furniture, www.mikodam.com)

In 2019, in the first Installation Market Update feature published by *audioXpress*, we essentially focused on commercial applications and highlighting key technical trends for that specific segment. This year, we will concentrate on residential installation and new products for entertainment and personal workspaces, as companies are refocusing on this segment.

Given the global pandemic context in which we are living, most of our activities are highly restricted to our own homes. Six months have passed since the beginning of confinements, and with the perspective of working and sheltering at home extending to 2021, homeowners are realizing they could probably improve a bit on their experience by investing in further residential systems. Luxury items for some, essential improvements for others, budget projects for many, for work or for leisure, the residential installation market has suddenly expanded in all directions.

Many corporations, large organizations, and even governments are realizing that working from home—or anywhere—is actually a viable concept now that people are always connected. The pandemic has accelerated the adoption of new remote collaboration and communication tools, and created a foundation for a flexible, distributed workforce, where the need for traditional office space is not as great.

Eventually, the solutions that people are adopting at home could also be leveraged to create new experiences elsewhere. To new styles of working spaces, where it will be productive to use unique tools and resources, while taking into account safety rules, including social distancing. Again, many companies that were until recently focused on the corporate/office installation market are adapting ideas from the new generation of residential projects and are responding to the increasing demand for installations in private homes and converted co-working and temporary working spaces. Likewise, manufacturers in all product segments have quickly adapted solutions to those unique needs.

### The Integration Perspective

CEDIA Expo 2019 in Denver, CO, took place four months before the United Nations' World Health Organization (WHO) declared COVID-19 a global pandemic, and yet, many of the trends that *audioXpress* reported from that important show are extremely relevant now. CEDIA Expo 2019 included a lot of information about home automation or what people like to call these days, smart home solutions. And the Custom Electronics Design and Installation Association (CEDIA) was quick to promote the concept within its robust residential integration network of vendors, distributors, and service providers, which in itself is one of strongest market channels, working in sync to provide the best solutions for the home.

This proved to be timely for the CEDIA network. The residential integration market is structured upon an extensive network of regional and local retailers, supported by hyperlocal installation providers and service professionals. Not many industries have such an extensive and highly specialized structure—sharing many aspects of the intricate and highly specialized electric power and HVAC installation network that thrives upon the home building and renovation opportunities.

And as the pandemic has proven, this strong CEDIA connection is vital to quickly respond to changing priorities in residential installation, embracing the latest trends in home automation and smart home systems. The CEDIA network in general adjusted to the wider scope of opportunities allowed by the connected home, while certain companies and professionals decided to specialize in specific product segments—home theater being one of those—while broadening their business opportunities in distribution and installation.

The residential installation industry was already feeling the effects of technology changes, amplified by the disruption caused by large technology players such as Amazon, Google, Apple, and others. After all, wireless, voice-controlled systems are relatively easy devices to buy off the shelf or online and put to use. Things are not that simple when it comes to expanding more extensive "smart home" features, such as controlling home temperature, closing and raising the shades, and security features—all from the same device. That's where professional help is usually requested. And even companies specializing in home theater and entertainment solutions, which have already evolved to whole home audio systems controlled with Amazon and Google voice assistants, couldn't say no to those requests from new and existing clients. The pandemic disruption is now accelerating those changes, creating a rising tide that is raising all product categories.

Also, given the pandemic restrictions, integrated home systems sometimes have to be quickly built and moved into place in a short time, making Wi-Fi infrastructure a preferred solution—simply taking over from the wired service router/hub that connects the home. Also, quickly emerging new installation areas that are now considered essential by homeowners include power and charging solutions, enabling users to always have devices ready to be moved around (when people get used to wireless, it's hard to go back...). These charging solutions increasingly have to provide both USB-C fast charging and Qi Wireless options. And if installation infrastructure usually means hiding cabling and installing easy-to-access, neat in-wall power solutions, the introduction of new home workspace requirements also demands new cable organizers and lighting systems to be quickly deployed. Increasingly, those requested workspace items need to be of the type that can be moved around. And of course, we could go on simply to address additional acoustical treatment requirements (a topic for a dedicated article).

Residential integration is increasingly about smart home control. (Image courtesy of Control4/SnapAV)

The market represented by CEDIA entails all the systems that fit inside the home, which require a different set of skills and marketing strategies when dealing with homeowners and families, not complex organizations, such as in the commercial sector. Residential integration is now embracing a wider range of technologies, from lighting and comfort automation (companies such as Lutron and Legrand seem to fully understand that...), and expanding with new disciplines of home security, connected locks, and home communications. And now, in 2020, all installation companies—both residential and commercial oriented adapting to increasing demand—have gained a strong new application segment: home workspaces.

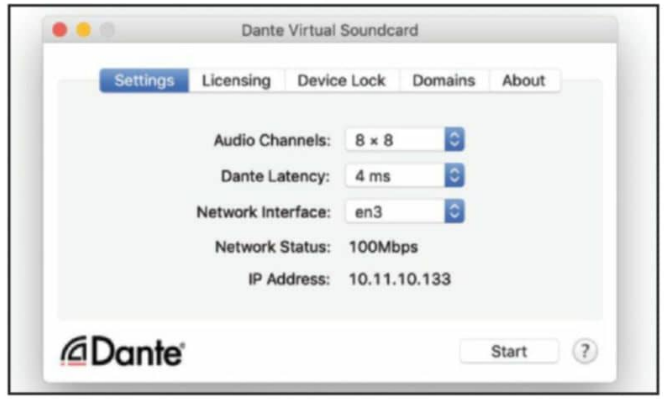
On the technology side, many other interesting trends were already visible at CEDIA 2019, such as the rapid adoption of networking infrastructures to connect systems together—from audio and video distribution to control and automation. But probably before integrators and manufacturers have the chance to decide whether to embrace Dante or any other protocol or to use 10 Gigabit Ethernet, wireless technologies will probably take over that space much faster. With the residential sector quickly embracing Wi-Fi for almost everything, the emergence of 5G networks and broader bandwidth solutions such as Wi-Fi 6 will quickly surpass the limitation of wired networks based largely on Gigabit Ethernet. Given the current trends in home offices and the increasing needs for connectivity at home, we can safely predict that is only going to accelerate.

Wi-Fi 6 (802.11ax) deployments will quickly gain worldwide adoption because the technology directly answers the demand for bandwidth-intensive, low-latency applications that now exist inside each household. As gigabit-speed broadband services are reaching the homes, video streaming, gaming, and smart home applications will expand, and propel the adoption of faster whole-home Wi-Fi coverage. And Wi-Fi 6 can provide a peak data rate up to 10 Gbps, allowing the creation of powerful mesh systems to provide an excellent whole-home Wi-Fi experience. The products are now reaching the market, and 2021 could be a very strong year for Wi-Fi 6 installations.

For integrators, this just serves to show the need to constantly adapt with technology and market demand. In a market segment that typically specialized in planning and passing complex wiring, residential integration now will require a new set of skills on setting up wireless systems.

## Audinate

# Dante Audio Networking Appeals to the Residential Market




Audinate's Dante—the audio-over-IP and now also AV-over-IP solution that already dominates the world of professional audio—is seeing rapid adoption by manufacturers of residential audio equipment. Dante is showing up in new products from leading brands in the residential space, including Harman, Bose, Yamaha, AudioControl, ELAN, Atlona, Control4, Nortek, and Powersoft.

With Dante now evolving to an AV-over-IP solution, the residential industry is seriously looking at the technology to replace many proprietary solutions, and effectively taking over from HDBaseT, by introducing all the advantages of AV transmission over Internet Protocol (IP) already allowed by SDVoE, and in large part embraced in Dante. But the reason to believe that Dante could be even more successful in home environments has to do with the strategic Dante Virtual Soundcard software tool that is widely available for any Mac or PC and even Virtual Machines.

Dante Virtual Soundcard turns any computer into a Dante endpoint for networked audio, providing a standard WDM or ASIO audio interface that enables any installed audio software to send and receive up to 64 channels of lossless audio over a standard 1 Gbps network to any Dante-enabled AV endpoints, including other instances of Dante Virtual Soundcard. Dante Virtual Soundcard is an ideal solution for computer-driven media playback, multi-channel recording and lecture capture, and is 100% compatible with the many thousands of Dante-enabled products available from multiple vendors. And networking is certainly a key differentiator in connected home solutions, opening new opportunities to integrators.

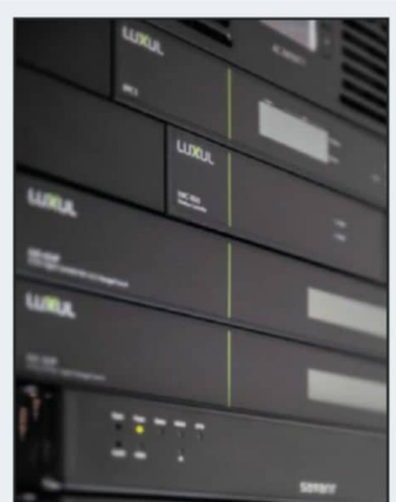
**www.audinate.com**

The latest JBL Synthesis multichannel amplifiers, AV receivers, and surround processors, like the SDP-55 and SDR-35 all include Dante connectivity as a key component of their designs, providing simplified component connectivity and advanced audio networking.

Powersoft's Mezzo amplifier range includes models with Dante for the residential market.

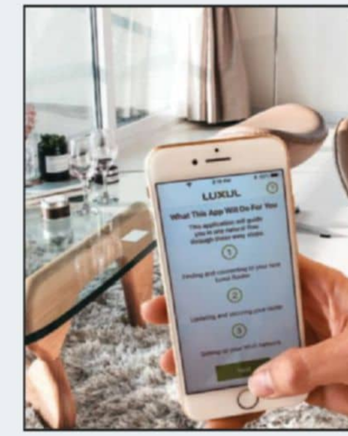
## Luxul

# High-Performance Networking Solutions for the Smart Home



With residential integration increasingly relying on wired and wireless networking solutions, Luxul—the IP networking solutions brand, part of Legrand's portfolio for AV integrators—has announced a complete range of equipment that bridges the needs of home entertainment systems and the wider range of smart home technologies.

Since all those solutions increasingly depend upon a robust home network infrastructure, that's where Luxul's solutions deliver complete, simple-to-deploy, professional-grade IP networking solutions for use by custom installation professionals, without the complexity associated with traditional networking gear. Luxul recently introduced a wide range of solutions for high-performance networks in smart homes, including a new wireless mesh solution and cloud remote monitoring and management system.



Luxul expanded its range of wireless access points with the XAP-1610 model—Luxul's fastest ever—featuring advanced Wave 2 4×4 MU-MIMO 802.11ac technology and 2.4 and 5 GHz beamforming for data rates up to 3167 Mbps.

Luxul's Epic Mesh kit comes with two dual-purpose mesh nodes that can each serve as either a router or satellite. Simple to install, integrators turn one of the two nodes into a router in under two minutes using the Luxul Easy Setup App. The other becomes the satellite node, which is placed around the residence to create a powerful wireless mesh. There is no need to run additional wires—integrators just plug the satellite node into an available outlet. Each node in the network can talk to all the others, forming a point-to-point mesh for fast, reliable, truly wireless Internet.

An innovative tri-band (2×2, 2×2, 4×4) antenna array enables two 2×2 bands for client connectivity and a third powerful 4×4 5 GHz dedicated backhaul channel to ensure strong connectivity back to the router, so homeowners get fast Internet anywhere in the home. Luxul leverages the most current technology available with Wave 2, utilizing MU-MIMO for this high-performance Wi-Fi mesh solution, and further meets the needs of integrators by providing eight SSIDs for networking flexibility.

Additional satellite nodes can be incorporated as needed to accommodate installations of any size, and if one node falls offline, the others still have a path back to the router node, maintaining a connection to the rest of the network.

Finally, the new Luxul Cloud Remote Monitoring and Management solution is a free service to integrators remotely managing customers' systems. The new system makes it simple for integrators to remotely monitor their clients' networks, get real-time notifications of failures, and quickly resolve issues. With cloud management, integrators can access their customers' Luxul networks and connected devices at any time—from anywhere—in map or list views via an intuitive web interface. Site-level indicators provide the online/offline status of connected devices and the availability of firmware updates, while customizable alerts notify integrators of issues via text or email.

Luxul also introduced its next-generation XWC-2000 wireless controller, featuring the company's exclusive Roam Assist technology to ensure the active roaming of mobile client devices and simple deployment of up to 32 wireless access points (AP) for large installations. In wireless networks that utilize more than one wireless AP, the XWC-2000's Roam Assist technology ensures that mobile devices are always connected to the best AP for the highest performance. This enables users to move freely throughout their home without the frustration of intermittent whole-home audio and slow Wi-Fi speeds.

Newly deployed APs are automatically added for plug-and-play installation, and the controller installs in minutes, using a simple, intuitive setup wizard.

**www.luxul.com**

## Josh.ai and Steinway Lyngdorf

# A New Luxury Home Entertainment Experience

Josh.ai, a Denver-based company with offices in Colorado and California, continues to deliver on its vision to expand the power of artificial intelligence in the home. Basically targeting a new type of home control alternative to traditional home automation controls and offering a new type of voice assistant dedicated to the residential integration market, Josh.ai has been establishing partnerships with almost everyone in the CEDIA space. Recently, the company unveiled a partnership with Steinway Lyngdorf, the leading brand in luxury audio.

With the integration, Steinway Lyngdorf high-end stereo and surround sound systems will now support the Josh.ai platform to control media content. The integration enables the convenience of Josh.ai's natural voice control and privacy-focused approach to home automation to fit into the design of the most refined residential projects.

As part of the Josh.ai Distributed AV workflow, Steinway Lyngdorf processors will auto-populate the web portal for configuration. Integrators are then able to efficiently assign input sources and output zones for whole-home media control. Josh.ai's Steinway Lyngdorf support offers a rich feature-set of functionality enabling users to switch inputs, control volume levels, distribute a single source to play across multiple zones, and route multiple sources across multiple zones verbally via Josh Micro or manually through the intuitive Josh.ai app.

Users will be also able to leverage voice commands to request their desired content wherever they want it. For example, saying: "OK Josh, raise the lights on the patio and listen to 'Bohemian Rhapsody' by Queen on the first floor." Josh.ai handles the content search using its available streaming services and applicable sources, activates the Steinway Lyngdorf processor, routes the requested content to each zone on the first floor, and sets the mood!

**www.steinwaylyngdorf.com | www.josh.ai**


### Wireless Speaker and Audio Association (WiSA)

# Wireless Support of Dolby Atmos Height Channels

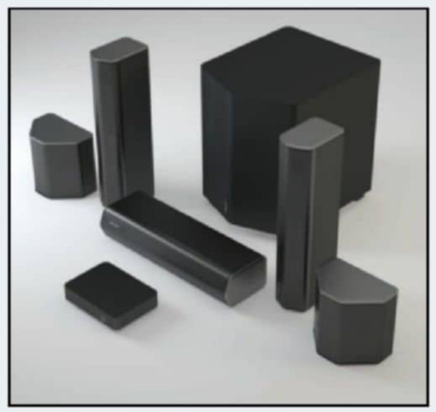
One of the most exciting announcements in home theater was the confirmation that the Wireless Speaker and Audio Association (WiSA), founded by Summit Wireless Technologies and supported by more than 60 leading consumer electronics brands, is finally implementing support of Dolby Atmos height channels. Until now limited to support 5.1 surround formats, the update will enable WiSA speaker partners to develop products that include immersive 3D audio experiences made possible by Dolby Atmos technology.

WiSA users will be able to assign front height, side height, and rear height audio channels to WiSA Certified speakers, making it easy for consumers to configure their systems for Dolby Atmos content during setup.

The announcement, will no doubt contribute to foster the adoption of the wireless audio technology even further and contribute to WiSA's expansion in home theater installations, by making the placement of height speakers for real Dolby Atmos experiences much more practical (sometimes the only option) in any home.

The ability for a user to select Dolby Atmos height channels will be included in the latest transmitter and speaker firmware, scheduled for release in 2020. Existing WiSA products can also take advantage of this feature through a firmware update.

**www.wisaassociation.org | www.summitwireless.com**



Enclave Audio introduced a THX Certified and WiSA Certified wireless home theater system at CES 2020, meeting consumer demand for more immersive home audio experiences with the convenience of wireless. Enclave Audio offers the CineHome II and CineHome PRO WiSA Certified wireless multi-channel surround speaker systems and the new CineHub wireless audio transmitter.

### ELAN by Nortek

# Networked Home Audio Distribution

Continuing to transform the smart home experience three decades after its inception, ELAN by Nortek Security & Control recently introduced the new ELAN IP Amplifiers range for whole home audio with support for Dante audio networking.

From simple media rooms to the largest luxury homes, ELAN intelligent home control and automation solutions offer an expanding range of options adopting all the latest technologies.

The new ELAN Intelligent Touch Panels features support for face recognition and a microphone-array that enables voice control, while the high-resolution display is ideal for monitoring and playback from ELAN Surveillance cameras and NVR. These panels can be integrated with the latest ELAN Video Doorbell, which integrates motion analytics to eliminate false alerts that cause many users of doorbell cameras to turn off the notices altogether. The new Doorbell camera also integrates with ELAN Surveillance for video recording and monitoring.

Another innovation is the ELAN Management Cloud, providing integrators full visibility and control over all of their intelligent system deployments and enabling smoother installations with more service opportunities.

The new ELAN IP audio distribution amplifier line with Dante audio networking adds flexibility and reduces installation costs for multi-room audio projects.

## IP Amplifiers

Among the latest new solutions in audio distribution, the new ELAN Audio over IP (AoIP) range of amplifiers simplifies and decentralizes audio installations by eliminating long speaker cable runs. And as primary TV sources are moving to set-top streaming boxes (Apple TV, and Roku), the range—including multi-channel amplifiers and local source pre-amps—also leverages Dante digital audio networking to give installers the ability to make use of these local sources in multiple rooms.

While Dante audio networking adds flexibility and reduces installation costs for multi-room audio projects, the audio is delivered between devices over the network, replacing expensive analog cabling with low-cost category cable.

**www.elanhomesystems.com | www.nortekcontrol.com**

Control4

## New and Expanded Smart Home OS 3

Control4, the fast-growing smart home solutions company, recently announced its merger with SnapAV and launched its new Control4 Smart Home OS 3 in mid-2019. The core product that allowed the Salt Lake City, UT, company to become a leading name in residential integration, was updated with more than a thousand new capabilities across lighting and whole-home music, video, security, comfort, and more.

Earlier in February 2019, Control4 acquired Switzerland-based NEEO, creators of an acclaimed smart home remote. With that acquisition, Control4 is now able to offer incredible remote controls, touch panels, keypads, and next-generation interaction devices across the smart home. The elegant Neeo Remote was quickly updated to leverage the power of Control4 Smart Home OS 3 and unify the whole experience to control home media and whole smart home.

Control4 Smart Home OS 3 was designed to unify hundreds of connected devices within the home, control them all from a single platform, deliver the personalization homeowners want, and facilitate the professional support smart home technology needs.

The new version introduced robust performance and functions to the OS, also expanding integration with the latest solutions from brands such as Honeywell, Sony, Sonos, Lutron, and Vivint (just to name a few), as well as control smart locks and even sprinklers, all on the same platform. In fact, Control4 OS 3 works with more than 15,000 devices and has nearly 8,000 products with Control4-SDDP software embedded.

On the entertainment front, OS 3 redefines multi-room control, making it easier to access the most popular streaming services with Active Media located at the top of the screen, and the all-new Sessions control that enables users to configure any room and easily start a video stream. OS 3 also added native support for streaming high-resolution audio in Control4 systems with the addition of Master Quality Audio (MQA), allowing homeowners to stream better-than-CD quality music through TIDAL Masters streaming services. The new OS 3 also expands the tools available for dealers and integrators, reducing repetitive tasks, and speeding up deployment.

**www.control4.com**

The latest Neeo Remote for Control4 touchscreen remote is a highly refined control of home entertainment experiences when combined with the powerful Control4 Smart Home OS 3 platform.

A Control4 Smart Home OS 3 touchscreen with a family room view on a custom background

Acoustic Geometry

## Starfield Acoustic Treatment Ceiling Tiles

Sometimes acoustic treatment can be converted into something very special when integrated with home entertainment spaces. Residential integration companies found this to be a clever way to differentiate their most sophisticated designs, often by combining acoustic panels with controllable lighting.

Acoustic Geometry offers an interesting approach with its StarField Ceiling Tiles, an acoustical ceiling panel with a realistic starry sky effect. This innovative, sound management product is ideal for home theaters, media rooms, studios, and even restaurants/bars and commercial businesses looking to inspire a "wow" factor.

Designed to add the beauty of the night sky to any room with a drop-in ceiling, the tiles are an affordable alternative to the company's original, custom-only StarField Ceiling Panels, which require professional installation. StarField Ceiling Tiles are available in small (2'×2') and large (2'×4') sizes for standard 15" to 16" ceiling grids. They connect to main lighting control systems through an RS232 adaptor.

Simple installation and acoustical functionality are key features of the StarField Ceiling tiles, and the fabric-wrapped fiberglass panels improve the quality of sound in any space by taming reverberation and ambient noise. The design itself is completely customizable, allowing for constellations, planets, shooting stars, galaxies, and even fireworks.

Custom-sized and shaped panels are available upon request. In addition, Acoustic Geometry offers several optional add-ons including shooting star and acrylic moon/planet inserts for an even more dramatic night sky look. All designs simulate "twinkling" with dim, medium, and bright stars for a natural starry sky appearance.

This new solution complements Acoustic Geometry's acoustic and noise-control products for a wide range of professional audio, residential, and business applications.

**www.acousticgeometry.com**

# Outdoor and Garden Speakers

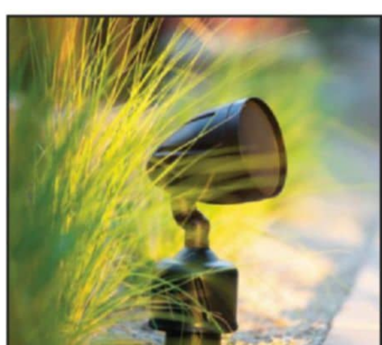
Transforming outdoor decks, patios, swimming pools, and garden spaces into extensions of a home is increasingly a valued proposition for those homeowners who are lucky enough to have such spaces. And residential integrators have long explored that field with fully integrated and sometimes fully automated solutions, which obviously explore the combination of scenic lighting with entertainment systems. Audio is increasingly one of the most demanded options and many users quickly discover the pleasures of upgrading that outside experience to high audio levels.

As an extension to the topic of this Market Update, we decided to highlight three examples of brands and products for those applications. From affordable to sophisticated, these solutions are purposely specified and ready to deliver the best performance in those challenging environments.

## MSE Audio Rockustics

MSE Audio is the parent company of SoundTube Entertainment, Soundsphere, dARTS, Phase Technology, Induction Dynamics, SolidDrive, and Rockustics. Several of these brands offer solutions that are ready for outdoor installations, but Rockustics specializes in outdoor and garden speakers. Recent product introductions from Rockustics include the SquareRoot 6.5 Coaxial Planter Speaker and the Rockustics Cherry Bomb full-range garden speaker—both purposely designed to enhance outdoor audio experiences.

The Kansas-based company launched the Rockustics SquareRoot Planter Speaker as a fully weatherproof, two-way, omnidirectional loudspeaker featuring weep holes to accommodate the watering of live plants, which is a very original proposition. Within this 17″×17″ planter, there is a 6.5″ polypropylene woofer and a 0.75″ Mylar dome tweeter, the SquareRoot 6.5 is designed to provide a full 360° of omnidirectional audio coverage. Constructed with rotation-molded, UV-stable polyethylene, the SquareRoot 6.5 is built with quality materials to withstand rain, frost, snow, and ice.



Rockustics Cherry Bomb is a premiere outdoor solution, providing full-range audio in one unit.

The Rockustics SquareRoot, a fully weatherproof, two-way, omnidirectional loudspeaker, doubles as a planter designed to house live plants.

Expanding its range of speakers for large-scale installations in gardens, the Rockustics Cherry Bomb is a premiere outdoor solution, providing full-range audio in one unit. Designed to blend into existing landscaping, the Cherry Bomb is a fully weatherproof three-way, direct-burial speaker designed to blend into outdoor landscaping without compromising audio quality.

Featuring a 10-year warranty, Cherry Bomb's patented three-way design includes a 5″ double-chamber, band-pass subwoofer, a 4″ woofer and a 0.75″ tweeter. Ideal for outdoor pool areas, patios and courtyards, residential backyards, and outdoor venues, the Cherry Bomb is an 8 Ω speaker with a frequency response of 42 Hz to 22 kHz, and a maximum program power of 100 W.

By including the subwoofer in the same cabinet that houses the midrange and the tweeter, the Cherry Bomb provides exceptional full-range sound without running into the typical phasing issues generated when installing a subwoofer in a different location from the other speakers in a traditional system. This makes installation much faster, easier, and provides cleaner, more accurate sound once installed.

## Dayton Audio

The Dayton Audio IO65XT 6-1/2″ IP65 speakers, together with the new IOSUB 10″ IP66 subwoofer are a perfect proposition for increasing audio fidelity outdoors on both commercial and residential installs. The IO65XT weather-resistant speakers are available in black or white finishes, with both 70 V/100 V taps and 8 Ω bypass, adding a passive radiator to extend the low-end response. When more bass response is needed, the new IOSUB2 10″ high excursion subwoofer with exclusive acoustic tuning will extend the response even more.

The Dayton Audio IO65XT full-range speakers are perfect for patios, pools, gardens, porches, offices, bars, and hotels. Featuring a high excursion 6.5″ polypropylene woofer and 1″ silk dome tweeter, the IO65XT speakers challenge the notion that outdoor speakers lack in low-end response. Without sacrificing durability, the IO65XT is able to produce extended bass by using a passive radiator design to provide a surprisingly deep, tight bass response while keeping the enclosure small.

Solid, affordable, and featuring unique designs, the Dayton Audio IO65XT 6-1/2″ IP65 speakers and IOSUB IP66 subwoofer for indoor/outdoor installations are another great example of how the brand's catalog continues to expand with differentiated products.

Flaunting an IP65 ABS enclosure (available in both black and white), the IO65XT is perfect for outdoor use. The water-resistant build allows for the speaker to be used in outdoor applications, even in unfavorable environmental conditions. Included brackets allow for the speaker to be mounted either vertically or horizontally with the ability to pivot, making installation simple and quick.

And any outdoor/indoor setup with the IO65XT speakers can be expanded by adding some additional bass with the Dayton Audio IOSUB 10″ subwoofer. The IOSUB utilizes a robust high-excursion polypropylene woofer with a stitched rubber surround to produce a powerful yet accurate low-end. Enclosed in a compact, easy to hide weather-resistant enclosure, the subwoofer also offers exclusive tuning by Dayton Audio, producing low-end down to 50 Hz with 300 W of power.

Remaining visually unobtrusive, and built to withstand Mother Nature, the IOSUB is an IP66-rated enclosure, providing even more flexibility for installation locations. Its compact footprint makes it easy to keep out of view, under decks or behind bushes. Since the subwoofer fires downward, sitting on four legs, it is extremely stable and also features an anti-theft eyebolt for a secure, permanent install.

## Coastal Source

For homeowners wishing to enjoy the best and most sophisticated experiences outdoors, Coastal Source is a recognized name. Still a family-owned company, Coastal Source focuses on tough outdoor products and systems to improve the quality of outdoor life. With decades of experience in integration and originally combining all the products and systems necessary to produce turnkey installation projects, Coastal Source gradually invested in creating an original portfolio of branded products. After years of extensive research and experimentation, it has the technology and methodology to produce a full line of quality outdoor lighting and audio systems, available through a network of certified dealers.

While the most visible side of Coastal's catalog remains in landscape lighting (we highly recommend the company's catalog for inspiration!), the company has been continuously investing in innovative and patented outdoor speaker solutions that stand out in sound quality.

As Coastal Source aptly states, "Why should beautiful sound be confined to your home?" In particular these days, when an existing home audio installation can be connected to cover the whole home and can be controlled from a smartphone. Coastal Source Outdoor Audio products can be tailored to fit any outdoor environment and include the traditional Rock Speakers for the garden areas, to Bullet Speakers for more fidelity and impact in creating outside experiences.


The Coastal Source Ellipse Bollard range is an impressive, high-quality solution for outdoor sound.

But the brand's range of solutions really stands out with its Coastal Source Ellipse Bollards line of outdoor speakers. These come in two-way or three-way versions, and are designed as true high-end audio systems for outdoors and have nothing to fear in direct comparison with some of the best indoor home audio products.

The large Coastal Source 3-Way Ellipse Bollard speakers are in/above ground systems that are not meant to be hidden, but are instead designed to stand-out in outdoor patios, decks, and open entertainment areas, where they can be seen and heard. Inside the uniquely shaped cabinets are bi-amplified full-range speakers, adding an 8″ subwoofer in the three-way models.

During CEDIA 2018, Coastal Source introduced a very high-end version of the 3-Way Ellipse Bollards, using an impressive ribbon planar vertical array. These massive speakers are also available in models equipped with light-changing color LED surfaces, for the ultimate all-in-one outdoor experience. The $7,500 line source and light-changing speakers were an obvious attention grabber at the company's booth in Denver, and it makes sense, since the system is also ready for parties.

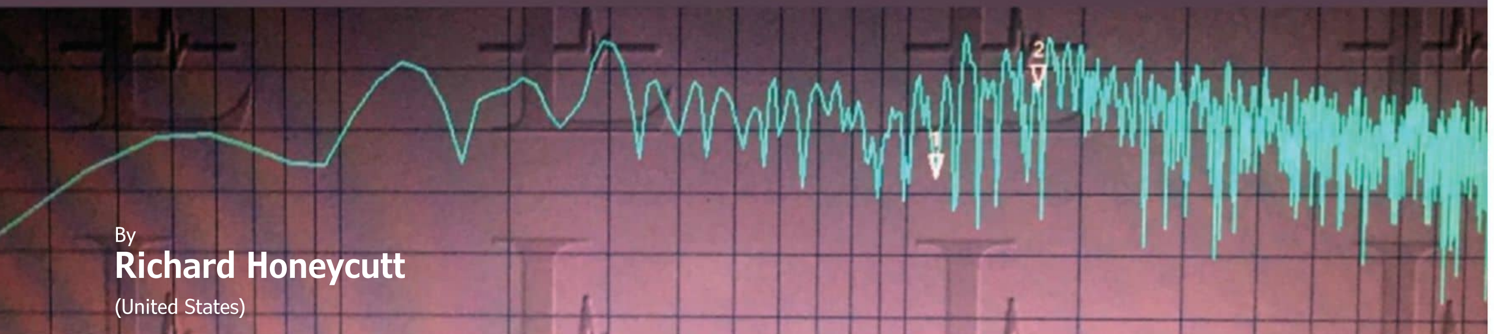
The 3-Way Ellipse Bollard Ribbon Planar speakers are designed for use in applications where directional sound is desired, benefiting from a large driver area, and offering higher performance, higher efficiency, and higher fidelity. Each speaker stands 54″ high and 12″ in diameter, weighing in at 88 lb. each, featuring 12 3″ ribbon tweeters, six mid-range 4″ woofers, and a 10″ laminate cone subwoofer.

The large enclosure Ellipse Bollard size allows for completely full-range speakers that can stand up to 1,000 W and use high-quality materials and a construction built to "Defy the Elements"—a Coastal Source trademark. And all Coastal Source sound systems use the company's patented Plug+Play cabling system that ensures superior reliability and speed of installation.


Images from Coastal Source's booth at CEDIA 2018 show the Ellipse Bollard ribbon line array speakers for outdoor installation.

**www.mseaudio.com | www.daytonaudio.com | www.coastalsource.com**

# Testing Electroacoustical Systems



By
Richard Honeycutt
(United States)

Figure 1: The frequency response of a room measured with a swept sine wave stimulus

In this article, our acoustical expert provides a primer on how to measure the frequency response of an electroacoustical system, using a calibrated microphone and an appropriate metering system.

When people ask about my career, and I tell them I'm an acoustical consultant, I usually get one of two responses: a deer-in-the-headlights look; or the follow-up question, "Oh, you install sound systems?" With a bit of further discussion, I usually am able to clarify what an acoustical consultant does, but in fact, sound systems are often involved. In my particular case, this can happen if a client complains that his or her theater, sanctuary, lecture hall, listening room, or studio "doesn't sound right"—a description that may be associated with any of a multitude of ills. If the venue has a sound system, verifying its correct operation becomes part of my job. Since I also consult in sound system design, commissioning sound systems after installation is another part of my job.

Sound systems typically involve electronic, electrical, electromagnetic, and acoustical components, so testing their operation in their native habitat is by nature electroacoustical testing. My introduction to electroacoustical testing occurred when I was in undergraduate school, and served as a student engineer for the 50 kW classical music campus radio station. Local donors had just enabled the station to buy a new stereo mixer and studio monitor system. The faculty member who served as station manager wanted to be sure the monitor was accurately reproducing what we were broadcasting.

At this stage of technology, the task mainly involved frequency-response testing, since the only user controls that could ever get misadjusted were bass and treble controls. We could have measured other performance aspects (e.g., harmonic and intermodulation distortion), but with new professional-grade equipment, such measurements were unnecessary.

## Frequency-Response Testing

Testing the frequency response of an electroacoustical system is always stimulus-response testing. A known stimulus having a known response—usually equal energy per Hertz (i.e., white noise) or equal energy per octave (i.e., pink noise). Then the system's response is measured using a calibrated microphone and some sort of metering system. The particular stimulus chosen depends upon the electroacoustical system being tested.

In the case of the radio studio, the acoustical part of the system was a room having dimensions of about 24′ × 11′ × 8′. A room this size is susceptible to room modes that would create very large peaks and dips in the frequency response, and these irregularities would show up in any measurements using a swept-sine-wave stimulus, such as the signal generator we used for testing electronic equipment (see **Figure 1**).

Many test platforms can smooth response curves made from swept sine waves. Since it is very difficult for humans to perceive frequency response irregularities less than 1/3 octave in bandwidth, 1/3 octave smoothing is commonly used. However, in years past, 1/2 octave or 1 octave pink noise was available on test phonograph records. Consequently, we chose 1/2 octave pink noise as our stimulus. A half octave is wider than any of the room modes, so the peaks and dips would pretty much average out with the pink noise stimulus.

Keep in mind that this was the mid-1960s, so Real-Time Analyzers (RTAs) were not yet an option. The available equipment for the job was a National Association of Broadcasters (NAB) phonograph test record with pink noise tracks, and a sound level meter (SLM). The SLM included the calibrated mic, and the meter needle indicated the level.

Each 1/2 octave band provided about 30 seconds of pink noise, so the process entailed placing the SLM at the desired listening location, starting the record, and writing down the level reading for each band as the various tracks on the record changed the frequency bands. The SLM was always set for flat frequency weighting and "slow" response time to average out the unsteady level of the pink noise waveform.

In this particular case, we never applied any EQ to our program material, except for occasionally high-pass filtering live interview shows from our studio to eliminate p-popping and mic bumps/thumps. For this reason, we were not concerned with the effect of room modes. With modest acoustical absorption provided by the ceiling and the carpeted lobby floor, modes were not a problem. A recording studio or control room would have required more attention, especially with regard to modes.

In a large venue (e.g., a lecture hall, theater, or sanctuary), frequency response measurements usually serve two purposes: to make sure the speaker positioning provides accurate sound coverage at all seats, and to discover whether there are any locations where acoustical effects degrade the sound in specific frequency ranges.

In commissioning a sound system using directional speakers—either horn systems or line arrays—we must find the areas at the edge of the coverage where the high frequencies are weaker or where the low frequencies are too strong. For finding high-frequency coverage edges, white noise is more effective than pink noise, since the white noise spectrum emphasizes high frequencies. Great care should be taken in using white noise with sound systems in order not to overdrive and damage the high-frequency speaker components. Pink noise works well for investigating low-frequency response.

In some cases, such as low-frequency irregularities produced by improper subwoofer location, using a narrower analysis bandwidth such as 1/6 or even 1/12 octave can help find response aberrations. Such defects cannot be remedied by even the best equalization, but must be dealt with by repositioning the speakers or, in extreme cases, by acoustical modifications. In one large sanctuary I designed, the baptistery was located at the end of the central aisle, just inside a domed window. The dimensions of the window were such as to make the dome act as a very effective focusing element, especially at voice frequencies. Had we not foreseen and avoided this focusing problem in the design stage, someone would eventually have had to be called in to measure the system's frequency response in the particular seating area affected by the dome's focusing.

The most common use of frequency-response testing of large venues is in setting system equalization. Nowadays, the test instrument is usually a RTA (see **Photo 1**), which is most often a combined 1/3 octave pink noise generator and a filter bank with a level display for each band. An RTA can be a stand-alone instrument or a software-based application operating on a computer or cell phone.

The gold standard frequency response for the electroacoustical system of any venue is ruler-flat response. Usually during system commissioning, the EQ is set to achieve this response as closely as possible. Once this has been accomplished, someone with excellent listening skills needs to "voice" the

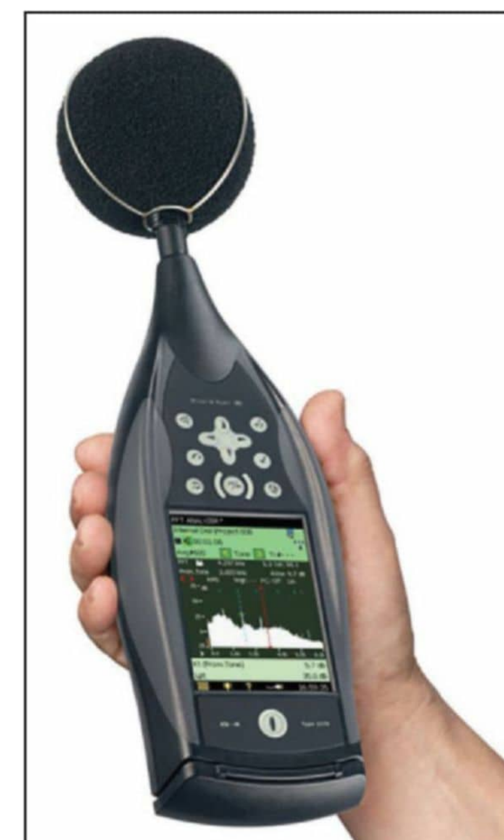


Photo 1: The Brüel & Kjaer Model 2270 SLM also functions as a high-resolution RTA. (Image courtesy of Brüel & Kjaer)

system. This may involve adding a "disco bump" of perhaps 3 dB to 5 dB in the range of 120 Hz to 180 Hz, in order to make the bass better meet typical user expectations. Or it may entail backing off the high frequencies a bit to keep voices from sounding harsh. These "voicing" adjustments are largely driven by the purpose, expected program material, and natural acoustics of the venue: contemporary music venues are more likely to require disco bumps, and rooms with a high reverberation time (RT) at high frequencies may need some high-frequency trimming to tame possible harshness.

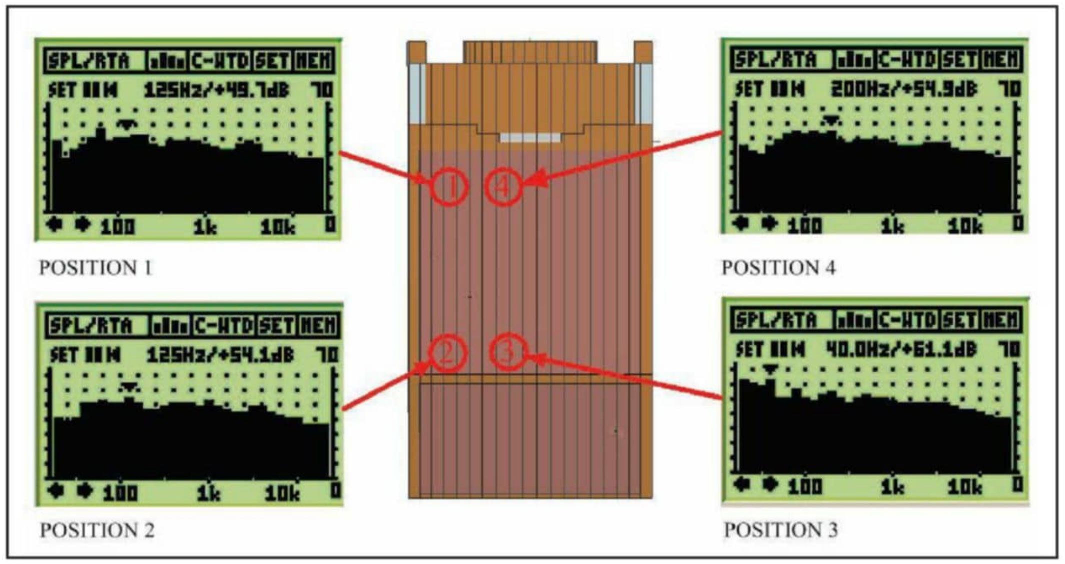



Figure 2: A 1/3-octave RTA response varies from one seating location to another.

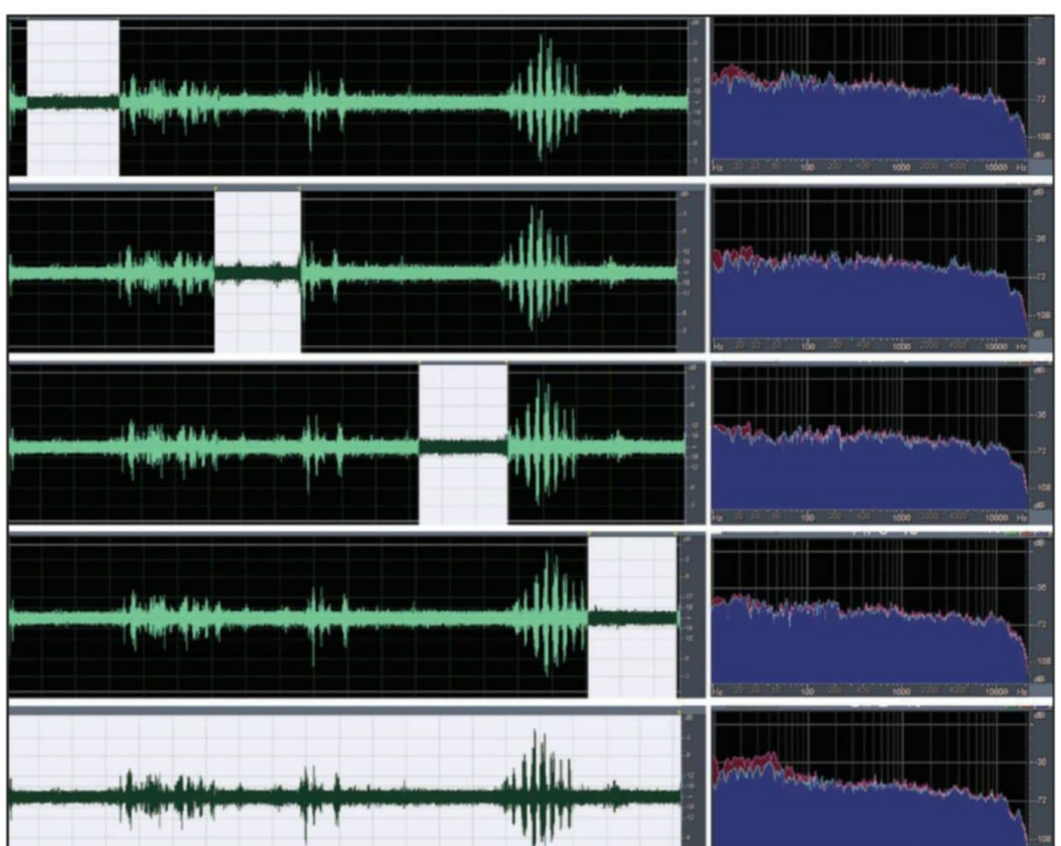


Figure 3: Here are the .wav files (left-hand column) and FFT results (right-hand column) from a walkabout test.

## Actual Sound Quality

In an ideal world, the speaker coverage would provide identical sound quality to every seat, but in the real world, this is never the case. **Figure 2** shows the frequency response measured at four seating locations in a medium-sized auditorium. The floor plan of the auditorium is shown in the center of the figure, and the four seating locations are identified. In this particular case, the four RTA plots are reasonably similar. However, close examination shows variations, particularly in the low-frequency region. Thus the question arises: At what seating location do we equalize the sound to be flat? (Or, which will be the best seat in the house?)

In theory, we could measure the response at a variety of locations, then plot and average the measurements and set the EQ to correct this average response. While having the advantage of not requiring additional equipment, other than a single RTA and calibrated microphone, this is a very time-consuming procedure.

Consultant Bob Thurmond introduced the idea of carrying around a portable audio recorder as he walked throughout the venue while playing pink noise on the sound system. He would stop at representative seating locations, which he would identify on a floor plan of the venue. Then he could play the entire recording into analysis software and see the average response on screen. If he noticed any anomalies, such as abnormal external or HVAC noise at any location, he could excise the results for that location from his composite wave before analyzing.

**Figure 3** shows the .wav file recording from such a "walkabout" in the auditorium whose RTA results are shown in Figure 1. The portions of the .wav file corresponding to each measurement location are highlighted, and the Fast Fourier Transform (FFT) of that portion of the .wav file appears to the right of the .wav file. The irregular portions of the .wav file correspond to times when the digital recorder was being carried from one location to another. The measurement locations are shown in numerical order from top to bottom, with the complete walkabout at the bottom.

As was true for the RTA results, the walkabout results do not show a great difference among seating locations, although there are noticeable variations in low-frequency response. A venue having poorer acoustics or a less well-designed sound system would show greater variation in frequency response from one location to another.

## Multiplexers

The most elegant way to measure the average frequency response is to use multiple calibrated microphones feeding a multiplexer. (See **Photo 2** for one example of a multiplexer.) The multiplexer essentially performs the function of Bob Thurmond's "walkaround": sampling the spectra of different locations by "time-sharing." The output of each microphone is switched into the output signal of the multiplexer for a specific period of time. The multiplexer's output then contains an equal-length spectral contribution for each microphone. The EQ can then be set while the measurement is being taken, allowing instantaneous observation of the effects of adjustments.


Photo 2: The Murideo microphone multiplexer allows simultaneous measurement of the frequency responses at up to four microphone locations.

## Measurement Standards

The process of selecting appropriate measurement locations has been addressed in the standard "Audio Coverage Uniformity in Listening Areas," now officially called AVIXA A102.01:2017. As stated in the standard:

"The procedures described in this Standard are to be applied to sound reinforcement and audiovisual (AV) presentation systems. These systems are implemented in a variety of applications, including conference rooms, training rooms, classrooms, auditoria, theatres, houses of worship, and other venues where sound reinforcement is employed. Additionally, the metrics and classifications in this Standard may be used to establish design criteria for new systems."

Section 4.2 of the standard specifies the use of octave-band measurements, no frequency weighting, and 50-ms time-windowing. Even if the walkabout method or a multiplexer and multiple test mics are used, selecting the measurement positions according to this standard is a wise practice, leading to uniform results from one project to another.

Other types of electroacoustical tests will be discussed in a future Sound Control article.

# Extracting Input Referred Op-Amp Noise Models

By
**Michael Steffes**
(United States)

In this article, our audio expert uses three different amplifier setups to emphasize the different noise terms, sweep those measurements across frequency, then solve for what the input noise terms need to be to match the measured data.

Using the noise preamp described in the September 2020 *audioXpress* issue, we can now proceed to take the different swept frequency spot noise measurements using a spectrum analyzer. This is usually done as log frequency steps, where the spectrum analyzer is set to a zero span and an average of the measured spot noise voltage is developed at each frequency step. That measured noise is referred back to the device under test (DUT) output pin, dividing by the net gain, then squared to get the power. The calibration noise power at each frequency (measured with the noise preamp input terminated with a 50 Ω termination) is subtracted to isolate on just the DUT output noise.

The most involved op-amp model will be for a current feedback device, such as the THS3491 (see Reference 1). The remaining output spot noise voltage for any op-amp is always a combination of the terms shown in Equation 1 (Equation 2, from Reference 2), where noise gain (NG) is the noise gain = 1 + Rf/Rg. This is summing each of the noise powers at the output, adding them assuming they are uncorrelated, then taking the square root to get spot output noise voltage as a root sum of squares (RSS). For many voltage feedback amplifiers (VFAs), the input current noise terms are negligible and drop out (particularly for CMOS or JFET input types).

$$e_o = \sqrt{\left(e_{ni}^2 + 4kTR_s + \left(i_{bn}R_s\right)^2\right)NG^2 + \left(i_{bi}R_f\right)^2 + 4kTR_f NG}$$

Equation1

The first three terms are physically on the V+ input pin while the last two are on the inverting side. This is a model where the aim is to force all the extremely complex internal noise terms into the three op-amp input terms $E_{NI}$, $I_{BN}$, and $I_{BI}$. **Figure 1** shows this analysis model with the resistors set to the first row of the test conditions shown in **Table 1**.

Every noise measurement at the spectrum analyzer will be a combination of the terms in Equation 1. The aim is to take three different amplifier setups to emphasize the different noise terms, sweep those measurements across frequency, then solve for what the input noise terms need to be to match the measured data. All of this math is done in powers that add linearly to the total output noise powers. Once the calibration power is subtracted from the measured noise reflected to the DUT output pin, the first step is to remove resistor noise powers contributing to that total. Those two power terms are given in Equation 2, where the first term combines the Rf and Rg noise to the output and the second term is the Rs noise times the NG.

$$4kTR_f NG + 4kTR_s \left(NG\right)^2$$

Equation2

These resistor noise terms will shift slightly with temperature where 4kT = 16.44e-21J at 25°C

(kT is 4.11e-21 for T = 25°C + 273 Kelvin). For the best measurement extraction, determine the resistor temperatures and adjust the 4kT term for the actual value during test.

What we are left with now is a matrix solution for the input referred noise terms given the power gain for each term and the resulting three total output noise power measurements at each frequency. One common error is to adjust those external resistors in a way that brings in either bandwidth limiting or peaking across frequency on the noise measurements. That was actually done erroneously in characterizing the THS3491 noise where the 5 kΩ Rs and Rf selections in Table 1 introduced bandlimiting on the output noise. Table 1 shows the three sets of resistor values used for the THS3491 lab characterization.

Going back to each noise power term in Equation 1, the power gain matrix for this set of test conditions is shown in **Table 2**.

The 5 kΩ values were unnecessarily high. When Rs = 5 kΩ, both the non-inverting current noise and resistor noise terms on the V+ input will be bandlimited to 27 MHz due to the 1.2 pF parasitic input capacitance on the V+ input (see Reference 1). Being a current feedback op-amp, raising the feedback Rf to 5 kΩ bandlimited all the noise terms to the output above approximately 20 MHz as shown by the small signal bandwidth simulation for Test 2 shown in **Figure 2**.

## Excel Matrix Inverse Function

The excel matrix inverse function is applied to values in Table 2 to generate the scale factors to multiple each of the three measured noise powers at the different resistor value from Table 1. That command looks like this:

{=MINVERSE(C23:E27)}

Where the target cells are the 3×3 matrix of power gains (see Table 2), and this places the matrix inverse values into a 3×3 matrix.

Multiplying each column element of the matrix inverse across the first row times the three measured noise powers at each frequency will give an input referred voltage noise power. Square rooting that will yield the input referred spot noise voltage at each frequency step. Repeat this for the second row of the matrix inverse to get the input referred non-inverting current noise power and the last row for the inverting input current noise power.

The actual measured and input referred data is "noisy" even with a lot of averaging. **Figure 3** shows the measured, and best fit, non-inverting current noise for a particular THS3491DDA used for this test.

The flat region of this curve matches the 15 pA/√Hz specified noise. The main aim for this measurement is to locate the nominal 1/f corner going down in frequency. The higher frequency roll-off is a measurement artifact and not an actual current noise roll-off. For most datasheet curves, the best fit 1/f model curve is used flattened off to

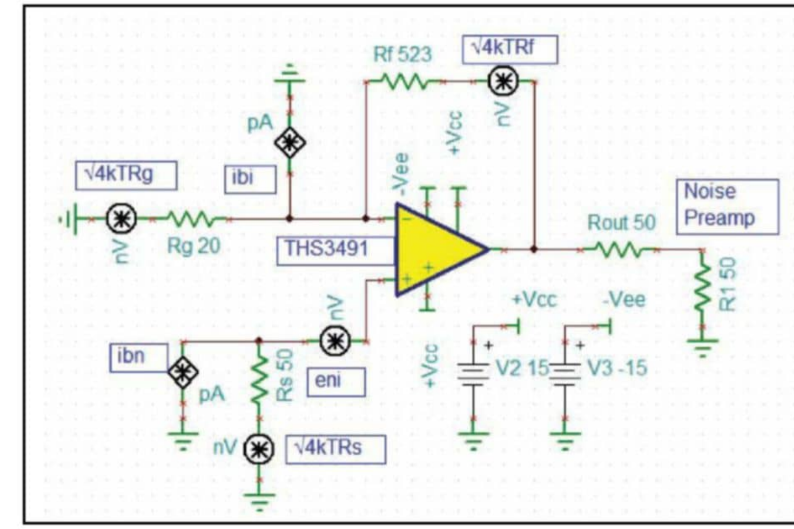



Figure 1. This general op-amp noise model shows each of the elements contributing to Equation 1.

| Test | Rg | Rf | Rs | Noise Gain |
|---|---|---|---|---|
| Test 1 | 20 Ω | 523 Ω | 50 Ω | 26.15 |
| Test 2 | 5 kΩ | 5 kΩ | 50 Ω | 2 |
| Test 3 | 523 Ω | 523 Ω | 5 kΩ | 2 |

Table 1: Resistor values used in characterization

| Test | $E_{ni}$ Gain | $I_{bn}$ Gain | $I_{bi}$ Gain |
|---|---|---|---|
| Test 1 | $(26.15)^2$ | $(26.15 \times 50\ \Omega)^2$ | $(523\ \Omega)^2$ |
| Test 2 | $(2)^2$ | $(2 \times 50\ \Omega)^2$ | $(5\ k\Omega)^2$ |
| Test 3 | $(2)^2$ | $(2 \times 5\ k\Omega)^2$ | $(523\ \Omega)^2$ |

Table 2: Noise power gains for each test

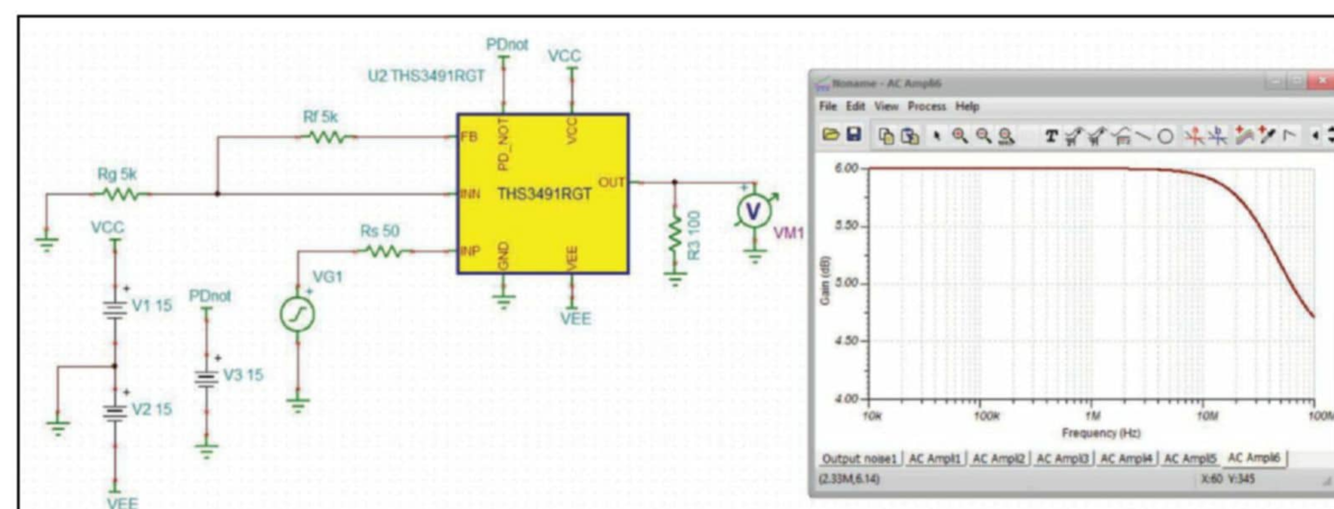



Figure 2: The THS3491 small signal response is shown for the Noise Test 2 values from Table 2.

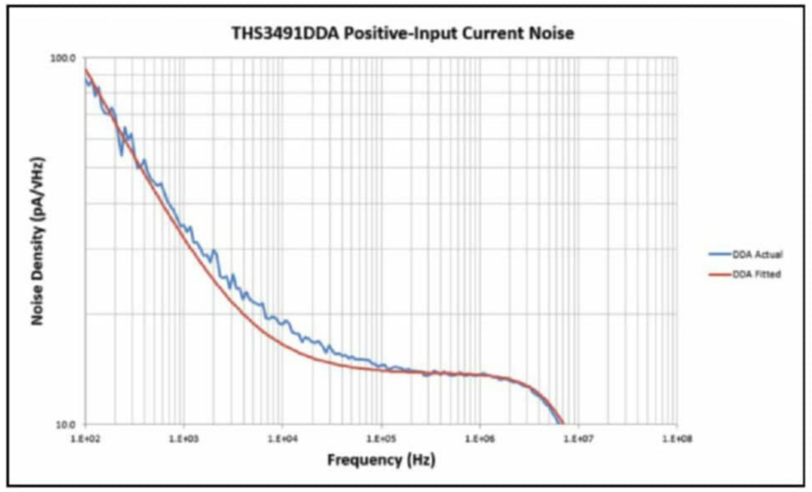



Figure 3: This is bench measured and best fit line for THS3491 non-inverting input current noise.

| | Test 1 (Eo1) | Test 2 (Eo2) | Test 3 (Eo2) |
|---|---|---|---|
| Rf= | 600 Ω | 600 Ω | 600 Ω |
| Rg= | 600 Ω | 100 Ω | 100 Ω |
| Rs= | 20 Ω | 20 Ω | 200 Ω |
| Noise Gain = | 2 V/V | 7 V/V | 7 V/V |
| Expected Eo | 13.3 nV/√Hz | 19.4 nV/√Hz | 30.9 nV/√Hz |

Table 3: Three sets of R values used in simulation for model input noise extraction

| THS3491 | Eni (nV/√Hz) | Ibn (pA/√Hz) | Ibi (pA/√Hz) |
|---|---|---|---|
| Flatband Numbers→ | 1.78 | 14.77 | 20.27 |
| 1/f corner value | 2.52 | 20.89 | 28.66 |
| Approx. 1/f freqeuncy (Hertz) | 2840 | 4606 | 2417 |

Table 4: Input referred noise terms for the THS3491RGT 2018 TINA model

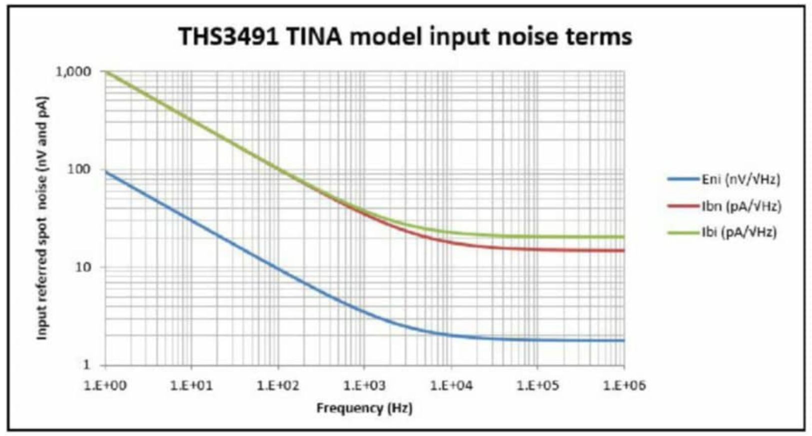



Figure 4: This graph shows the input referred noise over frequency in the THS3491RGT 2018 TINA model.

remove the higher frequency roll-off. Occasionally, a datasheet will show that higher frequency roll-off in the measured data (see Reference 3)—most often that is a measurement gain roll-off and not a noise term roll-off.

## Extracting Noise Curves

This same methodology may be used to extract the noise curves from the simulation models. More direct measures can be made in simulation, but stepping through the three resistor matrix also allows those values to be tested in simulation for flatness prior to bench measurements. **Table 3** shows a typical R selection for the validating the THS3491 model. Here, the feedback Rf is held fixed at a good value for a broadband flat response for this CFA device. Each test set up acts to significantly increase the expected total output noise giving good matrix solutions. These three tests are executed in simulation for the total output noise, that is ported to excel, and the steps outlined above (except for the calibration noise removal) are executed to extract the input referred model.

The models produce a flatband noise value with a 1/f shape. The simulation data can be analyzed for the 1/f corner frequency for each term as shown in **Table 4**. Note the THS3491 model generated input spot noise voltage flat at 1.78 nV—slightly higher than the datasheet 1.7 nV. The current noise terms match very close to the datasheet.

**Figure 4** plots these input noise models from 1 Hz to 1 MHz looking very similar to the datasheet plot. Keep in mind these are only typical numbers and parts will have a distribution around these shapes. Flatband noise can typically have ±10% variation where 1/f corners have been reported to have as much as ±3x variation. Also, the 1/f model only approximates each individual device noise going to lower frequency as shown in Figure 3.

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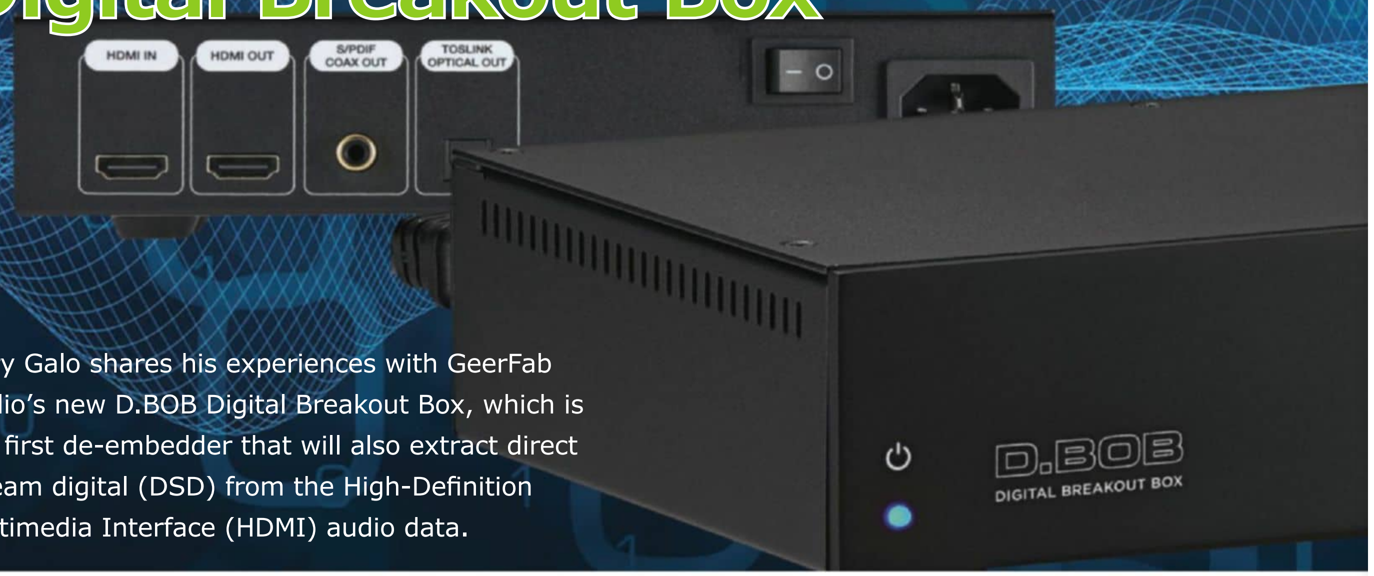
# GeerFab Audio D.BOB Digital Breakout Box




Gary Galo shares his experiences with GeerFab Audio's new D.BOB Digital Breakout Box, which is the first de-embedder that will also extract direct stream digital (DSD) from the High-Definition Multimedia Interface (HDMI) audio data.

By
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Price: $999 US

The most demanding audio enthusiasts normally use an outboard digital-to-analog converter (DAC), fed from a digital audio player that's used as a transport. The interface for PCM sources is usually S/PDIF coaxial in dedicated audio systems. Like many audiophiles, I use a universal disc player, currently an OPPO Digital UDP-205, but I've previously owned the BDP-105 and the BDP-95 players, and my wife still has a BDP-93 in her system (see Resources). Sadly, OPPO pulled the plug on its audio business in April 2018, but many of us will continue to use these excellent players as long as they function.

Unfortunately, the recording industry's paranoia about copyright protection—not entirely unjustified—has made it impossible for most players to send high-resolution datastreams to outboard DACs via S/PDIF. High-resolution Blu-Ray and DVD-Audio discs with copy-protection are normally down-sampled to 44.1 kHz or 48 kHz. But, players normally leave the HDMI datastream unchanged, since there's no way an end user can access the audio data.

The work around for this problem has been an HDMI audio de-embedder. A de-embedder extracts full PCM resolution from the HDMI datastream and sends it to an outboard DAC via S/PDIF. I reviewed the KanexPro HAECOAX de-embedder in the July 2016 issue of *audioXpress*, and have since used de-embedders sold by Tripp-Lite (P130-000-AUDIO) and Monoprice (10251). Monoprice calls its unit an HDMI Audio Converter, and the unit it carries is also sold as the ProLink HA-110SA. It's currently my favorite of the three, providing cleaner, more detailed sound than the KanexPro and Tripp-Lite.

## DSD Solution

Unfortunately, none of the HDMI audio de-embedders have been able to extract the DSD datastream from SACDs. I've normally used my OPPO Digital players as stand-alone units when playing SACDs, feeding the player's analog outputs directly to my preamp. Some audiophiles set their players to convert DSD into 24-bit PCM, so they can use their outboard DACs. But, the conversion is a compromise, partly because every player I've used converts the DSD to 88.2 kHz. In order to preserve something close to the full bandwidth of DSD, and consequently the lowest phase distortion in the audible spectrum, the conversion should be to 176.4 kHz. I have lamented the inability to decode my SACDs in my outboard DAC, native mode, in several of my reviews.

GeerFab Audio to the rescue! This Wisconsin-based firm is a the sister company of GeerFab Acoustics, which manufactures acoustical treatment products. Owner/designer Eric Geer shared my own frustrations about SACD playback and decided to take matters into his own hands. After three years of research and development, the result is the D.BOB Digital Breakout Box (see **Photo 1**). On PCM sources, the D.BOB functions exactly like any other HDMI audio de-embedder. PCM audio is extracted from the HDMI datastream and output over S/PDIF in full resolution—up to 192 kHz/24-bit. But, that's where the comparison ends. The D.BOB is the first de-embedder that will also extract DSD from the HDMI audio data. DSD is encoded on an S/PDIF datastream using the DoP protocol. DoP is short for "DSD over PCM" and was initially developed by dCS, a well-known manufacturer of high-performance digital audio converters. A detailed explanation can be found on the DSD Guide website, but dCS has summarized the system nicely:

"The original idea for DoP was invented by dCS in 2011. It involves taking groups of 16 adjacent 1-bit samples from a DSD stream and packing them into the lower 16 bits of a 24/176.4 datastream. Data from the other channel of the stereo pair is packed the same way. A specific marker code in the top 8 bits identifies the datastream as DoP, rather than PCM. The resulting DoP stream can be transmitted through existing 24/192-capable USB, AES, Dual AES or SPDIF interfaces to a DoP-compatible DAC, which reassembles the original stereo DSD datastream COMPLETELY UNCHANGED. If something goes wrong and the datastream is decoded as PCM, the output will be low-level noise with faint music in the background, so it fails safely. This can happen if the computer erases the marker code by applying a volume adjustment."

DoP is widely used in computer audio systems via a USB connection to an outboard DAC. But, the D.BOB is the first product to implement DoP between a stand-alone SACD player and a DAC. What's kept other audio manufacturers from doing this is fear of lawsuits from the recording industry. GeerFab was able to market the D.BOB by keeping the copy protection intact, a problem that occupied a great deal of product development time. GeerFab notes that one SACD manufacturer—Channel Classics—expressed enthusiasm about the D.BOB. They believe that offering audio enthusiasts a way to get better sound from their SACDs will help sell more discs—a very sensible attitude. It will probably

Photo 1: This is the front-panel view of GeerFab Audio's D.BOB Digital Breakout Box. (Photo courtesy of GeerFab Audio)

help sell outboard DACs, as well, since many now support the DoP protocol.

The D.BOB's connectivity is simple, as shown in **Photo 2**. There are two HDMI connectors—an input from the player and an output. The output is a pass-through that can be fed to a television, for players that don't have separate audio and video outputs. There are also two output connectors—S/PDIF coaxial and Toslink optical. DoP will work via Toslink, but the most exotic cables aren't always the most reliable.

If possible, use coax for the best performance. GeerFab wanted to include a USB output, but it wasn't possible to keep the copy protection intact via USB. The D.BOB is normally powered directly from an AC wall outlet, but GeerFab also includes a 12 VDC input, for those who might wish to experiment with high-end, outboard power supplies. There's also a Micro USB connector on the rear panel, for firmware updates.

## Design Details

The D.BOB design is centered around two chips made by Lattice Semiconductor (see **Photo 3**). The Sil9533 Port Processor is the main HDMI decoder chip, which extracts PCM audio data from the HDMI datastream and sends it out via S/PDIF. The HDMI pass-through functions are also supported by this chip. But, the Sil9533 does not support DSD. The DSD datastream is extracted from HDMI, and encoded as DoP, by a Lattice LCMX02-1200HC Field Programmable Gate Array (FPGA).

All the DSD extraction and encoding to DoP is handled by the custom programming of this chip. From the datasheets for these chips, it appears that the FPGA chip is connected as a loop in and out of the Port Processor. The Port Processor's S/PDIF output is fed to a 74HC04 Hex Inverter, used as the S/PDIF driver, divided between the coaxial and Toslink outputs. The internal switching-mode power supply feeds 12 VDC to the main PC board, where it's regulated to 5 VDC and filtered.

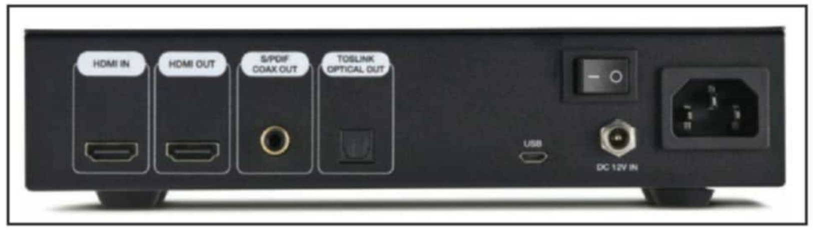


Photo 2: This is the rear panel of the D.BOB. Digital Audio connections, from left to right, include HDMI In and Out, S/PDIF Coax Out and TOSLINK Optical out. The unit can be powered from an AC power outlet using the IEC power connector on the far right, or from an external 12 VDC power supply using the DC power connector located below the power switch. (Photo courtesy of GeerFab Audio)

## Performance

When I received my D.BOB review sample, it didn't work with my Benchmark DAC3 HGC. When I played SACDs, the Benchmark DAC front panel LEDs indicated that it was receiving both PCM and DSD datastreams. Eric Geer and John Siau at Benchmark worked together to help solve the problem. In the end, the factory in China discovered a programming error in the FPGA chip.

Some DACs weren't bothered by it, but the Benchmark was (the Benchmark uses a very strict implementation of the DoP protocol, requiring "by the book" timing of the markers that identify the datastream as DSD). This wholesale reprogramming of the FPGA chip isn't possible using the rear-panel USB connector—it must be done through a six-pad connection on the PC board. My D.BOB was updated by GeerFab at his Wisconsin facility.

As I see it, there are three areas of comparison that are important in evaluating the D.BOB, which I'll address one at a time. First, how do SACDs played through the D.BOB and my Benchmark DAC3 HGC compare to using my OPPO UDP-205 as a stand-alone player? Anyone who has gone from a stand-alone player to a good outboard DAC with PCM sources knows what to expect, and the improvement with DSD sources is no different.

With SACDs and DSD files, nearly every aspect of performance is improved. The soundstage is larger and more precise, with considerably more depth. There is great inner detail, and a greater sense of the connecting acoustical space between instruments and sections of the orchestra. The bass is more extended and better defined, and the treble region is more open and airy. Dynamics are also improved.

Second, how do SACDs played through the D.BOB and the Benchmark DAC, as native DSD, compare to having my OPPO player convert them to 88.2 kHz/24-bit PCM? Converting SACDs and DSD files to 88.2 kHz/24-bit and feeding the PCM datastream to the Benchmark has always been a mixed bag. The soundstage is improved—larger and better defined, with greater depth. But, the treble region loses some of the analog-like smoothness that's one of the hallmarks of DSD. Using the D.BOB with native DSD conversion gives you the best of both worlds, and more. The positive aspects of the PCM conversion are further improved and the analog-like smoothness is delivered in full measure.

Third, how do DSD 64 files played through the D.BOB compare to the same files played from a phone or tablet, fed via USB to an outboard DAC? The D.BOB will also decode DSD files played on an external drive—I use the OPPO UDP-205 as the server. I have some of the same files on an extra memory

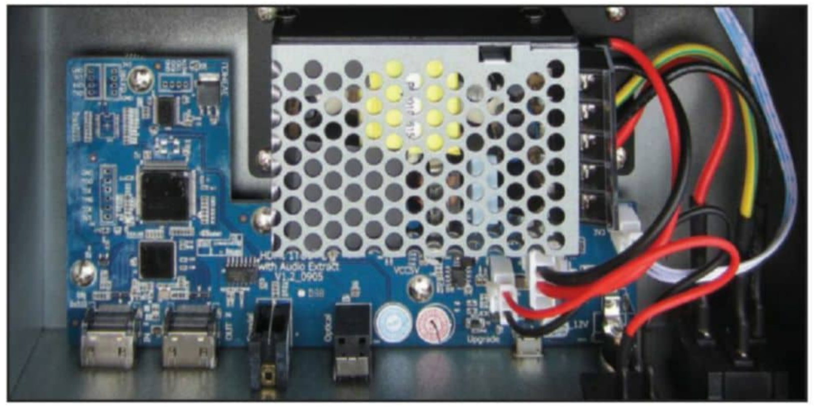

Photo 3: This is an inside view of the D.BOB. The design is built around two chips manufactured by Lattice Semiconductor, shown on the left just above the two HDMI connectors. The smaller of the two square chips is the Sil9533 Port Processor, which provides HDMI I/O and PCM audio extraction. The larger square chip is the LCMX02 FPGA, which extracts DSD datastreams and encodes them to S/PDIF using the DoP protocol. The shielded, switching-mode power supply is located in the center.

## Resources


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DH Labs | https://silversonic.com

**USB Audio Player Pro**
Audio Evolution Mobile Studio | www.extreamsd.com/index.php/products/usb-audio-player-pro

card in my Samsung Galaxy Note 4, which I play with an app called USB Audio Player Pro. I also own two commercial SACDs that were made from those files. The phone is connected to the Benchmark DAC3 with a short RadioShack USB "On The Go" adapter and a DH Labs Mirage USB interconnect.

I always suspected that the phone was not an ideal source for file playback. I have done several comparisons of PCM material played on my phone, with the same files played through my OPPO. The OPPO simply delivers a cleaner and more refined sound. The same is true of DSD sources.

File playback on the phone reveals many of the improvements I hear with native decoding of SACDs and DSD files through the OPPO and D.BOB, especially the sound staging and spatial qualities. But, the sound through the phone is somewhat veiled and lacking fine detail. Playback through the OPPO and D.BOB removes a fine layer of grunge that I hear with the phone. As an HDMI de-embedder for PCM sources, up to 192 kHz, the D.BOB is at least as transparent as the Monoprice, and perhaps even a bit better.

## DSD Quirks

DSD, especially over DoP, is not without its quirks. The D.BOB is an entirely new implementation of DoP, and users may experience a minor glitch or two with some SACD players. With my OPPO UDP-205, if I start by playing an SACD or DSD file after powering up, I hear a click through the system before playback starts. It's about the level of a loud click on an LP, but nothing that could be damaging. I get the same click if I play a 44.1 kHz source first, and then play a DSD source. I don't hear the click going from any other PCM sampling rate to DSD. On occasion, when I start an SACD or DSD file, I'll hear soft static on top of the music, similar to what dCS describes in the statement I cited earlier. If I eject the disc and reload, the problem disappears. Once an SACD properly starts, it will play fine from beginning to end.

GeerFab loaned me a Sony UDP-X700 and I had no problems with it. Benchmark uses an older OPPO DV-980H and Siau says they haven't had any problems (they kept their DV-980H because it doesn't down sample high-res PCM sources via S/PDIF). I did experience both the click and the static issues with my wife's OPPO BDP-93, and GeerFab reports occasional glitches with his OPPO BDP-105. So, these issues may be unique to certain players, and would not appear to be a problem with the D.BOB.

Siau notes that bursts of noise are a common occurrence when playing DSD, especially when the playback system is toggled between PCM and DSD. It's exceedingly difficult to build soft mute circuits in a 1-bit DSD environment. For this reason, players tend to do a hard switch when starting a new track. When DSD is passed from box to box, these problems are magnified. The DAC2 and DAC3 have soft mute circuits that are active when playing DSD, but pops, clicks, and bursts of noise can be produced by upstream DSD players and/or interface devices.

Correct audio settings on your player are critical to getting the best performance, and you may have to experiment. Make sure that the selected HDMI audio output is the one connected to the D.BOB. SACD modes should be Stereo and DSD, and HDMI audio format will usually be Auto. I consider the issues I experienced to be minor—far outweighed by the remarkable performance improvements the D.BOB delivers with DSD sources.

## Conclusions

For audiophiles who are SACD fans and who use an outboard DAC, the D.BOB is the product we've awaited. The remarkable improvement in sound quality afforded by decoding SACDs, native mode, with a high-performance outboard DAC, is remarkable. I'm sure, like me, you'll spend many hours rediscovering your SACD collection.

The price is admittedly a bit high—higher than many of the players with which it's likely to be used. But, being the first of its kind, considerable expense has gone into its development. Contrary to Sony's hopes, the SACD format has not become mainstream—it remains a specialty, audiophile format. Whether other equipment manufacturers will introduce similar products, now that GeerFab has broken the ice, remains to be seen. If you're serious about SACD playback, you're going to want the D.BOB.

## About the Author


**Gary Galo** retired in 2014 after 38 years as Audio Engineer at The Crane School of Music, SUNY at Potsdam, NY. He now works as a volunteer in the Crane Recording Archive doing preservation, restoration, and digital transfer of vintage Crane recordings. He is also a Crane alumnus, having received a BM in Music Education in 1973 and an MA in Music History and Literature in 1974. Gary is a widely-published author with more than 300 articles and reviews on both musical and technical subjects, in over a dozen publications. Gary has been writing for *audioXpress* and its predecessors since the early 1980s. He has been an active member of the Association for Recorded Sound Collections (ARSC) since 1989, and a frequent recording and book reviewer for the *ARSC Journal*. He has given numerous presentations at ARSC annual conferences, many of which have been published in the *ARSC Journal*. He was the Sound Recording Review Editor of the *ARSC Journal* from 1995-2012, and co-chair of the ARSC Technical committee from 1996-2014. Gary has also published numerous book reviews in *Notes: Quarterly Journal of the Music Library Association*, written for the *Newsletter of the Wilhelm Furtwängler Society of America*, *Toccata: Journal of the Leopold Stokowski Society*, and he is the author of the "Loudspeaker" entry in *The Encyclopedia of Recorded Sound* in the US. He has also written several articles for *Linear Audio*. He is a member of the Audio Engineering Society, the Boston Audio Society, and the Société Wilhelm Furtwängler.

# Dealing with Communication Challenges

By
**J. Martins**
(Editor-in-Chief)

In a series of editorials for *The Audio Voice* newsletter, J. Martins wrote extensively about the communication challenges that we all have had to endure when faced with the reality of shelter-at-home restrictions and everyone working from home. In this article, he addresses the lack of a commutation protocol for organizations, and the problems of audio quality in existing forms of communications and conferencing tools we use.

When the global COVID-19 pandemic was announced and caused countries one after another to impose shelter-at-home restrictions to contain the virus, we were all forced to completely rely on existing home communication networks, personal devices, and our own technology skills. Suddenly, a massive percentage of the world's offices were shut down and working employees (the lucky ones...) sent home. No one had planned for this.

No one had the time to even think of meeting solutions or those marvelous Unified Communications and Collaboration (UC&C) systems, which were simply left disconnected in the now-empty office buildings.

Suddenly, remote collaboration was quickly built on top of "informal networks" using Skype, WhatsApp, WeChat, Facebook Messenger, or even really obscure social media tools and apps that people used daily. And these extended to communications with fellow workers and to freely discuss business, exchange internal documents, and more. Gradually, people also resorted to using "free" videoconference tools, such as Zoom, without any thought about security or productivity.

## Communication Systems

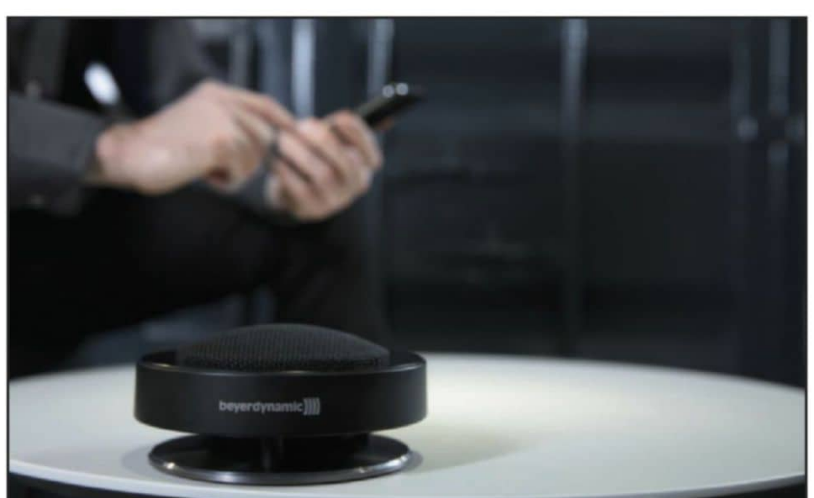
Traditional communication systems work mainly based on a few existing protocols agreed upon by the International Telecommunication Union (ITU), which were largely challenged by the interests of mobile network operators worldwide, and the massive transition to a global system of interconnected Internet Protocol (IP) networks and Internet services. With mobile networks, the IP convergence, and consumers changing devices almost every year—all happening faster than any other transition in the history of telecommunications—we could even say that the surprise is that things even work at all!

The lack of quality assurance on these connections was the unfortunate result. And profit-centered strategies also dictated that telecommunication systems set a relatively low bar for voice quality, even though technology has progressed immensely on that front. Telecom companies mostly ignored progress, and continue to offer a "good-enough" voice quality standard (GSM quality, basically). Even after the introduction of the 3GPP Enhanced Voice Services (EVS) audio codec for VoLTE, which allowed super wideband

(SWB) audio quality for mobile phones, only a handful of telecom operators globally introduced the service—surprising, given device support was in place, including from the world's best-sellers and main equipment brands.

And before 3G cellular networks, when the world's communication lines were transitioning from "digital" (ISDN anyone?) to IP, the industry already had improved voice codecs that didn't require increased bandwidth and not much additional investment in infrastructure to telecom operators. Yet, these remained stubbornly firm on their "no one is going to be able to tell the difference anyway" attitude.

Whatever the communication software used, manufacturers such as Poly have certified communications headsets such as this Poly Voyager 4200 UC headset for Skype and Microsoft Teams integration, as well as many new headset and USB speakerphone products to be launched in 2020. But all those were designed to be used in slightly different scenarios than what we have now.




Introduced in 2018 as "the future of team communication," this beyerdynamic Phonum fully integrated speakerphone was designed as a mobile, cable-free communications system for conferences and meetings, and the best possible voice and sound quality. Unfortunately, there are not enough of these used by working-from-home individuals yet.

Skype, a company founded by a group of Nordic entrepreneurs and a group of Estonian programmers and software developers who had created Kazaa, a peer-to-peer file sharing application, was not even the first solution available. But in 2010, with more than 600 million worldwide users, Skype was certainly the first Internet-based communication solution to be massively adopted by everyone who had a computer and a broadband connection—and it certainly took the office world by storm.

The reality was that, by using available software for PC and Macs, users were for the first time able to experiment the extremely high-quality audio of modern voice codecs. Skype and all its competitors used a mix of existing technologies and protocols, combined with new and more or less proprietary solutions, including audio codecs.

With the subsequent acquisitions of Skype by eBay and later in 2011 by Microsoft, Skype quickly evolved its audio codecs from G.729 to SILK, which was developed in-house. SILK is now open-source and available royalty free, and was included in Opus another open-source effort. SILK and Opus, are now widely used, and the voice-over-IP (VoIP) solution adopted by everyone, from WhatsApp to Zoom. Shortly after Skype's global success, regular folks had iPhones with FaceTime.

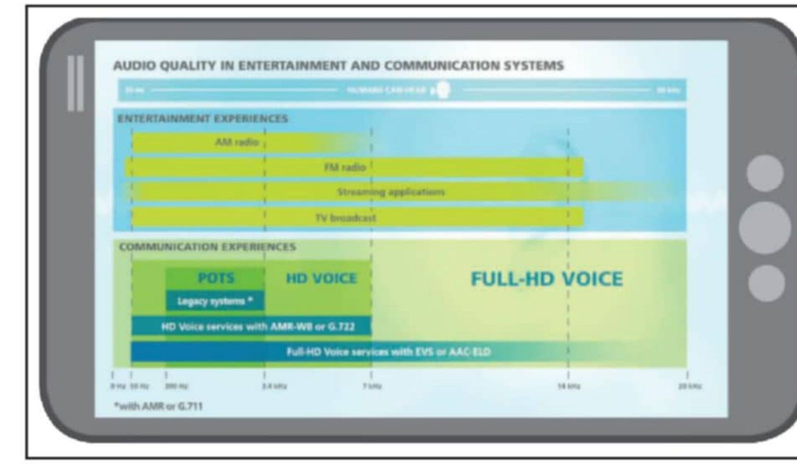
## Audio Quality

The obvious communication problem of bad quality audio is mainly the result of an inadequate use of resources, lack of adoption of (existing) standards and technologies, and deficient systems integration. Over the years, so many technology and standardization efforts were put in place that they should have improved communications and audio quality in those applications, but the truth is, we are far away from what could have been possible. It shouldn't have to be that way.

Evidently, broader-band infrastructure, 5G networks, superior Wi-Fi, and Bluetooth 5 wireless designs, the latest generation built-in microelectromechanical systems (MEMS) microphones, beamforming, and digital signal processing abilities, all work toward improved communications, whether they are applied to consumer products—including those for gaming—or for "office" and professional use. Current "professional" UC&C products are a nice way to illustrate how the industry was focused on providing solutions to existing practices, which suddenly became obsolete with the pandemic.

There was a time before and a time after Skype. When Skype reached critical mass, people wondered why their Skype-to-Skype calls sounded so much better than what they were used to on the phone. One of Skype's early codecs was the internet Speech Audio Codec (iSAC), an adaptive wideband audio codec that allowed twice the frequency bandwidth of standard phone calls, and added mechanisms to compensate for the delay, jitter, and packet loss challenges of IP networks. The transition to SILK allowed another dramatic improvement, offering even better performance under variable conditions, and very high quality in ideal conditions (and the reason why Skype communications were quickly adopted even for broadcast contributions—not surprisingly, most of the guest calls we see on TV these days are supported by dedicated Skype hardware on the receiving side, with direct IP connections and even balanced analog outputs).

Meanwhile, competition multiplied in the desktop and mobile space, starting earlier with Apple's FaceTime. The introduction of FaceTime with the iPhone 4, in 2010, was effectively the next big revolution in communications, introducing the world to mobile "video chat"—a new concept that was seen by many as effectively "taking Skype mobile." And




The Enhanced Voice Services (EVS) codec was a multi-industry effort, optimized for operation with voice and music/mixed content signals, with extended bandwidth, and was standardized in 2014 as part of the 3GPP project. Its potential remains to be discovered by consumers. (Source: Fraunhofer IIS)

the audio quality in FaceTime was great not only because of the typical superior hardware/software integration of Apple's devices, but also because FaceTime introduced the world to the next big thing in audio codecs: AAC-ELD.

This was part of the AAC family of codecs from the Fraunhofer Institute for Integrated Circuits IIS. Since 2007, the MPEG-4 AAC-Low Delay (AAC-LD) codec had already been used by Cisco to transmit

When everyone was forced to temporarily abandon workplaces and shelter at home, communications and collaboration changed forever. Skype is still Microsoft's best-positioned service to compete in the post-COVID-19 landscape.

audio in the Cisco TelePresence meeting solution. The Fraunhofer codec was praised at the time for providing "a realistic sound image through several channels of full-bandwidth, music-quality audio, with the low-latency delay needed for interactive conversations." The AAC-LD was an evolution of the AAC codecs used in the Apple iPod and digital television services, and satellite radio. It offered a similar audio quality to music codecs with coding delays of only 20 ms instead of 100 ms or more.

The AAC communication codec family was later expanded with the AAC-ELD and AAC-ELDv2 codecs,

The race is on! Facebook introduced Facebook Messenger Rooms to compete with Zoom and Google. Between WhatsApp and Messenger, more than 700 million accounts participate in calls every day. In many countries, video calling on Messenger and WhatsApp has more than doubled.

allowing for maximum speech and sound quality, raising the audio quality for any communication application and defining a new standard for high-quality audio in conferencing systems. Thanks to the native integration of AAC-ELD in iOS, Mac OS, and Android, app developers and service providers could also easily build communication apps with support for the full audio frequency range.

Today's IP communication software (web conferencing) evolved from the early technologies pioneered by Skype and Fraunhofer IIS, and offer an extremely high-quality level, under ideal conditions—allowing us to feel like we are in the same room as the person to whom we are talking. Frequently, the lack of ideal conditions is not so much the connection, but the end-point interfaces, as previously mentioned.

Full hardware, software, and built-in audio processing integration allows a trouble free experience. That is why Apple also developed an integrated hardware and software solution optimized from the microphones to the speakers for audio and video conferencing that rivals really expensive corporate systems. No doubt, the importance of having a fully integrated audio processing chain is crucial. Without the hardware/software integration there's no chance of optimizing any quality parameters, except on the actual audio transmission flow, where it is all is about compensation.

When people use modern smartphones for VoIP in a broadband network, the audio quality can be surprisingly good—a huge contrast with what we normally experience on "regular" phone calls. And yet, when people use the devices in speakerphone mode, the communication is also affected by the speaker distortion (by itself far higher than the transmission distortion), bad acoustics, and surrounding noise. Other problems can result from network/bandwidth issues, causing low intelligibility and frequent cuts.

Videoconference and meeting software, such as the now popular Zoom, typically target business applications, and it seemed a good idea to "certify" hardware devices—hence, they created a Zoom Certified program. But this just means that the devices receiving certification "have been tested with the software." Only in certain cases, such as for Zoom Rooms and Zoom Phone services, are certified IP devices tested to be fully interoperable with the application. Similar certifications exist for Microsoft Teams, Skype for Business, Cisco, Avaya, and other UC&C systems, but they all target professional applications in optimized environments, such as meeting and huddle rooms.

When a proper bidirectional connection is established, maybe the software can assure automatic signal processing optimization features according to a device profile, such as in fully integrated UC&C solutions. This is possible even in speaker mode for dedicated audio conference systems, or using a USB connected headset—the USB Implementers Forum did a lot of work in its latest updates to the USB specifications, knowing that video/audio conferencing was one of the key applications.

An additional problem seems to be now that people want to go all wireless, and of course all the latest business- and office-oriented solutions are now Bluetooth-based. Of course, to work with a lot of existing office systems and old computers, these new wireless communication speakerphones and headsets use a USB dongle to manage the connection—in big part to ensure they work in a longer range, establishing a radio connection up to 98' (30 meters) with Class 1 supported devices. These new-generation office headsets even use active noise canceling and support wideband voice, which means that they could potentially optimize the whole experience, as long as the software understands what is connected.

With the ongoing transition to "outside the office," "work-from-home" and eventually "work-from-anywhere," communication systems need to be fully interoperable no matter the usage scenario. And with that interoperability, audio quality issues need to be addressed in a fully seamless way for users. Because there is no going back.

## Resources


"Codec for Enhanced Voice Services (EVS): The New 3GPP Codec for Communication," Workshop at the 140th Audio Engineering Societyy (AES) Convention 2016
www.aes.org/technical/documentDownloads.cfm?docID=548

Fraunhofer Institute for Integrated Circuits IIS—Communication
www.iis.fraunhofer.de/en/ff/amm/communication.html

H. W. Gierlich, "Communication Quality in Conferencing Scenarios—Principles and Optimization Potential," https://audioxpress.com/article/speech-quality-in-conferencing-scenarios-understanding-all-optimization-requirements

International Telecommunications Union (ITU), https://www.itu.int/en/Pages/default.aspx

"The AAC-ELD Family for High Quality Communication Services," Fraunhofer IIS Audio and Media Technologies, Technical Paper,
www.iis.fraunhofer.de/content/dam/iis/de/doc/ame/wp/FraunhoferIIS_Technical-Paper_AAC-ELD-family.pdf

"The History of Skype"
https://blogs.skype.com/wp-content/uploads/2012/08/skype-timeline-v5-2.pdf

# Enabling High-Quality Voice Conferencing

By
**Manfred Lutzky**
(Head of Audio for Communications, Fraunhofer IIS,
Audio and Media Technologies Division)

The spread of the coronavirus and the subsequent move by many companies to have employees work from home has driven the use of audio and video conferencing systems, and also phone calls, to new heights. This has led to an increasing amount of discussion about the audio quality of each application. However, the audio quality that emerges on the receiver's end depends on many factors—two major ones are the codecs used as well as their robustness.

Patterns of work and life have changed a lot over the last decades, especially the way we communicate. Digital real-time communication services had become essential working tools for meetings with remote participants long before the coronavirus emerged. The majority of these services—including most telephony today—are based on the Low-Delay IP-Streaming System Internet Protocol (IP). In particular, Voice-over-IP (VoIP) applications continue to develop in their scope and popularity, bringing flexible and highly cost-effective communications to both business and personal users.

## Evolution of Audio Quality in Communication

Let us first take a look at the origin of the teleconference: phone calls. For the longest time, the vast majority sounded quite muffled compared to other sources of audio. This was due to plain old telephone systems (POTS) providing only narrow band (NB) audio signals, meaning frequencies of up to 3.4 kHz of audio bandwidth only. This standard, which was set in the first half of the 20th century, did not take into account high-pitched voices of women and children, who were consequently difficult to understand on the phone. At such a narrow bandwidth, it is also difficult to distinguish the consonants f and s, for example. And ambient sounds may even be perceived as noise, affecting moods and the feeling of transparent communication—making long conversations exhausting and not very productive.

The so-called HD Voice services, introduced in the 2000s, then raised audio bandwidth to around 7 kHz, resulting in wideband (WB) transmission. However, while the speech codecs used in HD Voice services provide perceptibly better quality compared to legacy calls, they still transmit just less than half of the fully audible spectrum. Considering the capability of the human auditory system, higher frequencies of 16 kHz and more—relevant for high-fidelity sound—are still missing. In short, most phone services are simply deleting a huge part of the audible spectrum, which decreases speech intelligibility and speaker recognition, and requires additional listening effort in everyday telephony.

When it comes to bit rates, the codecs used in landline telephone services (G.711 and G.722) provide either NB or WB quality and both operate at 64 kbit/s. In contrast, the codecs used for NB and WB services in the mobile world (AMR and AMR-WB) both usually operate at bit rates around 12 kbit/s. Some mobile networks allow even higher rates for AMR-WB (e.g., 23.85 kbit/s), although the quality improvement compared to the default rates is rather limited. These communications codecs are highly optimized for speech signals and, as a consequence, their capability for coding other signals (e.g., music) is inadequate.

Also, signal delays used to lead to involuntary interruptions, impacting natural conversation flow. These delays, also called latencies, should ideally be kept to a minimum of 150 ms to 200 ms end-to-end. The end-to-end delay of a VoIP call is aggregated by several processing steps and components, such as acoustic echo cancellation, noise suppression, automatic gain control, routers, mixing of audio signals in convergence calls, transcoding at network boundaries, jitter buffer, and speech/audio coding. As it is very important to maintain a low total latency, it becomes crucial for every component of the communication chain to use this resource responsibly.

To enable service providers to shake off the limitations of legacy services, HD Voice services were enhanced (e.g., "HDVoice+" for mobile telephony) to include the quality levels of super-wideband (SWB) and full-band (FB). While SWB represents an audio spectrum of 16 kHz, FB contains all frequency components up to 20 kHz. These new services offer an unsurpassed level of quality, resulting in calls that sound as clear as talking to someone in the same room, or listening to high-quality digital audio. This is achieved by using hybrid solutions uniting the worlds of audio and speech coding.

One such hybrid is the latest 3GPP communication codec, Enhanced Voice Services (EVS), specifically designed to improve mobile phone calls in managed networks. Another is the Low Complexity Communication Codec/-Plus (LC3/LC3plus), which brings EVS audio quality to wireless accessories and landline phones. There are also optimized audio codecs such as Enhanced Low Delay AAC (AAC-ELD), tailored for over-the-top (OTT) voice services.

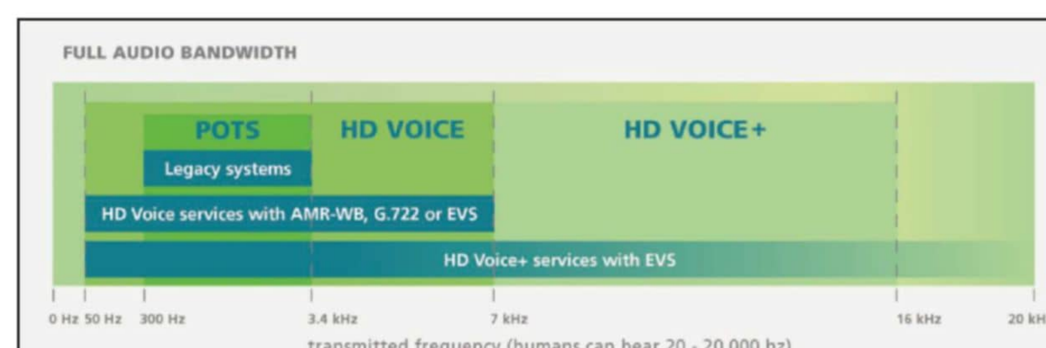

Call quality and codecs used today in mobile communications. The graphic shows how EVS significantly improves the audio quality over legacy codecs at popular mobile bit rates such as 13.2 and 24 kbit/s.

## AAC-ELD, EVS, and LC3/LC3plus: Designed for a "Same-Room Feeling"

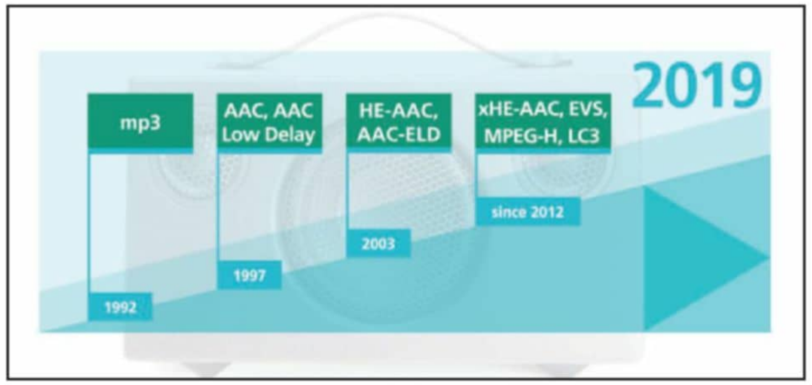
AAC-ELD is the communication codec that was specifically designed to meet the high demands of all-IP-based video conferencing and telepresence applications in terms of quality and latency. Natively supported in the mobile operating systems iOS and Android, AAC-ELD is based on the highly successful AAC audio codec, which has been in use in the Apple iTunes music store since 2001 and is deployed in billions of devices today.

In contrast to common speech codecs, AAC-ELD extends the application area from clean voice to a broad variety of source material, including speech, singing, music, and ambient sounds. It also delivers high audio quality at a wide range of bit rates (down to 24 kbit/s) and sampling rates up to 48 kHz. It is optimized for a low algorithmic delay, which is essential for natural real-time communication, and can handle mono, stereo, and multi-channel signals—all with latencies as low as 15 ms. Even lower delays of 7.5 ms at a 48 kHz sampling rate can be achieved by the low delay mode.

With the introduction of Long-Term Evolution (LTE), the fourth generation of mobile network standards, cellular phone networks switched to IP-based transmission. LTE is based on the GSM and UMTS standards, offering an all-IP architecture and

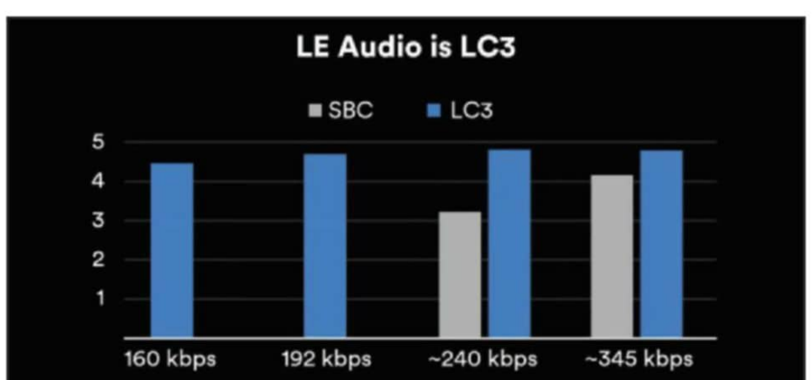



EVS bandwidth is compared to other voice technologies.




Four generations of audio technologies have been developed by Fraunhofer IIS.

low latencies. With the communication codec EVS, the international standardization organization for mobile telephony 3GPP and the GSMA have established a highly efficient audio codec solution for VoLTE services. EVS overcomes three major quality problems in mobile and landline telephone systems: limited audio bandwidth; poor performance on non-speech signals; and, most importantly, poor performance in the case of bad reception and overloaded networks. Combining state-of-the-art speech and audio compression technologies, EVS enables unprecedented audio quality for speech, music, and mixed content. The codec offers a wide range of bit rates from 5.9 kbit/s to 128 kbit/s, allowing service providers to optimize network capacity and call quality as desired for their services. Bit rates for NB and WB start at 5.9 kbit/s, while SWB audio quality is supported from 9.6 kbit/s on. This significantly improves the audio quality over legacy codecs, also at popular mobile bit rates such as 13.2 kbit/s and 24.4 kbit/s. Combined with the codec's low algorithmic delay (32 ms), EVS-to-EVS conversations enable users to enjoy a "same-room feeling." Another advantage is the codec's backward compatibility to AMR-WB, which is enabled by an interoperability mode: It allows seamless switching between VoLTE and circuit-switched networks when network conditions warrant a transition.




Bluetooth codec comparison (Source: Bluetooth)

The LC3 codec and its superset LC3plus bring EVS's SWB audio quality to Bluetooth Low-Energy (BLE) Audio-based accessories, VoIP services, and Digital Enhanced Cordless Technology (DECT) phones, thus enabling users of these devices to share in the feeling that the conversation is really taking place face to face. The codec meets the requirements of wireless communication platforms and accessories in terms of complexity, and it operates at low latency and low memory footprint. LC3 is currently being standardized in SIG Bluetooth and will be the audio codec for upcoming BLE profiles. LC3plus was standardized in 2019 as ETSI TS 103 634 and is included in the 2019 DECT standard, enabling DECT phones to deliver not only SWB, but also enhanced WB audio quality. At the same time, it doubles the number of telephones per base station and improves the coverage range at the edge of the cell as well as in environments with poor reception, thanks to its robustness.

## Increasing Robustness for Seamless Conversations

Transmission losses can occur in all kinds of network communications. Communication chains consisting of many different technologies are particularly prone to them and as a result, the speech packet loss rates (PLR) will add up and become clearly audible. Thus, it is essential to ensure the robustness of every component in the end-to-end chain. However, the requirements for error concealment and network adaptation rise with higher-quality audio connections: numerous challenges such as network delay jitter (when packets undergo a variable delay as they traverse the network) or packet loss have to be overcome to achieve a flawless audio conference. Packet loss issues in particular have an unmistakably negative effect on speech intelligibility. State-of-the-art audio codec technology is able to improve the quality and resilience of voice calls via mobile networks, OTT services, and VoIP and DECT devices.

Fraunhofer's communication audio codecs are equipped with overlapping frames and error concealment, which make it possible to use frame insertion/deletion in order to adjust the playout time to varying network delays.

The AAC-ELD codec resolves frame loss, in which the IP frame is discarded by the network, and thus, never delivered to the receiver through loss detection based on Real-time Transport Protocol (RTP) sequence numbers and subsequent error concealment. Based on the overlapping structure of audio frames in AAC-ELD, it is possible to maintain good audio quality despite packet loss.

As for network delay jitter, the recommended method is adaptive playout—offering an algorithm that estimates jitter on the network and adapting the size of the dejitter buffer in order to minimize buffering delay and late loss.

EVS comprises several unique concealment techniques, including a channel-aware mode (CAM) that uses partial redundancy, making this codec robust and enabling it to minimize the audibility of errors and quickly recover from lost packets. EVS's highly efficient jitter buffer management, which compensates for transmission delay jitter, as well as its minimized inter-frame dependencies, which avoid error propagation and provide for a fast recovery after lost packets, deliver a high-quality communication experience. EVS can tolerate a PLR of 6%, achieving the same quality of its forerunner AMR-WB at 1% PLR.

LC3plus implemented on VoIP or DECT handsets or Bluetooth headsets can minimize disruptions during phone calls, too, as the codec exploits techniques related to EVS and is extremely robust against voice packet loss and bit errors. In overloaded VoIP channels, the redundant transmission of LC3plus voice data verifiably ensures more stable phone calls. For DECT telephones, the LC3plus inherent tools for forward error correction were specially adapted to exploit typical characteristics of DECT links: LC3plus's channel coding permits the transmission of LC3plus payloads over heavily distorted DECT channels—enabling phone calls without interruption, even if the handset is far away from the base station.

## Conclusion

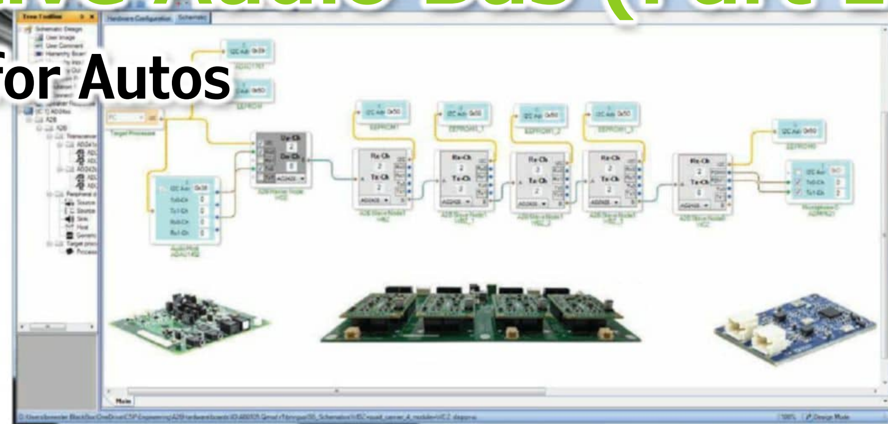
The audio quality of teleconferences should be as high as possible so as to make the conversation feel as natural as having all participants present in the same room. Audio codecs that provide high quality for speech and audio content alike, as well as enhanced robustness, allow service providers and hardware manufacturers to make this "same-room feeling" a reality. Tailored to specific use cases (e.g., phone calls on mobile devices or wireless handsets or accessories, or IP-based teleconferencing applications on a computer or cell phone), such codecs make video conferences more productive, and for instance, let team members sing "Happy Birthday" to colleagues in a crystal clear manner, even during times of social distancing.

### About the Author

**Manfred Lutzky** heads the Audio for Communications department at Fraunhofer IIS. He received his Dipl.-Ing degree in electrical engineering from Friedrich Alexander University Erlangen in 1997. Afterward, he joined Fraunhofer IIS, where he was initially responsible for efficient implementations of MP3 and AAC multimedia codecs. Today, he is responsible for the development of state-of-the-art speech and audio communication codecs for mobile networks, such as AAC-ELD, LC3/LC3plus, EVS and IVAS, as well as Voice Quality Enhancement technology for voice-controlled smart devices. He is also participating in ETSI NG-DECT and GSMA NG standardization.

# Getting Started with Automotive Audio Bus (Part 1)

## It's Not Just for Autos



By
**Brewster LaMacchia**
(United States)

Part 1 of this four-part article series discusses what Automotive Audio Bus ($A^2B$) is, its features, and its use with $I^2S$ and PDM interfaces. In Part 2, we'll start a dive into the technical underpinnings of the $A^2B$ bus design. In Part 3, we'll complete that dive and look at the Analog Devices, Inc. (ADI) parts that are used for $A^2B$. And last, in Part 4, we will look at creating systems and how to use the tools available from not only ADI but also a number of third-party suppliers. The wide range of tools and off-the-shelf hardware vastly simplify prototyping, development, and debugging of complex audio systems.

Several years ago I was lucky to be in a private suite at a trade show where Analog Devices, Inc. (ADI) was showing off its new IC to send up to 32 bidirectional digital audio signals, along with control, over a simple twisted pair of cable to and from multiple boards.

Having a year prior tried to solve a customer application for a consumer product that could send half a dozen audio channels plus control in just one direction between two devices over a low-cost cable and a connector, the benefit of a low-cost chip that could handle more channels and up to 11 nodes to distribute the audio to/from had obvious value. I wanted it!

It was quickly pointed out that these new wonder chips were only for automotive use and no amount of histrionics would get me access.

Fast forward a few years to the fall of 2019. ADI announces that the Automotive Audio Bus (or $A^2B$, which are a registered trademark of Analog Devices, Inc.) parts and tools would become regularly available through distribution. The audio business had changed in the intervening years, but there are still plenty of applications where $A^2B$ solves problems that don't have good alternatives. The name $A^2B$ still reflects the automotive roots of the product, but I like to think of it meaning "send audio from A to B, and C, D, E, etc."

### Digital Audio Interfaces for ICs

Analog-to-digital converters (ADCs) and digital-to-analog converters (DACs) for audio applications all have $I^2S$ interfaces, which like the well-known $I^2C$ specification was created by Philips [1]. In its basic form, $I^2S$ sends pulse-code modulation (PCM) data and clock over three wires: Data, bit clock (used to clock the data), and word clock (indicates the start of a new PCM packet, also called frame sync). Many devices also need a master clock signal that is a multiple of the word clock frequency, typically 128× or 256×. The data field for each PCM sample is typically 16 bits or 24 bits in length, but most devices use a 32-bit field size; the data is aligned MSB first so unused bits don't affect the value.

**Figure 1** illustrates the basic timing relationship for $I^2S$ signals. Normally, the sending end places data out on the falling edge of the clock (BCLK) and the receiving end clocks it in on the rising edge. The frame sync (FS) is shifted one bit position from where you might expect it to be at the start. Most parts have the ability to control which clock edges are used, whether the extra FS bit shift is used, and if the frame sync is a pulse or 50% duty cycle, and polarity of signals.

For typical 48 kHz sampled stereo audio data used in consumer devices, the frame sync frequency Fs is 48 kHz and the bit clock is 64 times that (two channels at 32 bits) or 3.072 MHz. A master clock of 256 x Fs would be 12.288 MHz. A 96 kHz sample rate would double those numbers.

The original $I^2S$ specification was for two channels, applications needing more channels needed multiple data lines. Over time the $I^2S$ concept was expanded to TDM $I^2S$, where 4, 8, or more channels of PCM audio data are sent in a frame [2]. In the 1980s and 1990s, as was the case with $I^2C$, other semiconductor companies developed $I^2S$ equivalent systems due to Philips' licensing policy. $I^2S$ is now generally used for audio data converters—there are other standards for high-speed RF and industrial converters.

AC97 was a popular (used in practically all computers in the late 1990s and early 2000s) TDM approach released by Intel [3]. It combined multiple data channels and control/status over the interface. There were similar schemes from ADC/DAC manufacturers that pre-dated AC97, and AC97 itself has been replaced by newer computer-centric audio interface standards.

All of these systems for digital audio were intended for use only on a single PCB. Sending the signals over even short pieces of ribbon cable could be problematic for both signal integrity and electromagnetic interference (EMI) reasons. The odd harmonics of the clocks radiate well at the lengths of trace and cable typically encountered in bigger systems. Modulation of clock edges by the data or other signals created timing jitter that early ADC and DACs were susceptible to, degrading their performance.

### PDM Interface

With the advent of voice input, it is common for products to have several microphones. In automotive cabins where voice input and Acoustic Noise Cancellation (ANC) is needed, there can be a dozen or more microphones. Traditional analog interfaces would have a number of cost and performance downsides.

For digital microphones, Pulse Density Modulation (PDM) has become a common interface format as it is easy to implement in a microelectromechanical system (MEMS) microphone design. The $A^2B$ transceiver parts include the ability for $I^2S$ or PDM input. In the case of PDM, four microphones can be directly attached and the $A^2B$ transceiver converts the PDM to PCM.

PDM requires only two wires, a clock input (usually 64 Fs), and a single data line shared by two microphones. Double Data Rate clocking is used (i.e., one mic uses the rising edge/high period and the other uses the falling edge/low period). (For more information about PDM and MEMS microphones see the Resources section.)

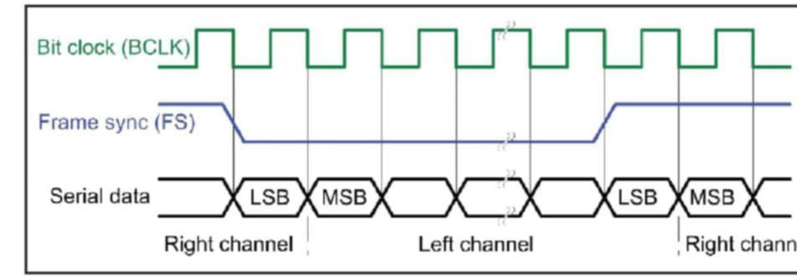



Figure 1: Basic $I^2S$ signal relationships. Multichannel TDM will have more than left and right data words. The frame sync may be a pulse or 50% duty cycle. Some systems also need a MCLK signal. (Source: wikimedia.org)

### Why $A^2B$?

The transportation industry is driven by cost, both for building product and operational cost in terms of fuel efficiency and reliability. Electronics use in vehicles continues to rise and a traditional approach of adding more wires for each new system would be a failure, as illustrated in **Figure 2**. Over the years a number of automotive buses have been created, Controller Area Network (CAN) and Media Oriented System Transport (MOST) being perhaps the better known of the multitude of choices. None of the available offerings have the characteristics needed for in-cabin audio, particularly for hands-free microphone and

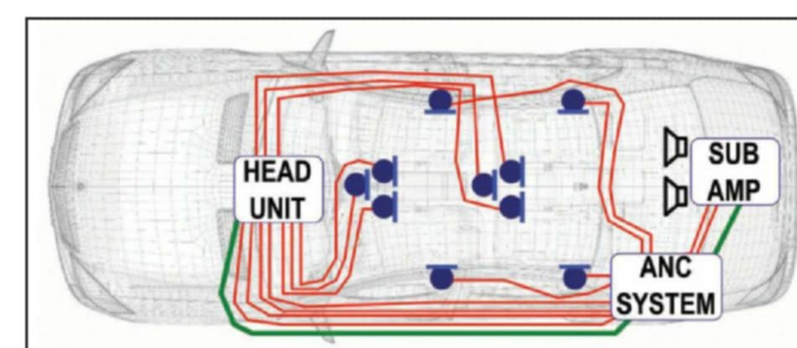



Figure 2: A traditional approach to connecting microphones, speakers, (red shielded analog wires) and control systems (green bundle) together creates a large, heavy, and expensive cable harness in a vehicle. (Source: Analog Devices, Inc.)

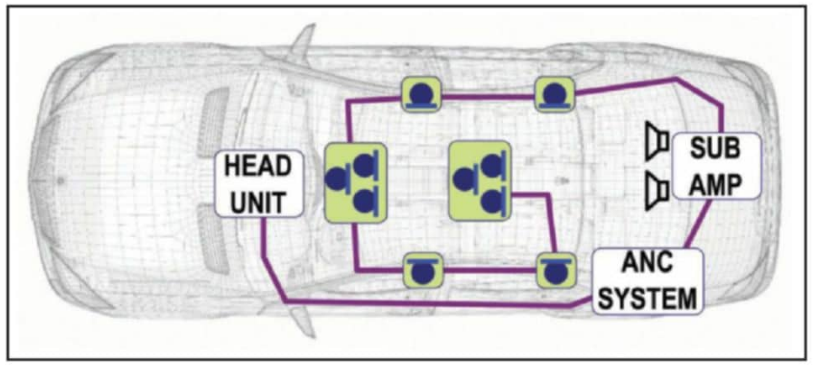



Figure 3: A simple two-wire interconnect carrying digital audio and control makes for a simple installation, more design flexibility, and overall improved energy efficiency and cost savings. (Source: Analog Devices, Inc.)

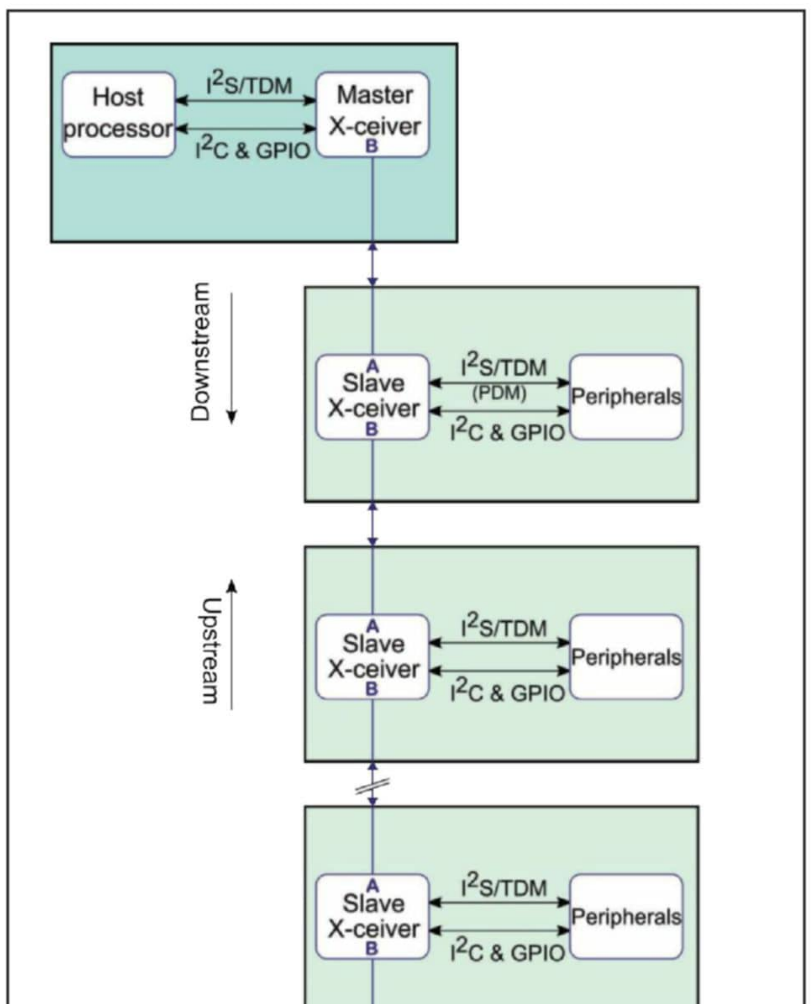



Figure 4: A²B systems consist of one master node and up to 10 slave nodes connected in a daisy chain. The A ports face upstream, B ports face downstream. The master node does not use its upstream A port. Slave nodes might contain ADC, DAC, or (PDM) microphones; or the could just be indicators and sensors that don't involve audio and I²S.

road noise cancelation. Adding more speakers for better sound field control also adds to the wiring complexity.

It would be much simpler if, instead of home-running wires from each audio device, we multiplex them on to a single wire pair. The system can then look more like the diagram shown in **Figure 3**.

We would like the system to have multiple drops, and for low power things such as microphones, phantom power would eliminate the need to run power cables to devices. Data will need to travel in both directions over the cable. The ability to interface with device nodes that have peripherals with I²C or GPIO control means we can eliminate local microprocessors. To be useful for active noise control, very short latency—a few audio samples at most—is needed.

As might be expected from the article title, A²B can do all of that and more. A²B does not replace the other types of network protocols in use in transport, industry, or consumer applications. Instead it complements them by removing the need to transport a lot of low-latency data that those other choices were not designed to support.

AES-67 or Dante-based Ethernet networks are great for large amounts of audio between multiple boxes over long distances. MIPI Soundwire, in the other end of the data transport application space as a "talk to the chip next door" method, to-date has no announced standard interface parts for audio. FPD-Link is point to point and unidirectional data, but can go over cables.

A²B can go 15 meters between nodes and up to 40 meters total length. In many applications, the entire A²B network of devices can be loaded up from the host processor over I²C at startup and no further interaction from the host is needed. It acts like a hardware "virtual I²S, I²C, and GPIO over distance" system. There's no software license or royalties. The per node bill of materials (BOM) parts cost is less than $8 in volume. Ethernet audio solutions can cost two times to three times more per node and have high latency.

## Basic A²B System

**Figure 4** illustrates a basic A²B system. The A²B master node sends data downstream to slave nodes and receives upstream data. Audio data does not need to go through from master to slave or vice-versa, it can also flow directly between slave nodes in either direction. Data going to more than one node only needs to be sent once (i.e., one node can broadcast audio to all nodes or subset of nodes).

A²B moves data in packets, the packet rate is either 44.1 kHz or 48 kHz, which is normally the same as the sample rate. The system can support 2× and 4× sample rates of 96 kHz and 192 kHz. The rest of this article will be based on 48 kHz. Audio data size can be 16 bits or 24 bits. Since data is always sent at full level (i.e., end output volume would normally be controlled locally at a node), 16 bits will exceed the available dynamic range of the source and output in many cases.

The A²B bus time division multiplexes data streams into a downstream packet and an upstream packet. Starting at the master node the downstream packet is sent through each slave node—each slave mode copies off the data it needs and/or adds new data destined for a downstream slave node. When the master node's packet hits the last node in the system, a new upstream packet is created, and each slave node adds its data and/or reads any data it needs locally. Once the Upstream packet is received at the master, the cycle repeats at the next sample start. If not all data packet slots are used, the bus is idled until the start of the next frame (sample rate) period, reducing power consumption.

For any given node, there is a maximum of 32 streams in the downstream direction and another 32 streams in upstream direction. There is an aggregate limit at each node for the total number of streams, independent of direction. For 16-bit data this is 51 streams. For 24-bit data, this limit is 34 streams.

This per node limit is not the same as the total number of unique audio streams that can exist in the entire A²B system. For example, data sent between two adjacent nodes doesn't add to the stream count of other nodes that are up or downstream from the two nodes exchanging data.

In an A²B system with all of the processing in the master node, all data would route to and from the master node, and the 34 stream total applies. Placing the processing in the middle of the network could effectively double the bandwidth across the entire A²B network, though a master node is still needed to manage the set up. Luckily, the ADI design software Sigma Studio makes it straightforward to create these complex scenarios and off-the-shelf evaluation hardware can be used to validate the needed routing for when the routing can be defined as a set of predefined static routes. Dynamically changing routing is a more complex undertaking, and it's worth mentioning that the system will pause briefly (the time to write out new routing tables to each node, less than 500 msec) under that specific scenario.

Not shown in detail is the I²C routing between the master node and peripheral nodes. From the master node's host processor perspective all I²C peripherals on any node are reachable through an I²C address muxing system. The A²B transceiver parts also have GPIO and the system supports a virtualized GPIO. By using the broadcast capabilities of the A²B bus GPIO across multiple nodes can be updated synchronously. For nodes with local processing, there are also mailbox registers that the local processor can read via I²C on the local node.

## A²B Latency

One of the key features of A²B is the fixed latency for audio data regardless of which node it originates on and how many nodes it is routed to. While we described A²B as packet-based, the contents

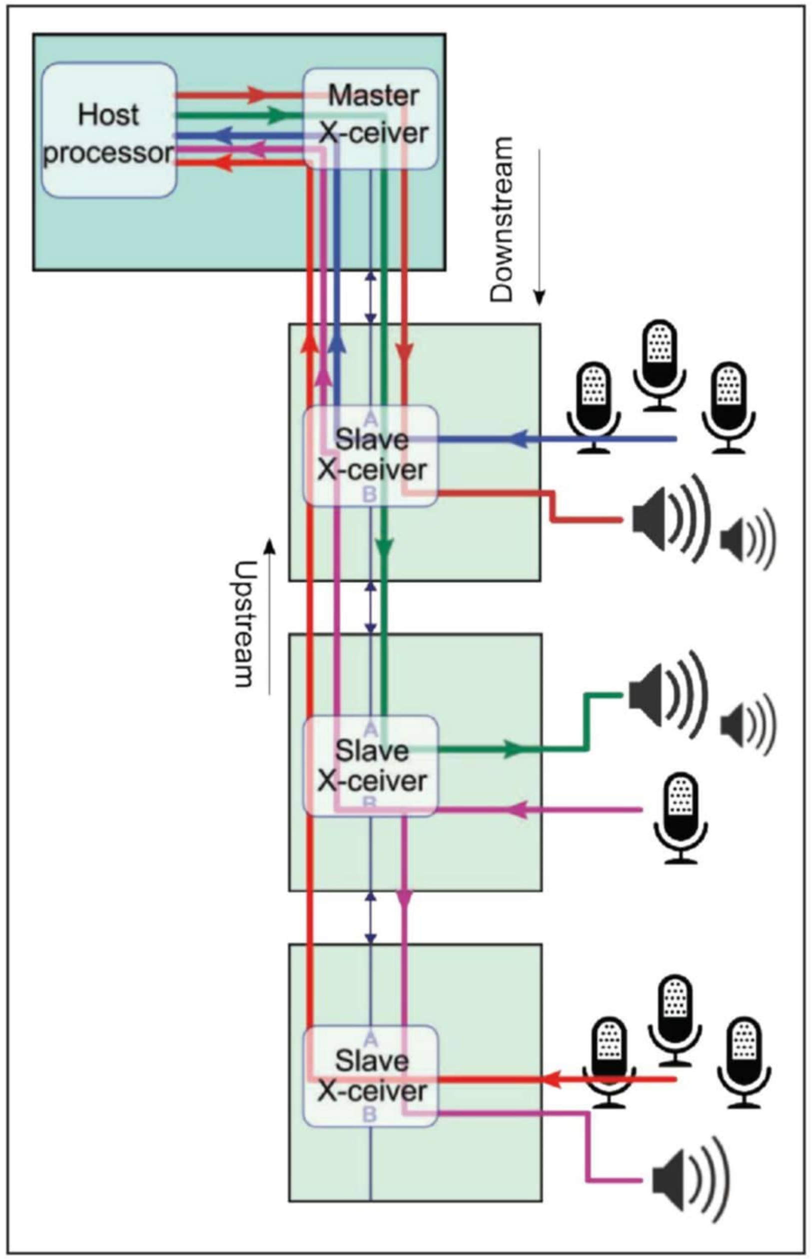



Figure 5: A somewhat fictitious example of an application with multiple audio sources and outputs shows how data can move between nodes. All data is moved in a little over two sample periods.

## About the Author

**Brewster LaMacchia** started in signal processing working on military satellite communications systems. Since then he has worked on a wide range of signal processing systems, including a number of years on video processing and surround sound audio for consumer home theaters. Recently, he started Clockworks Signal Processing to develop open-source hardware for multichannel signal processing. He also dedicates time to public astronomy events to expose people to science and critical thinking. In the past 10 years, he's helped more than 7,000 people look through a telescope for the first time.

of the packet are not stored and forwarded at each node. Each node starts outputting its incoming datastream within a few A²B bit periods (one A²B bit period is about 20 nsec).

Though not entirely correct, as empty slots are not sent and data packets in a given direction are contiguous, we can think of a simplified system with 16 downstream and 16 upstream slots (see **Figure 5**). Combined we call this the superframe, which also includes I²C data, GPIO, and other status and control bits. The master node starts by sending a packet of 16 channels (plus control) downstream. The first node grabs the data it's been programmed to output while at the same time passing the bit stream on to the next node. This process repeats until the last node is reached.

If a particular node is the last one that needs a given audio data stream than that stream is not passed on. If the master isn't sending (in this example) all 16 possible streams then an intermediate node can send data downstream to another node in one of the unused stream slots.

The last node, after receiving the last downstream packet, repeats the same process in the upstream direction. Intermediate nodes can read streams, stop passing ones no longer needed upstream, and add their own data destined for upstream nodes.

The entire data movement completes in one audio sample period. There's also a sample period to clock the data in via I²S before it can be transmitted, meaning that output of I²S data can start two sample periods (nominal, there's a small additional fixed delay) from when it was received. At 48 kHz sample rate, this totals to about 50 µsec.

The data exchange is illustrated in more detail in **Figure 6**, showing data being sent both downstream and upstream and how from input to output is just a little over two audio sample periods.

The superframe period (20.83 µsec, corresponding to 48 kHz) is divided in to 1024 bit times, for a data rate of 49.152 million bits per second (Mbps). For the remainder of the article sections, we'll round this off to 50 Mbps.

In addition to audio data, there are headers with addressing information, CRC and parity bits to detect errors, I²C and GPIO data. Corrupted audio samples are handled by the same style of algorithms used in CD players. Even an outrageously high 1% loss is barely detectable. A²B transceivers support built in loopback and Bit Error Rate (BER) testing. It has been the author's and others experience that bit errors just don't

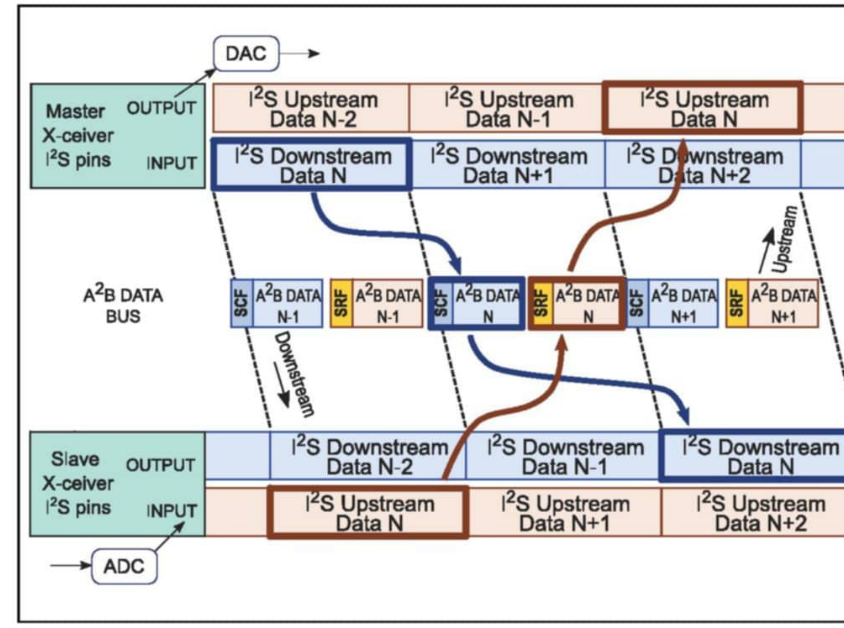



Figure 6: A²B moves data in packets at the sample rate. Illustrated here is the relationship between the I²S pins on two nodes sending data upstream and downstream. This figure applies regardless of where the nodes are in physical relationship to each other. (Source: Analog Devices, Inc.)

happen in normal use, but it's reassuring to know that if your car gets hit by lightning, the music will still sound great.

## Summary

In Part 1, we have looked at the properties for a locally networked audio transport that has very low and deterministic latency. Basic digital audio interfaces (I²S and PDM) were reviewed.

The next part of this article will look in more detail at what's happening on the wire pair that A²B uses to transport data and power around a system. Features designed for automotive use: excellent EMI/EMC characteristics, robustness to faults, and diagnostics, can help in designing systems that operate reliably and minimize downtime.

## References

[1] Philips Semiconductors, I²S bus specification, February 1986 and revised June 1996.

[2] Cirrus Logic AN301, Revision 1, "Time Division Multiplexed Audio Interface: A Tutorial," September 2006.

[3] Intel, Audio Codec 1997, Revision 2.3. April, 2002.

## Resources

Analog Devices, Inc., A2B information, www.analog.com/en/applications/technology/a2b-audio-bus.html

"PDM and MEMS microphones," Analog Devices, Inc. Technical Article MS-2472, www.analog.com.

# High-End Class-D Audio Board for the Arduino MKR Family

By
**Jens Tybo Jensen**
(Infineon)

Arduino boards are extremely popular among DIY enthusiasts and a number of add-ons or "shields" exist, particularly for audio players. Now, the Arduino MKR family has expanded the possibilities even more. This article details an Infineon amplifier board featuring integrated MERUS™ MA12070P proprietary multi-level Class-D ICs, which create an exciting platform for advanced wireless audio projects—including full-fledged Alexa-powered DIY smart speakers.

Photo 1: This image shows a board overview, including the Arduino MKRZERO.

The Arduino project, started in 2005 as a student program at the Interaction Design Institute Ivrea in Italy, aims to provide an affordable and easy way to create devices that interact with their environment while using advanced microprocessors, controllers, sensors, and transducers.

Since 2005, Arduino boards have been steadily growing in popularity among makers and DIY enthusiasts, who have developed a broad variety of projects—ranging from gaming consoles and drone controls to various smart-home automation projects.

Arduino boards are based on the open-source hard- and software, licensed under the GNU Lesser General Public License (LGPL) or the GNU General Public License (GPL), enabling board manufacturing and software distribution by anyone. This extends the development of high-performance embedded computing power to a much wider audience—including people without a professional hardware or software background.

When looking at the different types of DIY projects, it is interesting to see that many revolve around audio applications and connectivity (e.g., IoT audio interaction or stand-alone audio players). However, until recently, it was necessary to add several separate boards or "shields" together to realize such a system. This has changed since the release of the MKR family, where Arduino has now added audio features such as an I²S bus, SD card audio read/write, and Alexa voice-control compatibility. This also makes the Arduino platform a perfect demonstration vehicle for Infineon's high-performance integrated MERUS™ multilevel Class-D audio ICs.

By releasing a fully compatible and power-efficient amplifier board with enough output power to drive a variety of speakers—all with best-in-class audio performance, low power consumption and high efficiency—Infineon is providing yet another useful building block to the Arduino ecosystem.

Key parameters and features of the new Infineon Arduino board KIT_ARDMKR_AMP_40W (compatible with Arduino MKRZERO and MKR1000 Wi-Fi) include an integrated MERUS MA12070P proprietary multilevel Class-D audio amplifier; 5 V/2.5 A power input, sourced from the same single USB-C power supply or battery pack, and delivering up to 40 W instantaneous peak output power with those supplies. No need for external or extra power supplies.

With this board, Arduino fans around the world are now able to build their own high-end wireless audio players, with full hardware control, customization, and error monitoring possible through the Arduino programming framework. And thanks to the recent addition of Arduino's Amazon Alexa skill, it is even possible to connect Alexa to Arduino IoT cloud projects, without the need for additional coding.

## Cool Technology Optimized for Music Playback

The KIT_ARDMKR_AMP_40W is an audio amplifier board, compatible with the MKRZERO board from Arduino, and built around Infineon's MERUS MA12070P multilevel amplifier. **Photo 1** depicts, how Arduino's MKRZERO fits on top of the KIT_ARDMKR_AMP_40W, which has been designed to offer excellent audio quality and easy compatibility with the Arduino MKRZERO boards. More specifically, the boards cater to applications such as portable speakers, smart speakers, DIY hi-fi speakers, or any kind of consumer-oriented audio prototyping.

Infineon's MERUS MA12070P multilevel amplifier is optimized to effectively reproduce the dynamics of music signals. While operating extremely efficiently at low to moderate output power levels, it can deliver very high peak output power levels during highly dynamic music passages—all in a small form factor with minimum power loss and no need for bulky heatsinks.

Just by touching the amplifier during operation, its efficiency is noticeable. Even when it is playing at high volumes, the IC stays completely cool as almost all of the energy delivered from the power supply is passed on to the speaker.

**Figure 1** provides a more detailed system overview. The system's power section is designed to alleviate common challenges associated with the audio design (i.e., providing enough continuous power in combination with peak output power and sufficient voltage headroom to allow undistorted music playback).

To this end, the board incorporates a boost converter that boosts the voltage from a simple USB-C input power adapter or a battery from 5 V to 20 V. The boosted 20 V is used as the power supply for the power amplifier section, while the bulk capacitance stores peak energy for the short-burst peak power delivery.

MERUS MA12070P is by default configured to drive two speakers (BTL), but can also be configured as one single, more powerful channel (PBTL). The power section with its amplifier can deliver up to 20 W instantaneous peak power per channel (40 W total). Nominal speaker impedance can typically vary from 4 Ω to 8 Ω. Under specific conditions, lower or higher impedances are possible too. The used amplifier IC has a digital front end, including volume control and limiter functionality. On top of that, the amplifier configuration has (status) registers that can be accessed by using the I²C communication protocol.

## System Overview

The main processor unit in the system is the Arduino MKRZERO (or a similar board from the same family). This microcontroller board can stream I²S audio while at the same time taking care of housekeeping functions, such as amplifier register configuration over I²C and interrupt-based status monitoring. Besides that, by using the ADC and GPIOs, some user-interface functions (e.g., volume control, generic push-buttons, and LEDs) have been implemented.

## Debunking the Output-Power Myth

A common misunderstanding in the world of audio engineering, especially when it comes to specifying amplifiers, is that they require a lot of output power (i.e., that at least hundreds of watts and preferably more is necessary to have an immersive and engaging hi-fi sound experience). Although this may be true for some applications, it is definitely not true for most applications in the consumer audio space. Unfortunately, many audiophiles (and even a few professional audio engineers and marketers) wrongly equate output power to loudness. However, loudness is not measured in watts.

Quoting a certain output power without further information is a bit like saying that a specific town you want to visit is an hour away. Unless you also provide information about under which conditions (i.e., if you mean one hour of walking, riding a bicycle, or by a car while observing the speed limits), this information is useless. Although there is a correlation between the amplifier output power and loudness, the actual speaker(s) in the audio system play(s) a far more important role when it comes to loudness.

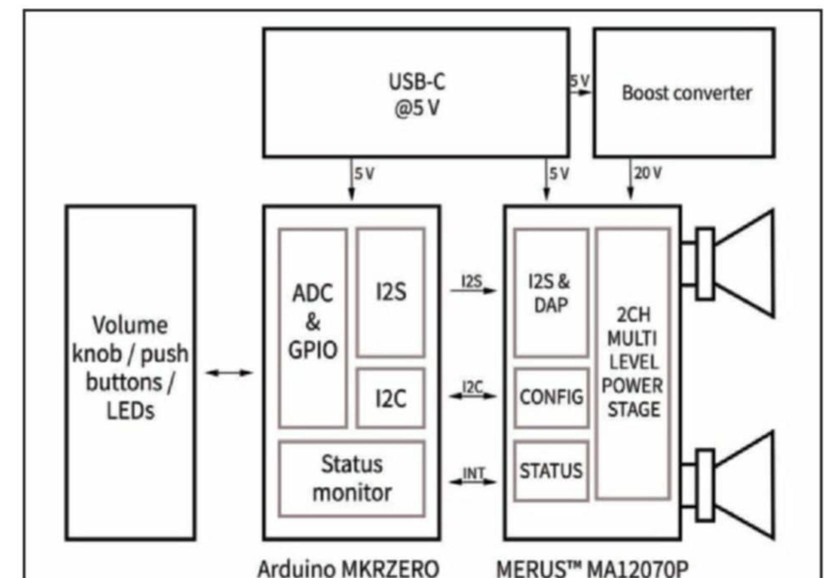



Figure 1: Here is a more detailed system overview.

A very important parameter, therefore, is the speaker's sensitivity, which determines the amount of power that is necessary to drive a certain loudspeaker to a certain loudness. More specifically, it measures the sound pressure level (SPL), typically expressed in decibels (dB), which will be derived from the speaker with 1 W of power input (or 2.83 V sinus input across an 8 Ω speaker), measured at a 1-meter distance.

Bookshelf speakers typically have sensitivity values of 85 dB to 89 dB, whereas floor-standing models typically range from 87 dB to 92 dB. High-efficiency speakers are in the 93 dB to 100+ dB range and pro equipment (e.g., horn speakers) can have sensitivities as high as 111 dB.

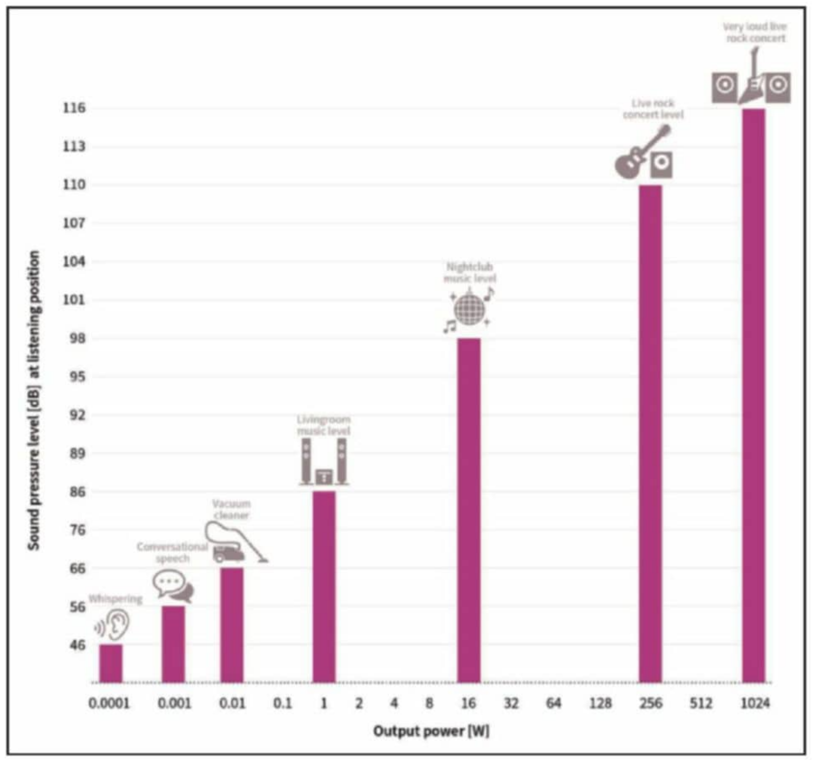



Figure 2: Typical sound pressure levels are calculated as $86 + 10 \times \log^{10}$ (output power).

## About the Author

After the Infineon acquisition of Class-D Amplifier IC startup Merus Audio, where **Jens Tybo Jensen** was serving as VP Sales and Marketing, Jensen became head of Marketing & Application Engineering for all Class-D Audio Products within Infineon Technologies. Prior to his tenure with Merus Audio, Jensen held a European Sales Director position with Knowles Electronics—a global supplier of high-performance micro-acoustic components and intelligent audio IC solutions. His global business development experience stems from a variety of start-ups and international hi-tech companies. Jensen was co-founder and VP Sales at Nangate, Inc., which was acquired by Silvaco, Inc. He also served as VP Sales & Marketing for Exbit Technology A/S, which is now a part of Microsemi Corp. Jensen holds a B.Sc.EE degree from the Technical University of Denmark and an Executive MBA from Copenhagen Business School.

Also, the number of speakers is an important parameter. When using two speakers instead of one at the same distance and with the same sensitivity, 3 dB of SPL are added at the listening position. Similarly, the placement of the speakers also plays a key role. So-called "room gain" means that the sound reflected from the walls can reinforce the sound that is coming directly from the speakers. Placing the speaker near a wall adds around 3 dB SPL at the listening position. This effect is most prominent at low (bass) frequencies and explains why subwoofers often sound louder when placed near a wall, or ideally in a corner.

The distance from a listener to the speaker (also known as dispersion) plays a critical role in the loudness at the listener's position. Therefore, please consider the data in **Figure 2**, showing how much loudness can be produced based on a single speaker with a modest sensitivity of 86 dB (for two speakers simply add 3 dB and another 3 dB if the speakers are placed near a wall).

Note that even very low output-power levels (from 0.0001 W to 0.1 W) will produce loudness levels comparable to that of speech or a vacuum cleaner and that the typical average output power levels for most standard-sized living-room music playback situations are not more than a couple of watts.

For every 3 dB increase of loudness, a duplication of the amplifier's output power is needed. The output power number starts to rapidly escalate when loudness levels equivalent to nightclubs and live rock concerts are reached. Those, however, are not the target applications for the Arduino board.

## How Loud Is Loud Enough?

How much loudness do people actually need? How much will they tolerate inside the living room when listening to music? Does this Arduino board fit the bill?

With the Arduino board, approximately 20 W instantaneous peak and maximum 5 W continuous (RMS) power per channel can be reached. Given two free-standing speakers with a modest 86 dB sensitivity, this corresponds to 102 dB peak and 96 dB average SPL, respectively, at a 1-meter distance. In a typical living room with a 3-meter distance to speakers near a wall, the maximum SPL values of 95.5 dB peak and 89.5 dB average, respectively, can be reached.

## Is This Loud Enough?

As a reference, a few years back *Stereophile* magazine did a study and discovered that the average listening level in most of the rooms of the audiophiles that they surveyed was around 83 dB SPL. As it turns out, above this level people are not just listening to the sound from the speakers but increasingly to the sound of the room (meaning that reflections and reverberations from the wall floor and ceiling dramatically change the frequency response of the music being played). Hence above this level (potential longer-term hearing loss aside), it is advisable to consider some form of room treatment to make sure that the room is matched to the listening volume level.

According to Audio Engineering Society (AES) Fellow Floyd E. Toole [1], professional installers of high-end home cinema sound systems report that their customers typically play movies at -10 dB or lower. That is a power reduction of a factor of 10, which is a significant cost factor when equipping a theater but also a considerable price difference for home audio equipment. As seen in the previous example, even a reduction of 3 dB is a factor of two in power. Hence, the obsession with output power may not only be misleading, it can also unnecessarily empty a wallet.

The best way to find out how much SPL is enough is to take a phone and download a sound pressure app. There are plenty of smartphone apps on the market, which allow for taking measurements on the pressure in a room. By doing some readings while playing the music in a typical daily situation, the average listening level can be discovered and then compared to the above information. It is surprising, how low a normal SPL listening level is, and even more surprising is the calculated corresponding power requirement based on the speaker's sensitivity.

If the Arduino amplifier is powerful enough for the targeted application, this compact multilevel Class-D audio amplifier design example should bring awesome sound qualities to the audio listening experience.

## What Is a Conventional Class-D Amplifier?

A Class-D amplifier, also called a switching amplifier, is an electronic amplifier in which the transistors operate as electronic switches and not as linear gain devices as in other amplifiers (e.g., Class A or Class A/B). Instead, they operate by rapidly switching the output back and forth between the supply rails, being fed by a modulator using pulse width, pulse density, or related techniques to encode the audio input into a pulse train. The train of pulses is subsequently applied to the loudspeaker—often through a low-pass filter to block high-frequency components of the transients. Since the pairs of output transistors in the output stage are never conducting at the same time, there is no other path for current to flow apart from the low-pass filter and the loudspeaker. For this reason, the efficiency of a Class-D amplifier can exceed 90% at full output meaning that only 10% of the applied energy from the power supply is lost as heat and 90% is used to move the speaker coil and produce sound.

## References

[1] F. E. Toole, *Sound Reproduction: The Acoustics and Pyschoacoustics of Loudspeakers and Rooms*, Third Edition, Routledge; July 2017, pp. 48.

## Resources

"Design of an Arduino Compatible Audio Amplifier Platform," Application note for KIT_ARDMKR_AMP_40W, Infineon, www.infineon.com/dgdl/Infineon-Evaluation_kit_KIT_ARDMKR_AMP_40W-ApplicationNotes-v01_00-EN.pdf

## What Is a Multilevel Class-D Amplifier?

In a conventional Class-D amplifier, there is a lot of switching activity—even in an idle and near-idle mode, meaning when there is a silent passage in the music. Each transistor pair in the (half-bridge) output stage changes the duty cycle to create the desired output level. However, even if no output and no sound is coming from the speakers, there is a lot of internal switching activity going on, which entails switching loss. Hence, conventional Class-D amplifiers are not very efficient at idle power, which dominates the average power level for music signals and has many passages of relatively low signals as well as many signal peaks.

This is visible in **Figure 1** as the shaded area in the curve. In contrast, due to proprietary and patented circuit architecture, the multilevel amplifier's eight smaller MOSFETs can scale the output to a much higher granularity in terms of resulting switching frequency and output levels. Essentially, this dramatically reduces power losses, provides much higher output signal granularity, and yields many other additional benefits such as less out-of-band noise, higher tolerance for power supply ripples, less need for bulky output filtering components, and so forth.

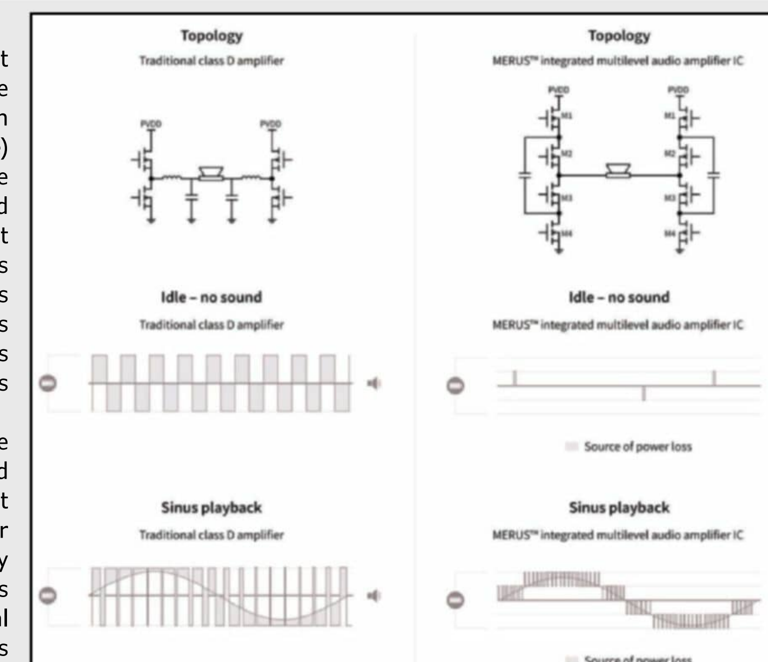



Figure 1: The speaker topology and of a traditional Class-D amplifier is compared to the topology of a MERUS Integrated multilevel audio amplifier IC.

# Coffee & Bluetooth Speakers

## Desktop Speaker System Using Recycled Coffee Cans

By
**Ken Bird**
(United States)

The completed system is shown connected via Bluetooth to a tablet.

Our esteemed speaker builder proposes a fun project of a 2.1 Bluetooth-enabled speaker system that ticks all boxes for both multimedia and desktop use with a computer, but also as a potential conversation piece for anyone wanting to demonstrate DIY and eco-friendly credentials.

With all the streaming audio and video now available on the Internet, the two decades-old powered speaker system I used with my current computer was in need of an upgrade. The specifications for the new speakers required that they be self-powered—with no external amplifier—have Bluetooth capability for use with a smartphone, good quality audio using a subwoofer, and include analog audio inputs for a CD or other source. The enclosures had to be compact, but not too small to affect audio response, and they had to be easily assembled. This project met most of the criteria—some audio enthusiasts might label it a bit whimsical, but it was fun to build and the components are not too expensive.


Photo 1: I selected these coffee cans for the system. The large can will house the woofer and amplifier, while the small cans will become the satellite speakers.

The system consists of three enclosures, two satellites, and a subwoofer enclosure that also houses the electronics. The enclosures are made from ordinary plastic coffee cans available from your or your neighbor's local trash receptacle. (Mine came from our local church's coffee hour empties.) I made the two satellites from 1.6 lb. size of cans of Maxwell House coffee and the subwoofer from the slightly larger 3.6 lb. coffee can made by Folgers. I have not been able to find the larger size of Maxwell House in my area. It is possible to use the Folgers 1.6 lb. cans as the satellites, but they would have to be used as omni-directional speakers unless you make some stands for them. The Maxwell House cans have the advantage of being used as omni speakers or as directional speakers (see **Photo 1**).

Photo 2: The main electronic parts are displayed with the woofer, two satellites speakers, the power supply, and the three-channel Bluetooth amplifier.

Photo 3: I used a hand reamer to enlarge the mounting holes for the parts, which I drilled first with a 1/8″ bit.

### The Amplifier, the Drivers & the Power Supply

The amplifier mounted in the subwoofer can is a multi-channel Class-D type from Parts Express (part number 320-644). This is a Class-D Amplifier board with Bluetooth 4.1 and a 2× 15 W and a 30 W subwoofer channel. It features three volume controls, a master control, a satellite, and a subwoofer volume. It will operate on an external DC power supply from 8 V to 20 V.

The satellite drivers that I chose are the Fountek FE 85 3″ full-range models, also from Parts Express (part number 296.717). They are rated at 25 W with a 100 kHz to 25 kHz response. They are run full range with no crossover components required. The subwoofer is the Dayton model ND105-8 4″ speaker (Parts Express part number 290-214). The power supply I used was a 15 V unit also from Parts Express (part number 120-057), which is shown in **Photo 2**.

### Construction

Once you have selected your cans, you can elect to paint them or leave them as is for a more eclectic-looking system. I found, even if primed with auto primer, the plastic does not take the


Photo 4: I used this technique to slightly flatten the bottom portion of the woofer/amplifier housing (see text).


Photo 5: The handled portion of the woofer/amplifier can shows, left to right, the right satellite output jack, the 1″ port, the 2.5 mm power jack, the RCA audio in jacks, and the left satellite jack.

Photo 6: The lower rear handle of the satellite can is where the binding posts are located.

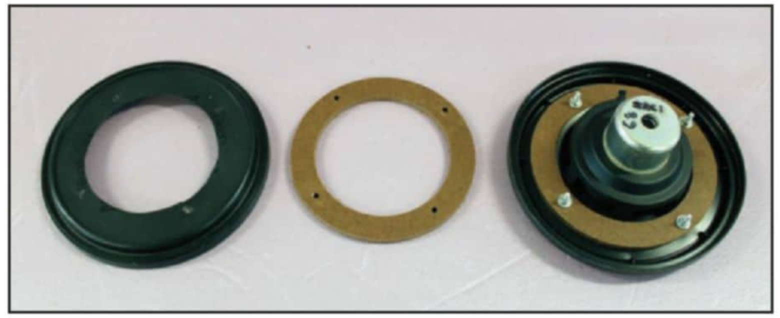


Photo 7: This image shows the modified can lid on the left with the baffle ring in the center. The completed baffle is shown on the far right. The ring is 5.5″ in diameter; the hole for the woofer is 3.75″, and for the satellites, 2.75″.

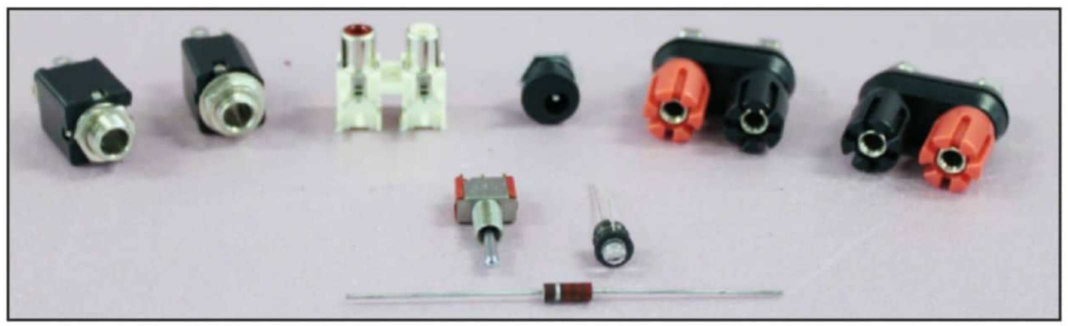


Photo 8: The hardware needed for the project consists left to right 1/4″ speaker out jacks, the RCA audio in jacks, the 2.5 mm power jack, the satellite binding posts, the power switch, the LED, and a 2.2 kΩ resistor.

paint well and it can be easily scratched off when handling. I built two of these and left the second one in its original finish. Before painting, you will need to cut the hardware mounting holes into the lids and sides of the plastic, and make a slight modification to the front of the woofer to accept the amplifier board. I drilled and sized the holes using a hand reamer (see **Photo 3**).

## Subwoofer & Satellite Modifications

I used the large can (the opposite side from the molded handles) to mount the amplifier, the on-off switch, and a LED. To do so, the can must be flattened out a bit. I did this by carefully using a heat gun to soften the plastic and applying slight pressure to the area until it softened. You could also use a hair dryer for the heat supply. Then, I applied a small hammer to the panel until the plastic cooled (see **Photo 4**).

Measure up 2″ from the bottom of the can and drill three 3/16″ holes spaced 3/4″ apart. Then, measure up 2″ and space two 1/4″ holes 1″ above the other to mount the switch and the LED. The plastic is very easy to work with and I made most of the holes with a hand reamer once I drilled a 1/8″ starter hole. I used a hobby knife to remove the rough edges of the openings. I also cut a 1″ port cut into the back, this was added after initial testing revealed that it improved the bass response and helps keep the audio amplifier cooler.

Next, place the rear of the subwoofer between the handle molding, and leave room for the line level audio inputs, power jack, and the speaker outputs jacks centered about 1.5″ from the bottom. Placement of these is approximate and the builder may elect to place them at different locations (see **Photo 5**).

I used two 1/4″ phone jacks for the speaker's output and one 2.5 mm coaxial power jack, and two RCA audio jacks for the line level inputs. The satellite speakers use a pair of binding posts mounted at the base of the handle. Drill two 1/8″ holes spaced 3/4″ apart, located 1″ from the bottom (see **Photo 6**).

### Parts List

| Amount | Part | Source |
|---|---|---|
| 2 each | Maxwell Coffee cans, 1.6 lb | |
| 1 each | Folgers Coffee Can 3.9 lb | |
| 1 | Three-channel amplifier | Parts Express (Part# 320-644) |
| 2 | 3″ speakers Fountek FE 85 full-range 8 Ω | Parts Express (Part# 296-717) |
| 1 each | 4″ woofer | Parts Express (Part# ND105-4, 4 Ω) |
| 1 each | SPDT on-off-on switch | All Electronics (Part# MTS-5) |
| 1 | 12 V 5 mm white LED | All Electronics (Part# 12WP) |
| 1 each | LED snap in mount | All Electronics (Part# HLED-4) |
| 1 each | 2.2 kΩ resistor | |
| 2 | 1/4″ Phone jacks | All Electronics (Part# PHJS-3) |
| 1 | RCA Phonon jack assembly | All Electronics (Part# RCJ-21) |
| 1 | 2.5 mm DC input jack | All Electronics (Part# DCJ-36) |
| 2 each | Dual banana jack assembly | All Electronics (Part# BP-25) |
| 1 | 15 VDC power supply | Part Express (Part# 120-057) |

## Making the Speaker Baffle Plates

I used the lids of the cans as the speaker baffles after I reinforced them with a speaker mounting ring made from 1/4″ material. I chose Masonite for this project, but 1/4″ plywood can also be used. The rings for the satellites are 5.6″ in diameter with the speaker cut-out hole being 2.75″. The ring for the subwoofer in also 5.5″, with a 3.75″ hole. I cut the rings with a router, using a Jasper circle tool. I mounted the speakers through the top of the lid with the ring mounted on the rear of the speaker. Use 1″ 6-32 screws with washers and lock washers. I then placed a piece of 1/4″ × 1/2″ foam weather stripping on the rear of the speaker flange before mounting.

The lid on the Folgers can made an adequate air seal for the subwoofer, while the lids for the Maxwell House cans have less of an air seal. However, this is not as important on the satellites, and there will be little loss of efficiency at the frequencies in the satellite channel. Builders can experiment by sealing the lids with caulk, but we found the sound to be excellent as built (see **Photo 7**).

## Installing the Hardware

All of the connections on the amplifier board, with exception of the audio input, is made with wire compressing screw terminals. Since there is little room to get a soldering iron into the can, I recommend prewiring all the connectors and then installing them into the can. The audio inputs and speaker jacks are attached with external screws and nuts while the power input requires the mounting nut to be threaded inside to mount it. The 12 V LED and switch require some fabrication for 15 V operation with a 1 kΩ resistor in series with the LED and positive power lead (see **Photo 8**).

Once assembled, the speakers are soldered and the baffles are merely snapped on. I added about 3 oz. of polyfill material to each satellite and 4 oz. to the woofer. There was no noticeable heat buildup in the woofer can after an hour of playing at a high volume level. I added a 1″ port to the rear of the woofer can to improve the bass output and help keep air circulating around the amplifier.

## Performance

The system is obviously not a chest thumper, but it produces a room-filling sound with minuscule distortion, and it lends itself to a lot of experimentation with other drivers and placement of the parts. It would be a fun project for youngsters interested in technology as it is simple to build with minimal help from an adult.

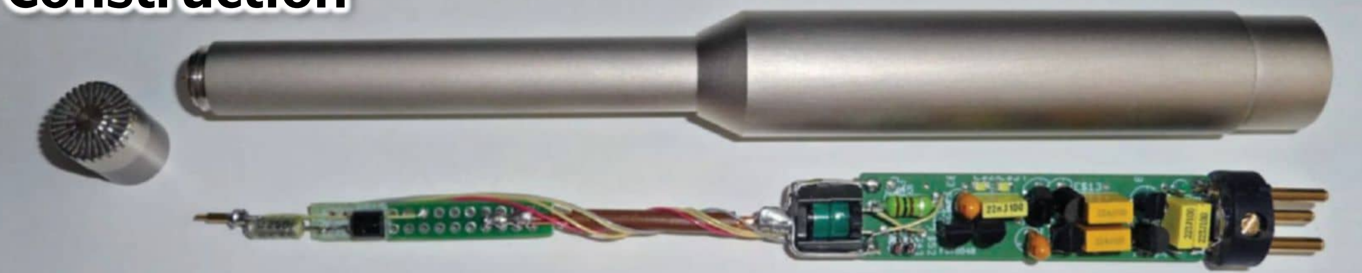
The specifications state the amplifier will operate on 8 VDC to 20 VDC. With an output of 15 W for the satellites and 30 W for the woofer all at 4 Ω. The 8 Ω satellite speakers worked well with the 4 Ω woofer, and produced more than adequate volume using the 15 V supply.


### About the Author

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

# Ultra-Low Distortion Microphone Buffer Amplifier (Part 1)

## Construction



By
**Michel Nieuwenhuizen**
(The Netherlands)

This is the first part of a two-part project detailing the construction of a new, extremely low distortion microphone buffer amplifier.

Some time ago, I discovered a pair of Bruel & Kjaer (B&K) microphone capsules type 4165 [1]. Knowing their excellent quality for both measurements and audio recording, I decided to make them into complete microphones by adding a buffer amplifier and housing. In previous experiments with other microphone capsules, I had used versions of the well-known "Schoeps circuit." This circuit can be found in many cheap and not-so-cheap Chinese microphones, but the fact is that the Schoeps circuit is more than 50 years old. It has its weaknesses, such as the need for large electrolytic capacitors (elcaps) to dampen the noise of the Zener diode in the power supply. So I imagined that it should be possible to design something better.

I started with some experimental circuits, which I then verified with both simulations and measurements. I do not believe in claims that cannot be verified by measurements. There is very little sense in applying expensive esoteric components in a suboptimal circuit.

I think that it is important to measure the intermodulation distortion (IMD) of microphone circuits, because practical sounds are complicated. If you have more than one instrument or voice, it is the IMD that determines if these sounds remain truly independent in a recording. That is also the reason that classical music is routinely recorded with microphones that are as distortion free as possible. Although I measured total harmonic distortion (THD) and IMD, I will only show you the IMD.

### From the Internet

I do not claim that everything in the new circuit is of my own invention. Why not use what is available and use it where applicable? Some interesting finds:

1. First of all, there is the set of articles by Scott Wurcer published in *Linear Audio* [2][3]. They explain a lot of details about JFETs, noise sources in microphone input circuits, and so forth.
2. Sometimes a remark on a website can inspire. In my case, this was a short description of the microphones used for recordings by Dutch record company Polyhymnia [4]. The company has developed its own buffer amplifier, which it uses in most of its microphones. Their unique design is completely balanced. The circuit, of course, is not available, but the idea is interesting.
3. I found a number of circuits from AKG, Sennheiser, Neumann, Schoeps, B&K, Rode, and others on the Internet. I analyzed and simulated the most basic versions. In a separate article, I plan to compare simulations of some of these circuits [5][6].
4. The excellent articles from Cyril Bateman on distortions in capacitors [7] set me thinking on how to design the new circuit with minimal capacitor influence (although given the other constrains, I did not completely succeed here).

### Requirements

My main requirements for the new circuit include:

- Performance should be better than the reference circuit
- No need for adjustments
- Low output impedance
- Normal 48 V phantom power drawing a current of maximum 4 mA
- No large elcaps and no elcaps in the signal chain
- No Zener diodes
- All components should be readily available
- The circuit should be small enough to fit in the housing I selected (57 mm × 15 mm)
- Especially for the Bruel & Kjaer 4165: approximately 200 V polarization voltage should be possible

For reference, I used an adapted "Schoeps" circuit as can be found in many Chinese microphones. **Figure 1** shows the reference circuit.

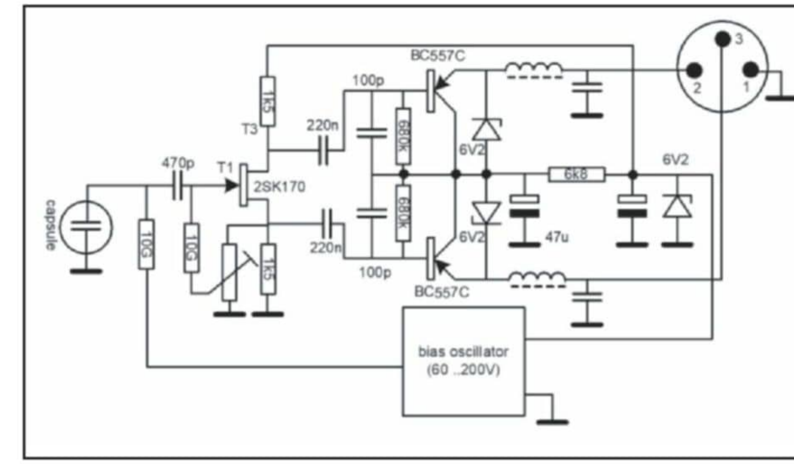



Figure 1: The reference circuit

The adjustment of the potentiometer in the reference circuit is critical (see Reference 1). Proper adjustment can reduce second-harmonic distortion with 40 dB or more via a compensation inside the junction gate field-effect transistor (JFET). But, the adjustment is temperature sensitive. A few degrees temperature difference will considerably reduce the compensation effect. Furthermore, only second harmonics are compensated in this way (and first-order intermodulation). Higher harmonics are not reduced at all. Most Chinese copies of this circuit lack the adjustment.

The gain is fixed at approximately 6 dB (without taking feedback into consideration caused by Cgd, the parasitic capacitance between drain and gate). Attenuation, filtering, and more are only possible with additional circuitry. The circuit looks very simple, but due to the large elcaps needed, it is not very compact and will not fit into the housing I selected.

### Optimal Use of Available Power

A phantom-powered microphone has limited power available for the electronic circuit. If we assume a maximum current of 4 mA, there will be a drop of 13.6 V across the 6.8 kΩ phantom power resistors. This means that for the electronics of the microphone we have 48 - 13.6 = 34.4 V available (137 mW). The question is how to use this power optimally. A standard microphone has three main processing blocks: the input stage with the impedance converter, the output stage with low output impedance for driving the (long) microphone cable, and the bias generator.

There are three basic options (see **Figure 2**). The first option is to put the three blocks in parallel. This means that the available current has to be distributed between three blocks. This is quite a challenge and since the blocks typically have a supply voltage of say 10 V, more than 24 V/4 mA is thrown away. This type of power distribution is found in older microphones, often with a transformer output.

The second option is more economical. Now the output stage has the whole 4 mA at about 10 V, making it easier to have a low output impedance. The input stage and bias generator each have say 2 mA at 10 V, and we throw away only 14 V/4 mA. This is basically the situation we find in the reference circuit.

The third option stacks all three blocks vertically. Each gets say 10 V/4 mA and all the available power is used. The challenge here is that all three blocks must be designed such that they work with the same current, but there is some freedom in the voltages available for each block. As I will illustrate, it combines very well with a symmetrical audio path.

### The Input Circuit

The size of the capsule of a condenser microphone is important. The off-axis sensitivity, high-frequency roll-off, impulse response, and intrinsic thermal capsule noise level depend on it.

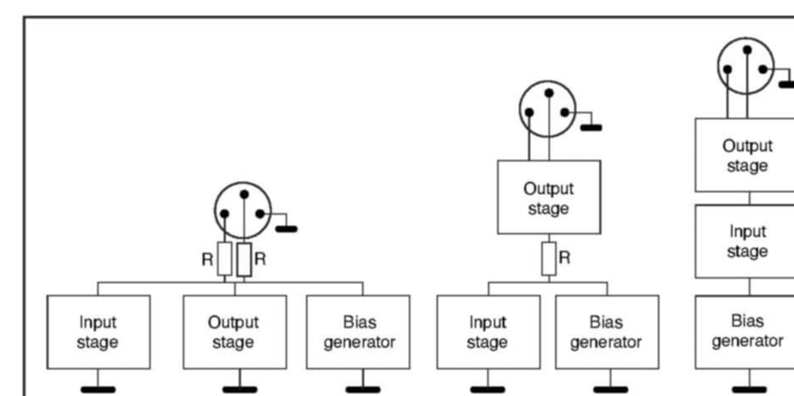



Figure 2: Three possible options for the use of the available power include: parallel, parallel/ serial, and full serial.

| Capsule Diameter | 1/4" | 1/2" | 1" |
|---|---|---|---|
| Capacitance | 6 pF to 7 pF | 15 pF to 20 pF | 40 pF to 60 pF |
| Thermal Noise | Level 30 dB | 15 dB | 10 dB |

Table 1: Capacitance and thermal noise of different size capsules is shown.

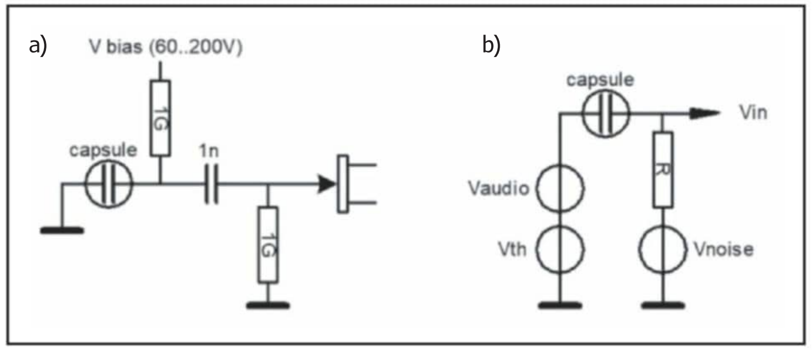



Figure 3: The basic input circuit: the circuit (a) and the equivalent noise circuit (b).

The smaller the diaphragm, the better the first three are, at the cost of the fourth. A good design of the input circuit should have less noise than the capsule noise (see **Table 1**).

**Figure 3** shows a basic input circuit. Vth is the thermal noise of the capsule. Vaudio is the audio signal and Vnoise is the combined thermal noise of the bias resistors and Vin is the resulting voltage at the input of the field-effect transistor (FET). Vth is determined by the choice of capsule and can be considered constant here.

We know that:

$$Vnoise = \sqrt{4KTR\Delta f}$$

This noise is low-pass filtered by the capacitance of the capsule.

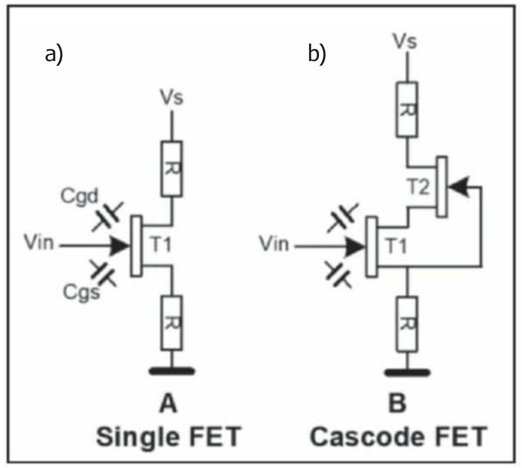



Figure 4: The options for the FETs include standard (a) and cascode (b).

$$\text{Vnoise at input} = \frac{1}{1+2\pi fRC} \times Vnoise$$

So the noise contribution of the R drops with the root of the value of R: Higher R means less noise. But it also means that the influence of the resistor noise increases with decreasing capsule capacitance, making it more of a challenge to make low-noise electronics for a small-size capsule!

On the other hand, we see that Vaudio is high-pass filtered by the combination of the capsule and R, so a higher R results in a better low-frequency response. For both effects it is best to have R as high as possible. I found 10 GΩ SMD resistors on the Internet for approximately €1.00 each.

It will be clear that any capacitance in parallel to the capsule will reduce Vin. It is, therefore, essential to construct the microphone so that any stray capacitance in the input circuit is avoided.

## The Impedance Converter

Approximately 99% of the impedance converters used in microphones (cheap, expensive, DIY, etc.) is asymmetrical, using a single N-channel JFET or tube. There are some problems with this:

1. The current drawn from the supply is not constant but varies with the audio signal. This could cause a slight variation in bias voltage when it is derived directly from the phantom supply.

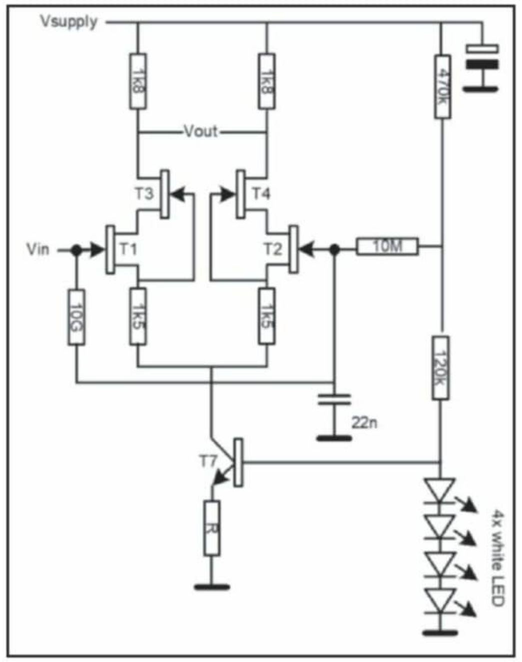



Figure 5: The impedance converter is shown with cascode FETs.

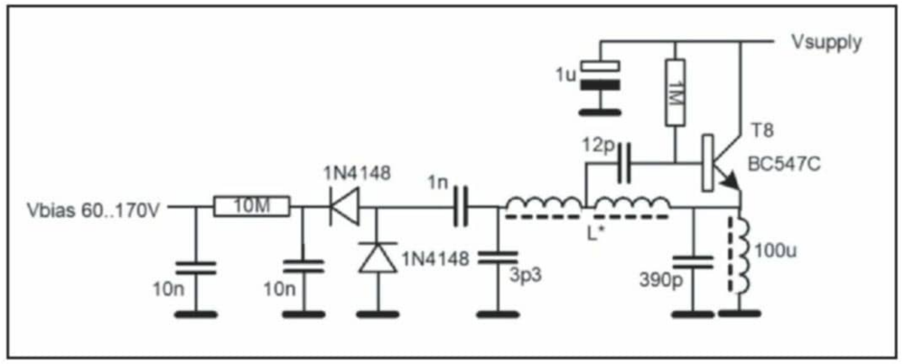
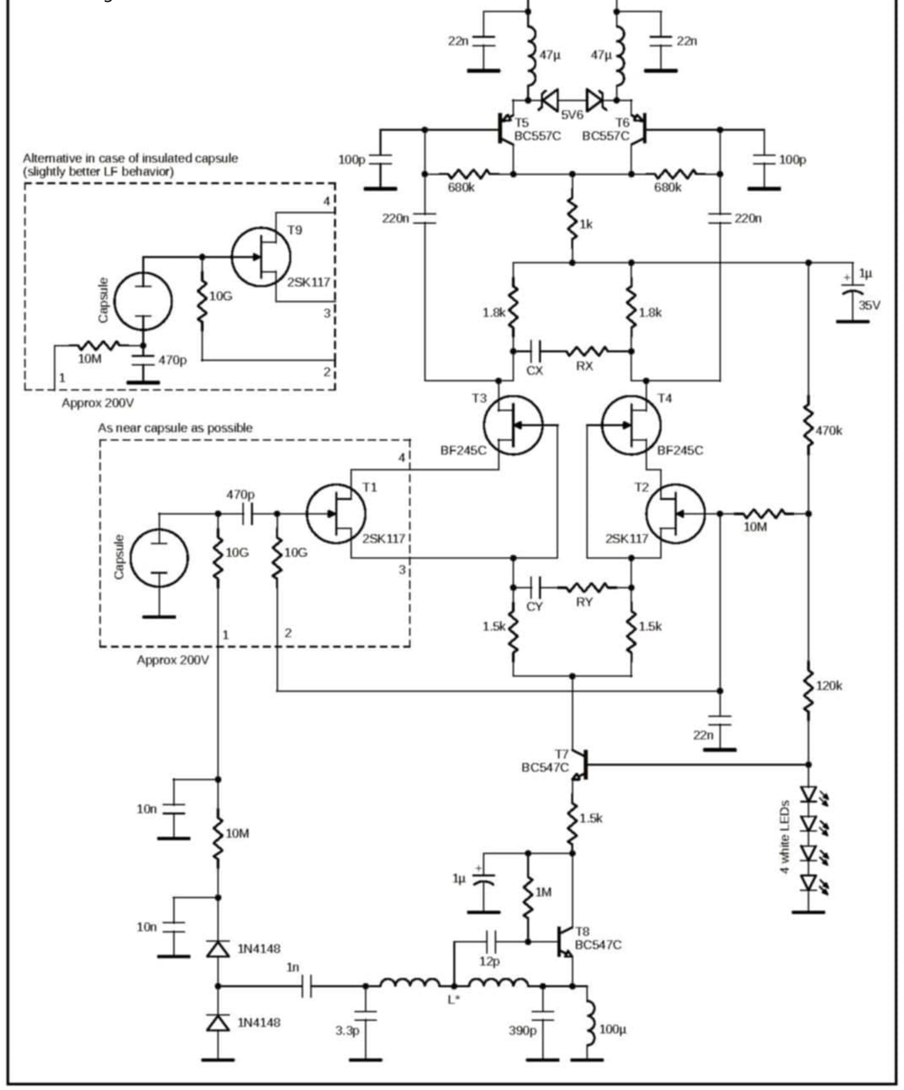



Figure 6: This schematic shows the bias oscillator.

Figure 7: The complete circuit, including bias oscillator



2. In most cases, the output is at the drain of the FET (or at both source and drain). In case the power supply is not 100% noise free (e.g., if the supply voltage is regulated with a Zener diode), this will have a negative effect on the signal-to-noise ratio (SNR).
3. The drain-gate capacitance of the FET works as a Miller capacitor, diminishing the output level. Since this capacitance is not linear, it also causes distortion.

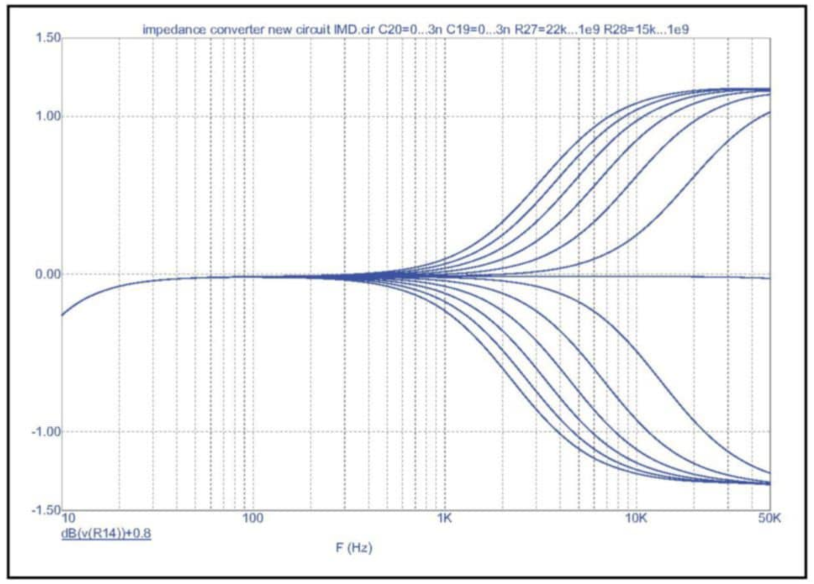



Figure 8: High-frequency tuning using Cx, Cy, Rx, Ry

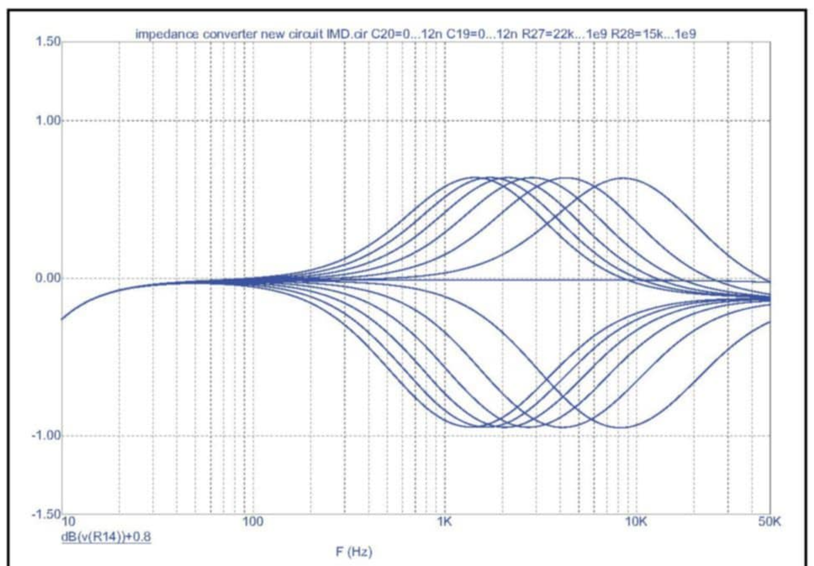



Figure 9: Mid-frequency tuning using Cx, Cy, Rx, Ry

### About the Author

**Michel Nieuwenhuizen** studied electronics at the Eindhoven Technical University in The Netherlands. He then worked for 27 years with Philips Electronics on audio and video processing, resulting in a number of patents on video image improvement. After his retirement he concentrated on audio projects such as a software package for automation of local radio stations called Caban and microphone techniques.

A symmetrical circuit made from a long-tailed pair potentially solves problems 1 and 2. For 3, there are several solutions. I opt for a simple one: Using a cascode FET with its gate connected to the source of the input FET (see **Figure 4**).

With the cascode circuit FET, the source of T2 will have the same AC voltage as its gate, which in turn has the same voltage as the gate of T1. So T1 has the same AC signal at all three of its pins. Cgs and Cgd both become irrelevant since there is no AC voltage across them, combining the cascode with a long-tailed pair gives the circuit shown in **Figure 5**. Any ripple on Vsupply will be the same at the drain of T3 and T4 and will cancel in Vout = VT3-VT4.

The current drawn by the circuit is constant, independent of Vin and only dependent on the current source T7. Since I certainly do not want a noisy Zener diode in my circuit, I used four white or blue LEDs to stabilize the reference voltage for the current source.

It is possible to design the circuit to have equal input impedance at both gates. However, this gives slightly more noise and my simulations found only a very small benefit for this trade-off. Therefore, I chose to have the impedance on the gate of T2 to be low.

## The Bias Oscillator

The bias oscillator has to reliably run on a voltage of approximately 5 V and a current of 3 mA to 4 mA, and (in the case of Bruel & Kjaer capsules) ideally need to deliver 200 V. I used a circuit that is not dissimilar to that of the Chinese microphones.

However instead of the standard axial coils, I use a 10 mm pot core with two windings each having 80 turns, giving 0.8 mH per winding. The output voltage is a bit too low (150 V), but **Figure 6** shows the best I could find given in this size.

## The Output Stage

For the output, I use the old "Schoeps" design. I could not find a circuit that is significantly better. A small improvement over the original circuit is possible by raising the base resistors to a value such that Vce is about 3 V. The downside of this is that we now must pair the two transistors to have the same beta, but this is a simple process. A negative point of the "Schoeps circuit" is that its large signal drive capacity is limited. So although the circuit works well with most microphone preamplifiers, I would suggest using preamplifiers with at least 5 kΩ input impedance since distortion is significantly higher at lower values. This is the case for real Schoeps microphones and all Chinese copies as well.

## The Symmetrical Circuit

**Figure 7** shows the complete circuit. Most small diaphragm capsules are constructed so that one side is connected to the housing, but the large diaphragm capsules are often floating. In that case, the alternative input circuit shown in Figure 7 can be used, saving one 10 GΩ resistor. Note that the 470 p capacitor must be high quality (polystyrene or polypropylene) in either circuit, since the audio signal flows through it! The overall gain of the circuit is approximately 0 dB.

With CX, RX, CY, and RY it is possible to fine-tune the microphone's mid-frequency and high-frequency response (see **Figure 8** and **Figure 9**). If this is not needed, they can be left out.

## Next Time

In the second part of this article, I will discuss the simulations and the measurements of this circuit as well as the practical implementation for it.

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# Design Considerations for a Hollow-State Mixer

Our columnist explores why the lure of the classic hollow-state recording mixer would cause so many engineers to prefer the UA 610 when we are well into "the solid-state era."

By
**Richard Honeycutt**
(United States)


Photo 1: This Allen & Heath GL-2400 analog mixer illustrates the plethora of controls available on a modern mixer.

Most mixers being released these days for either live sound or recording are digital. Until a couple of decades ago, most were analog. It goes without saying that all are/were solid-state units. **Photo 1** shows a 24-channel analog mixer typical of the late 1990s. A digital mixer would have similar functions, but they would be implemented differently.

## The Controls

Analog or digital mixers functionally similar to the GL-2400 shown in Photo 1 are available with up to at least 80 channels. Notice the vertical strips of controls—a slide fader, knobs, and switches—on the GL-2400. There are 14 mono channels with black fader knobs at left, then two stereo channels with blue fader knobs. Next, there is a group of seven faders controlling subgroups 1-4, plus the left, right, and mono outputs. At the right, there are eight more mono channels. Each of the mono channels has a slide fader, an LED level indicator just for that channel, assign push-button switches allowing the channel's signal to be panned between subgroup 1 and 2, subgroup 3 and 4, or the left or right output. Above these controls, there is a channel mute switch, then six level controls to send the channel's signal to any number of the six auxiliary outputs. The "aux sends" can be set to be taken either before (PRE) or after (POST) the fader.

Above the aux send controls, there are four channels of equalization (EQ): LF (bass), LM (low midrange), HM (high midrange), and HF (treble). A push-button switch enables the operator to bypass equalization for the channel. Above the EQ, there is a high-pass filter (HPF) so that the operator can easily filter out bumps and thumps. Next comes the input level (GAIN) control, a switchable pad to reduce the level of line-level input signals, a switch for reversing the polarity of the channel, and finally a +48 V phantom power switch.

Readers of the August 2020 Hollow-State Electronics article will know that this profusion of controls dwarfs anything available on the old hollow-state mixers by Western Electric, RCA, Collins, Gates, Altec-Lansing, and others. Those mixers, while serving admirably for radio broadcasts, live sound, and monophonic recording, were mainly limited to switching and gain-control functions. With the introduction of stereophonic phonograph records in 1957, plus three-channel movie sound introduced in 1940 and becoming more popular in the 1950s, more sophisticated mixers were needed.

### Resource

E. Allen, "Around The Shop: Arcade Fire's Custom Universal Audio 610 Console," Vintage King Audio, November 2018, https://vintageking.com/blog/2018/11/arcade-fire-universal-audio-610-console

## More Sophisticated Mixers

In 1960, Bill Putnam—sometimes called "the father of modern recording"—introduced a groundbreaking three-channel hollow-state mixer manufactured by his company, Universal Audio. The UA 610 was modular, as were many other mixers of the day. However, the 610 allowed defective modules to be unplugged and replacements to be plugged in without powering the mixer down.

In previous modular mixers, the module types included the power supply, input preamps, and program amps (output amplifiers). The 610 added bass and treble controls, the fader, a mic/line source switch, a switch to assign the channel output to the left, right, or middle channel, and an echo control—all within a single module. Echo was created by feeding the signal to a monitor speaker located in a reverberant room—typically a bathroom—and picking up the reverberated sound with a dedicated mic. This produced a more lifelike reverb sound than the "tape echo" reverb (also invented by Putnam) that was a signature part of the sound of Elvis Presley, Jerry Lee Lewis, and other rock-and-roll artists of the day.

This pluggable module was more than just a preamp, so Putnam dubbed it a "channel strip." The term has stayed with us: the controls running from the fader up to the phantom power switch on the solid-state GL-2400 mixer are still called a channel strip. **Photo 2** shows a UA 610-A channel strip.

**Figure 1** shows a block diagram of the 2-610-A (dual 610A). As with the older hollow-state mixers, the input and output were transformer-coupled. The hollow-state voltage amplifier (preamp) was a two-stage triode circuit using a 12AX7 and a 12AT7. Its gain control varied the negative feedback, allowing distortion to be controlled by the combination of mic placement and fader setting. The EQ was switched in 2-dB steps, rather than continuously as with most mixers today. Both the low-frequency and high-frequency filters were "shelving type," not too different from a Baxandall circuit in their effects.

**Photo 3** shows a complete UA-610 mixer. In the early 1960s, it was common practice for recording engineers to build their own mixers, and many 610s were hand-assembled using channel strips and other parts from UA.

UA 610 mixers were used for recording virtually all major artists of the 1960s, including many live performances by all the "Rat Pack" members, Ray Charles, the Beach Boys, and even the very last concert under the baton of Igor Stravinsky in Los Angeles, CA. Different engineers had a variety of favorite points concerning this mixer. Some loved the warm "tube" sound. Some liked the simplicity, predictability, and musicality of the EQ, and some even admitted that just the "feel" of turning a real knob on a nice warm tube mixer beat any experience they'd ever had with solid-state units.

## The Lure of the Classic Hollow-State Mixer

This overview of what many would call the classic hollow-state recording mixer brings up the question, what would cause so many engineers to prefer the UA 610 when we are well into "the solid-state era"? Possible answers include differences between the design goals of those who build solid-state mixers vs. the recording engineers who prefer the 610.

Most solid-state mixers are designed to be as clean (free from distortion) as possible. Ever since the beginning of the "hi-fi" era in the 1940s, very wide frequency response has been an almost universal design goal. Solid-state mixers can achieve a much broader frequency response than a triode-based mixer can. (Pentode-based mixers can achieve very wide frequency response due to their freedom from Miller effect, but their distortion signature is less pleasing to most engineers than that of triode-based mixers.) The equalizers in solid-state mixers usually have steeper slopes and are

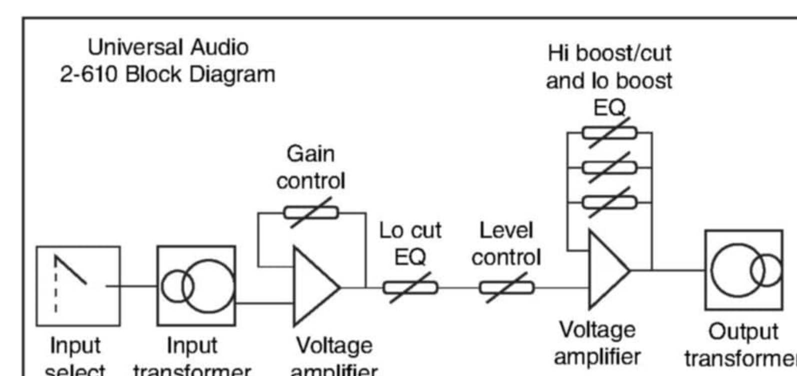


Figure 1: This is the block diagram of a 2-610-A channel strip.


Photo 2: The UA 610-A pluggable channel strip was one trendsetting feature of the UA-610 mixer.

more complex. Many solid-state mixers have a polarity switch on each channel. Most solid-state mixers have four to six or more auxiliary channels and at least two subgroups. Virtually all solid-state mixers include phantom power. In exchange for all this, solid-state mixers are more complex to operate than a UA 610.


Photo 3: This UA-610 is shown in a carrying case.

If the recording engineer prefers the sound of triode-generated harmonic distortion and minimalistic EQ with gentler slopes over lowest possible distortion and widest possible frequency response, and makes sure all mics are wired in the correct polarity (or uses an external switch box in case polarity reversal is desired for a special effect), (s)he may well prefer the hollow-state UA 610. And if the engineer routinely uses outboard mic preamps that supply their own phantom power, the UA 610 can be a viable choice that provides a good-sounding, easy-to-use basic mixing engine.

Engineer Eric Heigle describes the reaction of many engineers to the 610:

"This console is absolute magic. The sort of equipment you can do anything with from a production standpoint… The EQs are super musical, and you can be as extreme as you want without anything sounding harsh. The 610 modules also just have this really wide range in the preamps and EQs. There is an insane amount of headroom. Ain't nothing like the real thing."

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