INNOVATIONS IN AUDIO • AUDIO ELECTRONICS • THE BEST IN DIY AUDIO

www.audioxpress.com

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Market Update—Home Theater
Raising the Bar on Residential Entertainment
By J. Martins

Audio Electronics
Ultra-Linear Amplifiers: Why It Works (And When It Will Not)
By Morgan Jones

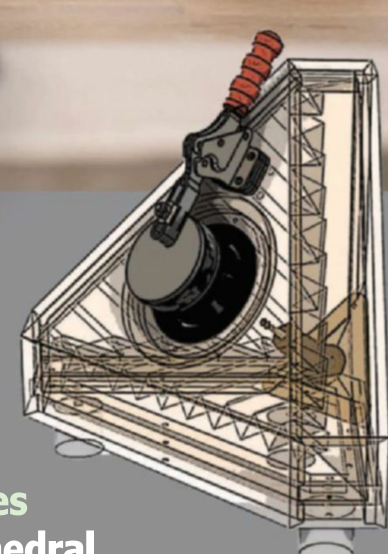
Amplifier Design: Cascodes, Folded Cascodes, and Current Mirrors
By Morty Tarr

Fresh From the Bench
USound Megaclite 4.0 UH-R2040 MEMS Earphones
By Brent Butterworth

50 Years

You Can DIY!
How to Refurbish 300 Pound Vintage Altec Lansing Speakers
By Bruce Heran

Sound Control
A Tale of Speakers in a Room
By Richard Honeycutt



R&D Stories
The Tetrahedral Test Chamber Story (Part 2)
By Geoff Hill

$9.00US $10.00CAN

OCTOBER 2021


HAMMOND
NEW HOMELAB &
PRO-AV RACK CABINET
READY TO SHIP
HAMMOND
MANUFACTURED IN
NORTH AMERICA
SINCE 1917
Knockdown Rack Cabinet | RCK Series
12U, 18U, 20U, 24U, 36U, 42U in 25″ & 32″ overall depths
Top and bottom feature a removable plate compatible with standard hardware components.
Venting options at front and side.
Top-rear of cabinet includes a bonus 1U rack panel with conduit panel installed.
NEW & IN-STOCK
Ready-to-assemble self-squaring cabinet body is load rated for up to 1,050 lbs.
Two pair of 10-32 threaded adjustable mounting rails. Includes 50-pack of mounting screws.
Cabinet base includes large opening for cable pass through and is designed for optional casters or leveling feet.
Bottom-rear of cabinet features a removable conduit knockout panel.
Interchangeable solid top panel compatible with Hammond Fan Tops. Conduit knockouts are also included for additional cable entry.
Optional doors (solid, window, vented) include a locking swing handle, compatible with electronic security options.
Hammond Manufacturing | www.hammondmfg.com
rackusa@hammfg.com | 1 800 526 2266

October 2021 ISSN 1548-0628

www.audioxpress.com

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

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

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

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

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

Advertising rates and terms available on request.

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

Home Theater and Home Entertainment

This edition of *audioXpress* offers a balanced mix of content addressing audio electronics knowledge, technology reviews, speaker building and measurement, and our latest market update report focused on home theater, and associated residential technologies.

I always find curious that some audio-centric publications choose to label themselves "two-channel" focused and refuse to address anything that falls outside that artificially created sweet spot. I understand the reason for that self-imposed restriction originates from the (limited) perception that home-theater speaker audio designs and integrated AV receivers that are dominant in the space—not to mention the dreaded home-theater-in-a-box products—compromise high-fidelity by design. I believe that perception to be misguided. Many great audio companies and many more installation professionals have already proven that's not the case. And we should not underestimate the fact that audio systems surrounding a living room TV was always part of valued home entertainment experiences for many families.

Because home theater is directly associated with a combination or integration of audio and video components for enjoyment of movies and audiovisual content, that has nothing to do with the ability of an audio system to be able to generate "fidelity," and neither does multichannel.

We evolved from single-source mono to stereophonic recording and reproduction, and naturally embraced multiple-channel formats to better accommodate different types of content. Multichannel designs don't need to compromise superior two-channel music reproduction when desired. Anyone who knows the residential custom-installation market understands that high-end stereo is very much part of that. In fact, volume sales for many high-end audio brands are channeled through the residential integration channels. There's a reason why many renowned audio brands are respected in both two-channel and the most sophisticated home cinema installations—and that includes some of the most famous North American audio manufacturers. On the European side, we can find some of the most respected brands active in both fields, while some even extend into studio monitors, reference cinema sound, and very high-end designs, all with commendable results.

The home theater and residential installation topics were planned for this October edition, directly targeting the edition that should be distributed at the CEDIA show, as it happened in 2019 in Denver, CO. That year, in the first Installation Market Update feature published by *audioXpress*, we essentially focused on commercial applications. In 2020, we focused on residential installation, home networking, and new products for entertainment and personal workspaces, as companies were refocusing on those applications segments. This year, the second year reflecting the restrictions caused by the global pandemic, *audioXpress* appealed to multiple manufacturers to submit information about their latest products and solutions intended for CEDIA Expo 2021.

This report, has largely benefited from those efforts, helping us to better project the market trends going forward, even if, as we wrapped this Market Update feature, most major manufacturers and exhibitors were forced to withdraw their onsite presence due to safety concerns. We weren't surprised by the decisions of most companies and professionals not to attend a trade show at this time. But the realization that we will probably have to wait until 2022 to witness relevant industry events, reminded me how important these industry reports are. Without a fully featured show, it's more important than ever to identify and underscore the trends. I think this report does that.

However, there are two important trends that I did not address directly. One I just mentioned here and is the importance of object-based audio formats and content pushing home entertainment to new exciting heights (literally). The second is how home theater is changing with streaming services and Smart TVs—essentially a connected display with built-in media players and network streamers. According to Strategy Analytics' Connected Home Devices report, global sales of Smart TVs grew by 7.4% in 2020, and household ownership in North America has surpassed 70% in 2021. More than a display, Smart TVs are a gateway for home entertainment, and increasingly they are shipping with support for wireless speakers, and object-based audio reproduction. This represents a huge opportunity for the audio industry to address.

J. Martins
Editor-in-Chief

The Team

Publisher:	KC Prescott	Associate Editor:	Shannon Becker
Controller:	Chuck Fellows	Graphics:	Grace Chen
Editor-in-Chief:	João Martins	Advertising Coordinator:	Nathaniel Black

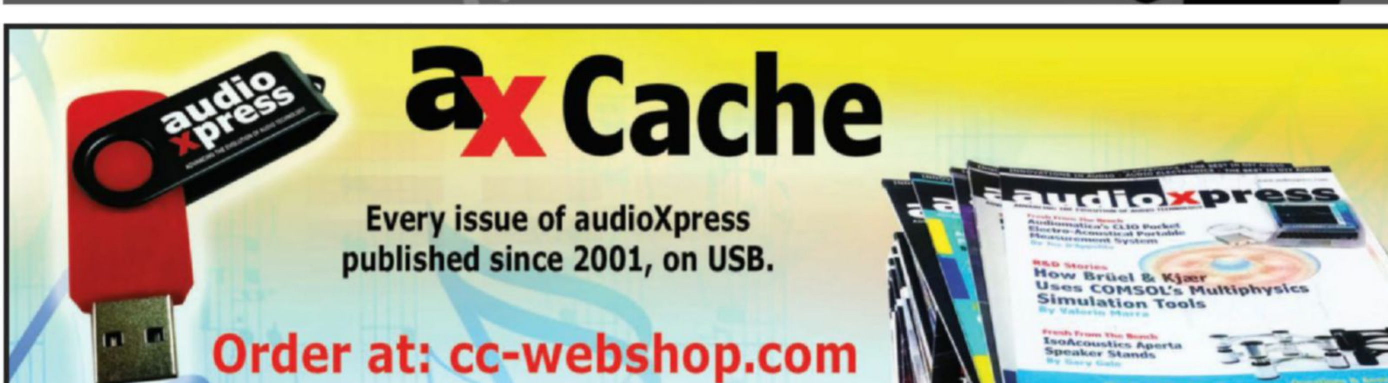
Technical Editor: Jan Didden

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

OUR NETWORK

audioxpress | circuit cellar | LIS Loudspeaker Industry Sourcebook | Voice Coil

SUPPORTING COMPANIES



Contents

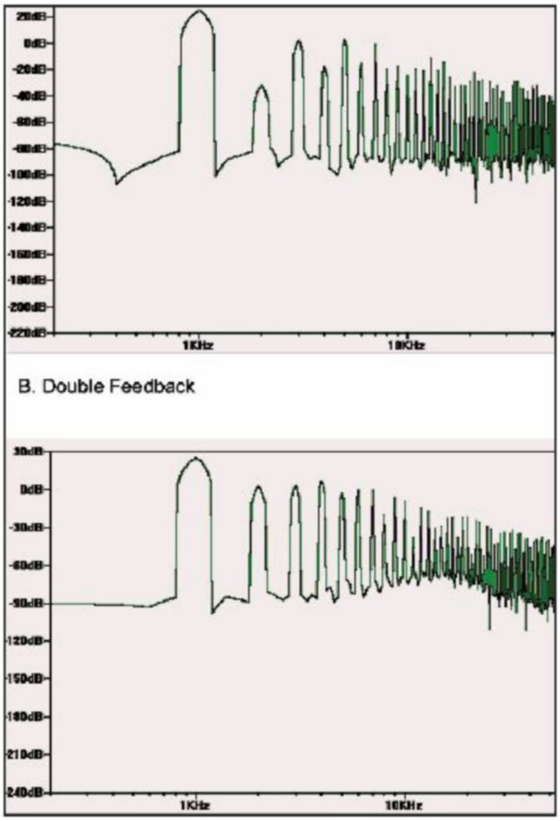
Features

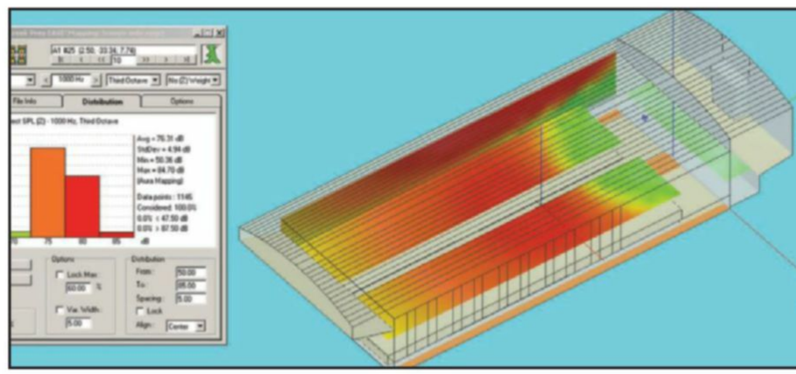
COVER ART

Photo Credit:
Polk Audio
Reserve Series
Speakers

Volume 52 – No. 10

October 2021





Websites

www.audioxpress.com

www.voicecoilmagazine.com

www.cc-webshop.com

www.loudspeakerindustrysourcebook.com

www.circuitcellar.com

www.linuxgizmos.com

@audioxp_editor audioxpresscommunity

linkedin.com/company/audioxpress

USound Megaclite 4.0 UH-R2040 MEMS Earphones

By
Brent Butterworth

The Megaclite 4.0 UH-R2040 is a reference design intended to prove that MEMS drivers can work well in a standard earphone design. Brent Butterworth listens and measures to see if this new technology from Austrian microspeaker company USound is ready for the market.

Microelectromechanical system (MEMS) technology has already had a huge impact on the microphone industry. It's now used in all smartphones, and in most microphone-equipped headphones and earphones. But doing the opposite—making sound, as opposed to receiving it—is much more challenging for MEMS devices because their size limits them to minuscule diaphragms with almost microscopic amounts of excursion.

However, MEMS speakers seem suited for applications that don't require high sound pressure level (SPL), such as earphones. Considering how popular balanced armatures—another technology that struggles to produce high SPLs—have become in hearing aids and higher-end earphones, MEMS speakers would certainly have a chance in applications where balanced armatures are currently used.

That's exactly what USound, an Austrian start-up focused on MEMS technology, is attempting to demonstrate with the Megaclite 4.0 UH-R2040, a reference-design earphone with one USound Achelous UT-P2018 driver per earpiece.

MEMS offers a few potential advantages over balanced armatures and dynamic drivers. The packages are slim. The Achelous UT-P2018 speaker measures just 6.7mm × 4.7mm × 1.58mm, comparable in length and width to smaller balanced armatures, but thinner (**Photo 1**). They're surface-mount technology (SMT) reflow-compatible, so easy to incorporate into automatic assembly processes. The piezoelectric drivers generate no magnetic field and minimal heat.

MEMS drivers can also be waterproof to a degree that challenges balanced armatures and dynamic drivers—USound's Conamara drivers carry an IPX8 water-resistance rating, which means they can be continuously immersed in 1 meter or more of water, and USound rates them for up to 30 minutes' immersion in 3 meters of water. (Currently, the most water-resistant earphones I've seen carry an IPX7 rating, which means they can be immersed in up to 1 meter of water for at least 30 minutes.)

For the most part, USound's performance claims aren't extraordinary. The Achelous UT-P2018 is rated for full-bandwidth applications, and USound's

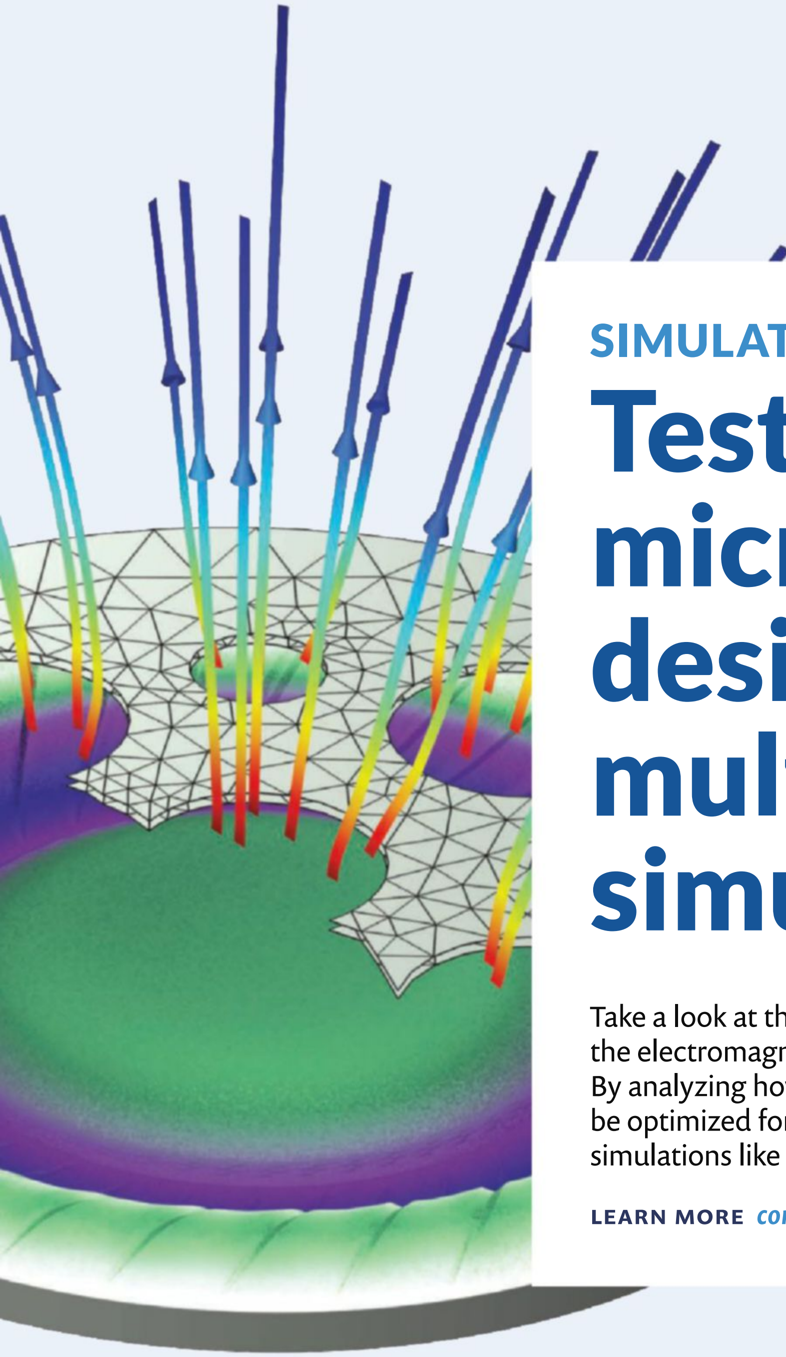
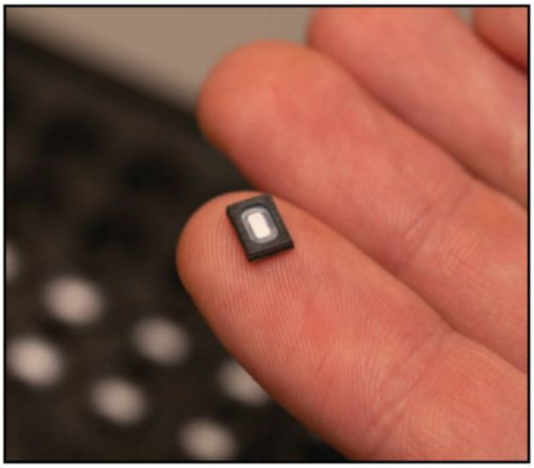


Photo 1: This is the Achelous UT-P2018 MEMS device, used on the Megaclite earphones under review. These full-bandwidth drivers are designed for in-ear audio solutions such as wired earphones or true wireless stereo (TWS) systems. Its small size and low weight allows greater design and tuning flexibility. One of the strongest arguments of the technology is volume consistency.



published driver specs show measurable response up to 40kHz. Maximum SPL is rated as 117dB at 1kHz, although that's with a 15V peak drive signal. Sensitivity is rated as 94dB at 1kHz with a 1Vrms signal. This low sensitivity, combined with the driver's need for a bias voltage, eliminates the possibility of passive applications.

When I found out about the Megaclite, I was eager to hear it. I've heard a few earphones that attempted to go beyond dynamic drivers and balanced armatures—specifically, with miniaturized planar-magnetic, electrostatic, and air-motion transformer drivers—but only one or two that, in my opinion, matched or exceeded the performance of conventional drivers. Considering that USound makes no extraordinary performance claims for its MEMS speakers, the bar was set pretty low—to succeed, the Megaclite merely had to deliver performance comparable to that of an ordinary earphone.

The Product

Looking at the Megaclite, you'd never guess that it incorporates a new and potentially revolutionary technology. From the outside (**Photo 2**), it's not much different from standard dynamic-driver earphones. The aluminum enclosures measure 12mm in diameter and 21mm long, terminating in a 6mm sound tube for which USound includes silicone tips in three sizes. The cable leg for the right channel incorporates an inline microphone/remote.

Photo 2: The earpieces of the Megaclite each contain a single MEMS speaker.

The source end of the cable incorporates a USB-C dongle (**Photo 3**) with an internal DAC, amplifier and drive electronics. The active design of the Megaclite prevented me from doing some standard measurements, including driver input impedance and sensitivity, but otherwise, I was able to use the Megaclite just as I'd use a set of passive earphones plugged into a USB-compatible DAC-amplifier, such as one of AudioQuest's DragonFly series.

I used the Megaclite with my Samsung Galaxy S10 smartphone and my Lenovo Ideapad Flex 5-1470 laptop, which has a USB-C jack. The earphones fit me just like typical dynamic-driver earphones, although as often happens when I review earphones, my extra-large ear canals required that I use SpinFit XL-size silicone tips to get a tight seal.

Listening

I compared the Megaclite with two sets of earphones: the $199 Campfire Audio Satsuma, which uses a single balanced armature; and the dynamic-drive AKG earphones (model number unspecified) that came with my Samsung Galaxy S10 phone. Unfortunately, the AKGs were the only dynamic-driver passive earphones I had on hand, because these days almost all the earphones I review are high-end, multiple-driver types with balanced armatures, or true wireless models. (That's a reflection of where the high-end and mass markets are going, and not necessarily indicative of a preference on my part.)

One thing I noticed right away is that the Megaclite's maximum volume was a little on the low side—not so low that I couldn't enjoy listening to them with most recordings, but with conservatively mastered music recordings that have a relatively low average level, I often found myself wanting 2dB or 3dB more volume. I'll address this in depth in the Measurements section.

When I played my all-time-favorite tonal balance test, Tracy Chapman's "Fast Car," through all three sets of earphones, it quickly became clear that the sound of the MEMS drivers was well within the norms of currently available earphones. Bass and treble both had extension comparable to what I heard from the other earphones. The balance of bass to midrange to treble was fairly similar with all three—the Megaclite and the AKG sounded very close, while the Satsuma sounded like it had a couple decibels more overall treble output than the others.

I heard no undue emphasis or muting of specific frequency bands, although the Megaclite's lower treble sounded a little coarser than that of the AKG and Campfire models, sometimes making Chapman's voice sound a little congested, as if she was suffering a mild cold. I heard the same vocal coloration when I played Holly Cole's version of "Good Time Charlie's Got the Blues." However, I've heard similar colorations with dynamic-driver earphones, and I imagine it could easily be fixed through acoustical tuning or DSP. It's important to note that AKG and Campfire Audio have extensive experience in tuning earphones, and devote considerable time to the process. It's hard to imagine a MEMS device company equaling their efforts.

Through a few weeks of regular listening, I didn't develop any major complaints about the Megaclite earphones—and for a reference design that's created merely as a proof of concept rather than a finished consumer product, that's pretty good.

Measurements

I performed measurements using a GRAS RA0402 high-resolution ear simulator/coupler (**Photo 4**). I used my Audiomatica CLIO 12 QC to feed signals to the Megaclite through my laptop's USB-C jack, and to analyze the signals coming off the RA0402 coupler. For volume (SPL) measurements, I used the RA0402 with an M-Audio Mobile Pre USB interface running Room EQ Wizard software.

After hearing how the Megaclite seemed relatively limited in maximum output, I measured its maximum output the same way I measure the maximum volume of kids' headphones for Wirecutter.com—I played pink noise at a level of -10dBFS, and measured the A-weighted output with the measured diffuse-field response of the RA0402 coupler subtracted. This test gave me a result of 90.2dBA SPL. To put this in perspective, active kids' headphones are (in theory, at least) limited to 85dB SPL, which is considered a reasonably safe level for 8 hours of daily exposure. For most adult listeners, 85dB will sound somewhat quiet.

Despite my reasonably careful listening habits (which have resulted in only average hearing loss for my age and gender, despite having played in lots of bands and making a living in the audio business for 30-plus years), maxing out at 90dB sometimes left me wanting a little more volume. USound's ratings suggest the drivers can take a lot more juice, but whether typical amp chips found in true wireless earphones can deliver that much power, I can't say without being able to test the Megaclite's drivers and amp separately.

For me, one of the joys of evaluating new technologies is seeing strange new things appear in the frequency response plots, and having to figure out what's going on. Sadly—or happily, depending on how you look at it—the Megaclite robbed me of this experience. Looking at the frequency response in **Figure 1** without knowing what was measured,

Photo 3: The integral amp-DAC dongle on the source end of the Megaclite's cable

Photo 4: The Megaclite is being measured in a GRAS RA0402 high-resolution coupler.

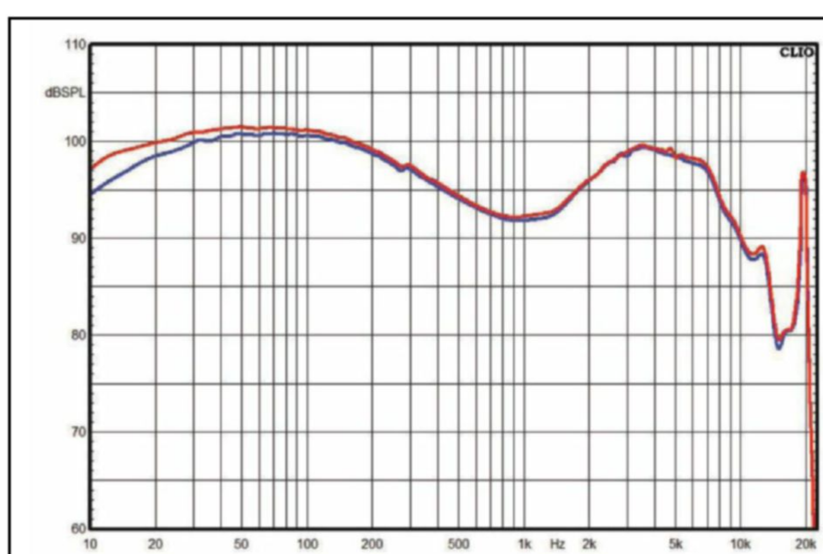


Figure 1: Frequency response of the Megaclite in left (blue) and right (red) channels, referenced to 94dB/500Hz

I'd never guess that the earphones use a new technology. I see two unusual characteristics. One is a single, very broad rise in the treble, between about 2kHz and 8kHz. With most earphones, we'd see two separate resonant peaks, one centered at 2kHz or 3kHz, and another centered at 7kHz or 8kHz. The other is a very high-Q peak centered at 19.5kHz—something I've never before seen in an earphone measurement. Both anomalies could be fixed with EQ. The latter would be audible to extremely few (if any) listeners.

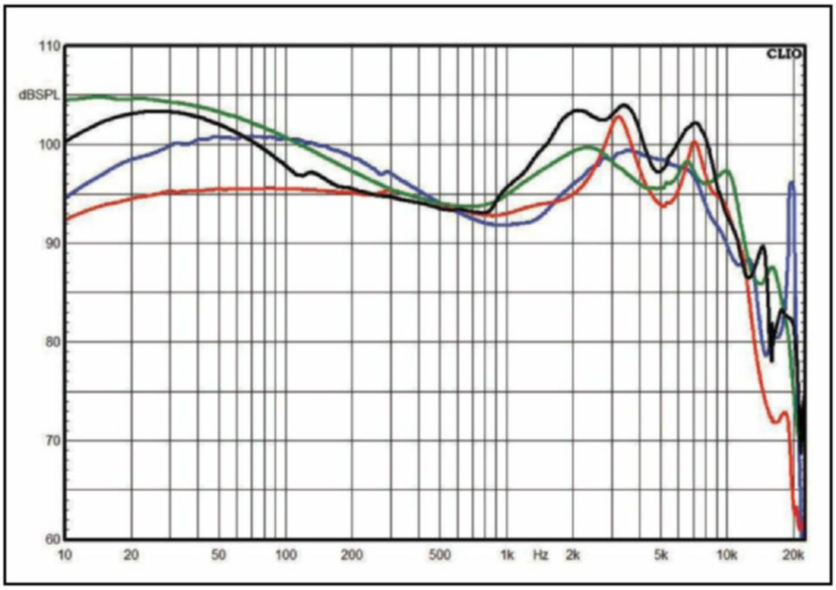


Figure 2: Right-channel frequency response of the Megaclite (blue), AKG N5005 (black), Campfire Audio Comet (red) and Technics EAH-TZ700 (green), referenced to 94dB/500Hz

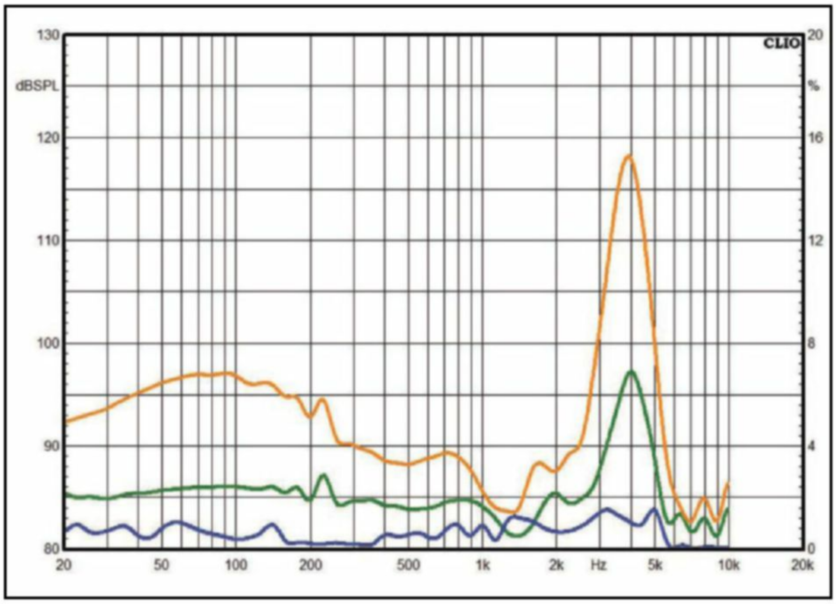


Figure 3: THD vs. frequency plot of the Megaclite at 90dBA (green) and 95dBA (orange), and the EarFun Free 2 at 100dBA (blue)

Figure 2 shows how the Megaclite's response compares with a couple of commercially available single-driver earphones: the Campfire Audio Comet (predecessor to the Satsuma), which uses a single balanced armature; and the Technics EAH-TZ700, which uses a single dynamic driver. I also threw in the AKG N5005, a hybrid design with a single dynamic driver and four balanced armatures, because it's the passive earphone that's said to come closest to the so-called "Harman curve"—the frequency response found in Harman International research to correspond best with average listener preference.

Again, there's nothing really unusual about the Megaclite's response. It does show a little excess energy in the upper bass/lower midrange region, around 150Hz to 400Hz, but that was a common characteristic of earphones before the Harman curve started to have an influence. I assume it can be fixed through acoustical tuning or the addition of a dynamic driver to cover the bass frequencies—something commonly added to balanced armatures.

Figure 3 shows the Megaclite's total harmonic distortion vs. frequency at a level of 90dBA (measured with pink noise) and 95dBA. Normally, I run these measurements at 90dBA and 100dBA. I chose these levels not because they are typical—they're actually much louder than most listeners would like—but because they straddle the typical break-up point for most headphones and earphones. Most I've tested can play at 90dBA with negligible distortion, but quite a few average 4% or 5% total harmonic distortion (THD), or even higher, at 100dBA. However, the loudest the Megaclite would play was 95dBA.

The rise we see here in distortion below 400Hz at 95dBA isn't that unusual, but the large distortion peak centered at 4kHz is. (These are average figures; as usual distortion varied with each sweep, so these are the most "middle of the road" of the 10 sweeps I tried at each level.) When testing distortion of active headphones, it's difficult to separate the performance of the amplifier from the performance of the driver, but I have to suspect this anomaly is coming from the driver, as I've never seen an amp produce such a distinct distortion spike, at least not within its normal operating range. I also included the 100dBA measurement of the EarFun Free 2, an inexpensive true wireless earphone, for reference.

Knowing that balanced armatures are rarely, if ever, used for noise-cancelling applications because their bass output is inadequate (noise-cancelling earphones with balanced armatures tend to use a dynamic driver for the bass), I ran some 100Hz sine-wave distortion measurements of the Megaclite plus the Campfire and AKG earphones previously noted. THD at 100dB and 110dB, respectively, was 0.143% and 0.618% for the Megaclite; 0.349% and 0.508% for the Campfire Satsuma (again, a balanced-armature model); and 0.129% and 0.405% for the AKG (a dynamic-driver model). Both passive models were driven by the headphone output of the Realtek audio system in my laptop.

Although the Megaclite's distortion was the highest of the bunch at 110dB, the margin wasn't large, and the fact that it came close to the dynamic earphone's performance at 100dB is encouraging—especially considering that the driver in the Megaclite is one of USound's first-generation MEMS speakers, while the drivers in the other earphones have been through countless generations and a few decades of refinement. So there's hope that single MEMS drivers might someday be up to use in noise-cancelling earphones.

Conclusion

Based on what I heard and measured from the Megaclite, I have no doubt that a MEMS driver can deliver sound quality that's competitive with dynamic-driver and balanced-armature earphones. It's likely that most earphones employing these drivers will include DSP, which should make it easy to tune out the few anomalies I heard and measured—as easy as it would be with dynamic drivers, I expect.

I will be curious to see how well these drivers integrate into typical active earphone systems, and if their power demands and distortion performance will be adequate to the task. But considering that this is a first-generation product, I think the results are promising, and I'm eager to see how earphone manufacturers can take advantage of the size, ruggedness, and practicality of these new drivers.

About the Author

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

Raising the Bar on Residential Entertainment

By
J. Martins
(Editor-in-Chief)

This article focuses on home theater and residential installation technology and solutions, with the goal to project the market trends going forward. It addresses both the products that appeal directly to the consumer, and installation-oriented solutions for home entertainment.

The wireless audio solutions available with support for the Wireless Speaker & Audio (WiSA) Association technology are great examples of products that will increasingly appeal to the consumer and end-user, while also becoming more attractive for residential integration. Not many home entertainment technologies are designed and promoted with a focus on installation simplicity and ease of configuration, such as WiSA, which continues to be an essential part of the technology puzzle to enable multichannel audio in the home. The interesting aspect about WiSA in the past two years is the clear momentum that this consumer electronics consortium—founded by Summit Wireless Technologies—continues to maintain, in great part as a result from direct integration in the latest generations of TVs and growing support from TV manufacturers.

By offering an interoperability standard for all leading brands and manufacturers to deliver wireless immersive sound, WiSA allows multiple components to be combined and ensure robust, high-resolution, multichannel, low-latency audio, while eliminating the complicated setup of traditional audio systems.

The number of TV manufacturers supporting WiSA Certified speakers keeps growing, with LG Electronics expanding its range of WiSA Ready TVs in four different product families at this time of writing, including the brand's latest LG OLED TVs and LG NanoCell TVs, as well as its CineBeam laser projector. Other large TV manufacturers, such as Hisense, Skyworth Group, Compal (Toshiba), Xiaomi, Foxconn, and TCL, all have joined the association and offer products that work together with more than 60 leading consumer electronics brands. On the speaker side, WiSA Certified speakers are available from well-known brands such as Bang & Olufsen, Harman, Klipsch, System Audio, and many more.

WiSA functionality allows for 24-bit at 48kHz/96kHz wireless audio distribution, in less than 1/10th the latency of a Bluetooth audio device, meaning no visible lip sync. WiSA installations will also largely benefit from the fact that devices are able to automatically recognize audio configurations from 2.0 to 7.1 to 5.1.2, which improves the user experience exponentially. The number of speaker brands is expected to dramatically increase as WiSA delivers on the promised Dolby Atmos support, as *audioXpress* detailed in this same Market Update a year ago.

European speaker manufacturers in particular seem to be very supportive of WiSA integration, and apart from a growing range of flagship and cutting-edge products from Harman Kardon and Bang & Olufsen, the association was recently expanded with products from Buchardt Audio, a Danish high-end direct-to-consumer loudspeaker manufacturer.

Getting such a wide variety of companies to agree on the use of a technology, which requires tight integration in order to deliver on its promises, is no simple feat and Summit Wireless Technologies continues to work hard to support integration efforts for smaller manufacturers, while making sure its WiSA Certified and WiSA Ready solutions work on multiple hardware platforms and operating systems, and get to market clearly advertising the benefits to the consumer.

One of the most recent efforts by the WiSA Association was the launch of its WiSA SoundSend Certified program, a new certification initiative developed to work with TV manufacturers to ensure simple connection and interoperability with the WiSA SoundSend wireless audio transmitter.

The WiSA SoundSend is the Association's first branded product to hit the market since its inception, and is an HDMI (or optical)-connected transmitter designed to make wireless multichannel audio accessible in minutes from nearly every smart TV. Smart TVs will receive WiSA SoundSend Certified status after they are successfully tested to work perfectly with SoundSend. This includes all audio connection and control requirements between the SoundSend and the smart TV. The strategy is projected to open up the WiSA speaker market to an estimated 1 billion+ TVs by the end of 2021.

The WISA SoundSend device easily connects to smart TVs via HDMI ARC/eARC and utilizes WiSA's wireless audio technology to allow immersive home cinema audio systems to be set up and connected quickly and easily. SoundSend decodes audio up to Dolby Atmos and sends it wirelessly through a low-latency, tightly synchronized high-definition, wireless connection to any WiSA Certified Speakers from WiSA Association member brands.

WiSA also recently launched a custom-branded Amazon Storefront, enabling all WiSA Certified products being sold on the Amazon platform to be purchased easily from one page. In one easy-to-navigate location, consumers are now able to shop for WiSA Ready TVs, WiSA Certified Transmitters, WiSA Certified Speakers, and promotional bundles with WiSA products.

Smart Home Matters

Our biggest regret for missing another CEDIA Expo show this year is the unique perspective that the event provides on the convergence between the very specific custom integration and installation sector and the growing smart home efforts from the large technology players. Even a relatively new audio company such as Sonos, which essentially grabbed a large share of the home audio market by cleverly and consistently growing its whole home audio ecosystem, is now doubling its efforts to support integrators and the CEDIA-space companies. Not seeing a show floor with all these companies together, makes it easy to overlook the similar expansion efforts that are being made by Amazon, Google, and other tech companies in the smart home space. And much in the same way as today it's impossible to disregard Sonos and wireless audio streaming in the home, it would be a big mistake to ignore the ongoing smart home efforts.

In addition to technology and certification programs, the role of WiSA is continuously expanding into a consumer-facing cornerstone of wireless home cinema. The WiSA SoundSend Certified Program is another step toward supporting the consumer electronics industry's TV manufacturers with the growth of WiSA technology.

As CEDIA Expo 2019 already anticipated—and the Integrated Systems Europe (ISE) in February 2020 highlighted even more—the discussion around interoperability, currently a dominant topic, remains one of the key issues, particularly because a lot of important developments have taken place in the past year and half.

It's important to follow the efforts initiated in 2020 between the Bluetooth Special Interest Group (SIG), the trade association that oversees Bluetooth technology, and the DiiA, the global DALI alliance of companies from the lighting and sensor industries. While the initial focus of the collaboration was to accelerate the adoption of connected commercial lighting systems with control via Bluetooth mesh networks, the availability of such solutions will sooner or later reach residential systems as well and will greatly expand the scope for Bluetooth applications.

As announced in May 2021, the Zigbee Alliance, the organization supported by hundreds of companies to promote open global standards for connected devices—the so-called Internet of Things (IoT)—with an important focus on the home, announced its organizational rebrand as Connectivity Standards Alliance (CSA), and the launch of Matter—a second brand, formerly known to the industry as Project CHIP (Connected Home over IP).

Matter is intended to serve as a seal of approval for smart home solutions, "secure by design, and compatible at scale." The Alliance will continue to develop Zigbee technology and will retain the Zigbee technology brand, but the expanded focus on Matter as the new IP-based standard promises to have a significant impact for the integration industry in general.

Matter combines existing connectivity standards, including specialized protocols (e.g., Smart Energy, Green Power, rf4ce, and others) into a common data model, benefiting from the direct support of 350 member organizations of all sizes, including Amazon, Apple, Google, Legrand, Lutron Electronics, IKEA, NXP, STMicroelectronics, and Texas Instruments.

Matter is intended to be the interoperable, secure, royalty-free connectivity standard for the future of the smart home, based on a unified IP-based connectivity protocol that enables communications among a wide range of devices. This standard will make it easier for device manufacturers to build devices, and to ensure they are compatible with smart home and voice services such as Amazon's Alexa, Apple's HomeKit with Siri, Google's Assistant, SmartThings, and others.

Products supporting the Matter protocol will not be available before the end of 2021, and will run on existing networking technologies, such as Ethernet (802.3), Wi-Fi (802.11), and Thread

(802.15.4)—and for ease of commissioning, Bluetooth Low Energy. In early May 2021, the feature-complete base specification was approved by the Matter Working Group, paving the way to build out the open-source implementation and test the specification.

This will be extremely important to promote confidence among consumers, who will see increased choice, compatibility, and more control of their experience. Developers will be able to get solutions to market faster. Key for developers is that Matter isn't only a specification—it also offers an open-source reference implementation with multiple use cases to test against.

Retailers will also gain confidence to stock compatible products, and more importantly, consumers will have more choice, and integrators will have expanded business with less questions from end-users.

"Matter will be a leap forward in interoperability. It also demonstrates the power of the collaborative and open-source process within the Alliance that embraces the full IoT value chain and yields results. We are convinced that Matter is a great opportunity, therefore Legrand supports the project since the beginning notably by involving engineers and by participating in test events," said Bruno Vulcano, the Chair of the Board at the Connectivity Standards Alliance.

We highlight this statement because Vulcano is also R&D Manager at Legrand, and for the company's announcements previewed for CEDIA Expo 2021, the company focused precisely on multiple home and residential installation product lines working in tandem to bring connected spaces together. Legrand already provides the foundational structured wiring and rack solutions for those connected systems, including reliable wired and wireless networking, connectivity, and power management products.

Legrand now combines a powerful portfolio of brands that even includes a wide variety of TV mounts and speaker stands for entertainment spaces, designed with the professional installer in mind. An interesting example is the SANUS suite of Sonos mounting solutions, which includes the Soundbar TV Mount for the Sonos Arc, and Extendable Wall Mount for the Sonos Arc, which are complemented with an In-Wall Power Kit for Mounted TV and Soundbar, which safely and discretely extends power to a soundbar, accompanying display, and other devices without the need to modify existing wiring.

While many custom-installation solution companies in the CEDIA space offer multiple options to support more exclusive high-end audio options, and the multiple integration-oriented audio manufacturers, Legrand is betting on supporting volume products that appeal directly to the consumer, such as Sonos. If you can't beat them...

Audio and AV-Over-IP

The third technology trend that we think deserves a mention in this Market Update focused on Residential integration has to do with Audio-Over-IP (AoIP, an acronym that seems to now also represent the growing range of AV-Over-IP network solutions, given that all available solutions now also carry at least one channel of video). And more importantly, as with commercial installation systems, residential integration is increasingly based on devices that can be powered directly over Ethernet—Power-over-Ethernet (PoE).

Launched in March 2021 by Canadian audio specialist Paradigm, the Founder Series is a six-model range of premium loudspeakers for home cinema systems, with new very high performance driver designs. This is the first completely new speaker line to be introduced since Paradigm founder Scott Bagby reacquired the company in 2019.

Independently of what technology or protocol is used, PoE is an established solution to deliver data and power safely for all sorts of devices over local area networks (LAN) using Ethernet cabling. This saves an enormous amount of time and money and, more importantly, allows designing complex systems with confidence, which is crucial in building and residential installations. As a result of its fast evolution, we now have three types of PoE solutions in the market, including switches, splitters, and injectors to support PoE devices.

PoE technology was defined in the IEEE 802.3af standard, published in 2003, allowing a powered device (PD) to receive up to 12.95W, and from 37V to 57V. In 2009, the PoE+ upgraded specification was introduced, extending power delivery up to 25.5W, with a voltage range of 42.5V to 57V.

More recently, in 2018, given the widespread use of the technology for all sorts of devices and applications, the standard was updated to PoE++. Instead of just using two twisted pairs in

Speaker manufacturers in the high end or more price conscious segments are all renewing their home theater options. Sonus faber expanded its acclaimed Lumina loudspeaker for both stereo and multichannel systems, now offering five models, and higher power options for larger environments.

Products with the Matter mark will serve as a seal of approval, taking the guesswork out of the purchasing process and enabling businesses and consumers alike to choose from a wider array of brands to create secure and connected homes and buildings.

an Ethernet cable, PoE++ allows using four twisted pairs. PoE++ delivers up to 71.3W to the PD, and allows 1.7A at 52V to be sent over the same cabling with Gigabit Ethernet. Developers are now exploring the full potential of the PoE++ standard, creating new power management features for optimized systems. For the audio industry, this means that powered speakers can now be conveniently installed using a single cable structure, while at the same time the network data can be used to remotely optimize and reconfigure an existing installation for immersive audio applications, virtual acoustics and even noise cancelling applications—which is the ultimate level of comfort for residential installations (cancel all external noises, define an ideal acoustical environment, and playback multichannel audio recordings in perfect conditions).

To transmit and distribute the audio from source to destination, the dilemma remains opting for AES67, Ravenna, AVB TSN, Dante, or more or less proprietary solutions.

A PoE-capable external audio amplifier. The market offers plenty of adapters for PoE, but still lacks fully integrated amplifier modules designed for PoE, as required for true audio networking (e.g., for immersive applications). A PoE amplifier is the ideal module for self-powered speakers, allowing a single cable for power, audio and control, and great installation flexibility.

Evolution in the domain of audio protocols for AoIP has stagnated in the past two years, with the industry apparently focused on adding video, or establishing further options for control, monitoring (AES70), and device discovery. There is still much to do to expand interoperability in AVoIP systems for the residential space.

Audio networking still remains far too complex for non-trained users, unfortunately keeping AoIP technology in the domain of specialized integrators. A detailed description of advances in audio networking is out of the scope for this Market Update and is something that *audioXpress* will address in a dedicated feature.

Still, it remains clear that between HDMI and USB interfaces that are common and increasingly dominant in all sorts of audio and video equipment, covering the distances required for residential integration remain the domain of networking. Whether this will evolve within the Ethernet/IP framework or converge with wireless standards is an important question—not to mention the already-discussed integration with smart home ecosystems.

When the choice is not simply relying on wireless systems—as consumers will increasingly tend to do—the choice of a wired local network currently implies opting for one of the existing audio and video transport protocols, including dumbed-down physical layer solutions.

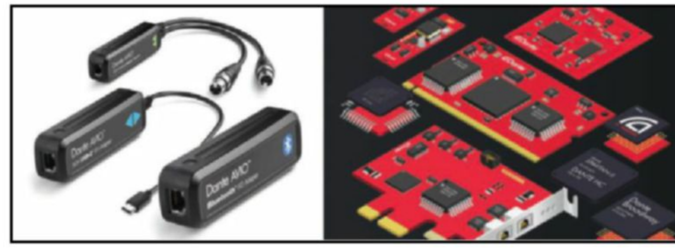
Of course, when the complexity of audio networking gets in the way of simply doing an installation that connects a few source devices to a few destinations, proprietary implementations over Ethernet are available as well. As an example, Just Add Power is a company that has proven that there is room for the approach.

The company just expanded its range of black-box devices, with the new MaxColor 4K60 Series devices, allowing 4K source content to be distributed over existing network infrastructures. Just Add Power also offers the 2GΩ/3G ST1 Sound Transceiver, an audio-only component that functions as either an input or output. With the ST1, integrators can easily add a stereo audio source to any Just Add Power system or extract stereo audio from an existing transmitter. Just Add Power could very well be renamed "Just Add Boxes."

Dante or Not

The Alliance for IP Media Solutions (AIMS), which inherited all the remarkable AES (AES67) and SMPTE (SMPTE ST 2110 suite) standardization efforts in audio-over-IP, is now focused on achieving an all-IP ecosystem using open standards, working in partnership with the Advanced Media Workflow Association (AMWA) and the Video Services Forum (VSF), mainly targeting AV and broadcast applications. Unfortunately, there doesn't seem to be any focus to implement the existing standards in simple, ready-to-use, plug-and-play networking solutions for home/residential environments, which is frustrating.

In contrast, Dante, the technology developed and supported directly by Audinate, is the de facto standard for digital audio networking in both commercial and residential installation environments, increasingly making inroads even in demanding, time-sensitive broadcast and sound reinforcement applications. Dante supports the distribution of hundreds of uncompressed, multichannel digital audio channels over standard Ethernet networks, with near-zero latency and perfect synchronization. In addition,



Audinate offers ways for manufacturers to support Dante audio networking with multiple choices of boards, silicon, processor architecture, and even embedded and software solutions. For end-users the availability of the AVIO adapters enables easy conversion to/from Dante.

Dante allows audio, control, and all other data to coexist effectively on the same network.

Recently, Audinate focused on expanding its efforts to integrate video, with its Dante AV development, which is now available in the first commercially available products—networked PTZ cameras being an example.

Dante AV also uses the same Dante Controller software to configure and manage the system and combines the efficiency of a JPEG2000 codec with a single network PTP clock for accurate time alignment of both audio and video. For manufacturers, this means licensing the technology, but at least it also means getting access to thousands of compatible products in the market. The Dante AV over IP networking solution is now also proven and, more importantly, available in a variety of silicon, embedded, and even software-only platforms.

And the pandemic also served as an important incentive for many integrators to fully explore the potential of Dante in completely new use cases, from corporate meetings and presentations, remote collaboration, all the way to home integration. Another big reason why Dante expanded like no other audio networking technology over the past 3 years is directly the consequence of a clever strategy by Audinate, of delivering simple and affordable interface solutions. Now available, the AVIO range of Dante interfaces has expanded to cover USB-C and even Bluetooth, making it easy for anyone to jump in.

And there's also increasing reasons to convert to Dante, since an even larger number of products is expected to reach the market as a direct result of Audinate's expansion into ARM-based and software implementations.

Following the announcement of support for Analog Device's SHARC processors, earlier in 2021, Audinate announced the availability of a software development kit (SDK) for ARM-based audio products, expanding on the company's growing number of ready-to-use modules, development kits and software, enabling faster time-to-market. Dante Embedded Platform is a software implementation targeting audio and video products implemented on Linux running on commodity ARM processors (32- and 64-bit ARM Cortex-A), and there's even a reference kit based on NXP's i.MX 8M Mini QuadLite.

These platforms and development ecosystems are a very comfortable space for audio developers, opening up a world of convenient processing, signal routing, and amplification approaches, making it even easier to embrace networking as well.

What About Home Theater?

All the technologies previously discussed are available to support both complex residential installations that carry whole-home audio and entertainment options, as well as integrating dedicated home theater installations. In fact the "home cinema" and home theater market has significantly expanded as consumers are investing more in their existing homes, and as a direct result of the expansion of OTT streaming services and distribution platforms, promoting the ability to enjoy higher-quality video (4K HDR) and sophisticated immersive audio content.

As CEDIA, the residential technology integration association, discussed in preparation for its CEDIA Expo 2021 and related education events, the Home Theater category is front and center in the minds of the industry. Both projector and TV-based rooms, media servers, speakers and electronic components, acoustic treatments, and even mounting accessories and seating are seeing strong demand.

"Home theaters are becoming increasingly popular. As more people invest in their entertainment spaces, new trends are also arising. Private cinema/dedicated theaters are back strong, along with multipurpose room home theaters," the association announces. In 2021 a number of new exciting companies have made an appearance in the segment, including US-based Perlisten Audio, a new audio brand focusing on high-end two-channel, home theater, and immersive audio applications, which we address in a dedicated highlight in this report.

CEDIA's education initiatives, which are available online and on-demand are clearly expanding the focus of home theaters from entertainment spaces to versatile installation that can also extend into wellness—a new "thing" to a lot of integrators; however, not a foreign concept to homeowners, as CEDIA notes. Obviously the whole residential installation segment is also now growing with communications, remote learning and collaboration, and work from home spaces. Serious trends of which integrators are already taking note, and many manufacturers will benefit to address. How? It's all about integration, and making sure the technology pieces of the puzzle we just described in this article are in place.

Resources

Audinate | www.audinate.com

CEDIA Expo | www.cediaexpo.com

Connectivity Standards Alliance | www.csa-iot.org

Institute of Electrical and Electronics Engineers (IEEE) IEEE 802.3bt-2018 - IEEE Standard for Power over Ethernet | https://standards.ieee.org/standard/802_3bt-2018.html

Just Add Power | www.justaddpower.com

Matter | www.buildwithmatter.com

OCA Alliance | www.ocaalliance.com

RAVENNA (ALC Networx) | www.ravenna-network.com

Summit Wireless | www.summitwireless.com

The Alliance for IP Media Solutions (AIMS) | www.aimsalliance.org

The Ethernet Alliance | www.ethernetalliance.org

Wireless Speaker & Audio Association | www.wisaassociation.org

Lithe Audio

First WiSA Certified Dolby Atmos Ceiling Speakers

Unveiled earlier in November 2020, and launched in April 2021, the new Wireless Speaker & Audio (WiSA) Association Certified Pro Series in-ceiling speaker from British brand Lithe Audio is aimed firmly at the custom installation market. It also represents WiSA's first certified speaker in the installation space, priced at approximately $950 USD.

Lithe Audio is a UK-based manufacturer that creates a range of high-quality installation products including integrated speakers, garden speakers, and TV accessories (e.g., wall mounts for commercial or residential applications).

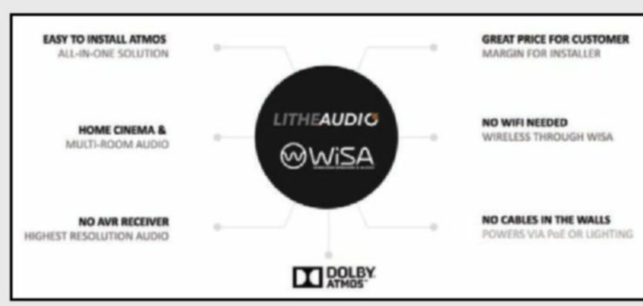
The new Lithe Audio Pro Series in-ceiling speakers with Dolby Atmos support will expand any WiSA system with up to height channels, also expanding other wireless and wired connectivity options for custom integration. The series makes its performance and integration abilities possible by including the latest chip sets for PoE++ and the growing category of WiSA interoperability, ushering in a new era of convenience in creating Dolby Atmos home cinema systems when combined with speakers from WiSA member brands such as Bang and Olufsen, Harman Kardon, and Savant.

Requiring no speaker cables, only the provision of power, WiSA technology means a full Dolby Atmos system can be achieved quickly and conveniently when combined with Dolby Atmos decoding transmitters (e.g., the new WiSA SoundSend device), opening the full immersive audio experience to more rooms and more users.

The Lithe Audio Pro Series speaker—largely based on the technology and speaker design that the company introduced at ISE 2020—was named a "Best New Hardware" winner at the 2020 CEDIA awards. Products in the range work via Bluetooth- and Wi-Fi-based solutions, including AirPlay 2 and Chromecast, supporting also major multi-room and control ecosystems.

"We have been building this product for two years, basing our approach on what has been successful with our other speakers and listening to the wants and needs of the custom installation community, which included bringing the wireless operability of WiSA technology into the fold," says Amit Ravat, Co-Founder and Director of Lithe Audio. "Since attending numerous industry events, shows and CEDIA Tech Forums, we are confident that we created a product that delivers what the market is demanding, our most high performance and feature-rich creation yet."

www.wisaassociation.org | www.litheaudio.com

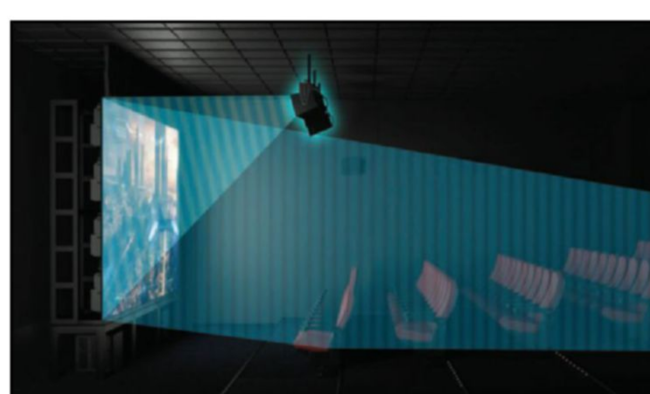


Meyer Sound

New Ultra Reflex Sound for Residential Cinema

California pro audio speaker manufacturer Meyer Sound is the perfect example of a company that over the past years invested decidedly in the residential space, and more specifically, home cinema applications. Meyer Sound is a renowned supplier of cinema sound systems for commercial theaters and content production applications. As part of a broad initiative to boost its presence in the high-end residential cinema market, Meyer Sound even signed on as a Platinum Supporter of the Home Technology Association (HTA) and is promoting its certification program.

As more and more film industry creative professionals wished to "take the sound home," Meyer Sound successfully migrated into the wider upscale private cinema market. The company offers a full line of cinema products and solutions for applications of all scales, from the Bluehorn System full bandwidth studio monitor,



Meyer Sound Ultra Reflex sound solution for direct view displays

Amie Systems, and Acheron screen channel loudspeakers to the new Spacemap Go spatial sound design and live mixing tool, and the Constellation acoustic system.

In 2021, Meyer Sound announced a breakthrough for both the post-production and private cinema markets with the introduction of the Ultra Reflex sound solution for direct view displays. This relates to a fundamental shift to replace traditional projectors with large active displays that occurred first in luxury private cinema rooms, now taking over large studios and scoring stages, and starting to conquer commercial theaters. With speakers no longer allowed to be positioned behind the screen, Meyer Sound perfected its Ultra Reflex approach, which combines the superior experience of direct view displays while still enjoying precise localization and full fidelity from the screen channels.

The Meyer Sound HTA speakers are positioned hanging from the ceiling, facing the screen, and tilted in a manner that directs the sound reflected in the hard surface of the display, directly toward the audience. Subwoofers are positioned under the screen and surround and height speakers are positioned normally around the room.

www.meyersound.com

A Meyer Sound Ultra Reflex solution installed in a home cinema

Polk Audio

Reserve Series of Premium Loudspeakers

Polk Audio recently launched the Reserve Series, a high-performance, versatile loudspeaker line that combines Polk's flagship components and advancements in cabinet design at an extraordinary value. IMAX Enhanced, Hi-Res Audio Certified, and compatible with both Dolby Atmos and DTS:X, the Reserve Series includes nine models available in matte black, matte white, and walnut woodgrain finishes.

The Reserve Series uses the same custom-made transducers originally developed for Polk's Legend Series loudspeakers, including the company's proprietary Pinnacle Tweeter, Turbine Cone midrange and Polk's latest bass-management and resonance control technologies—PowerPort and X-Port. The new series is well suited for an array of applications, including immersive multichannel home theater systems and classic stereo listening.

The series consists of nine models—three floorstanding models, three center channels, two bookshelf speakers, and a wall-and speaker-mountable height module—all designed to give listeners flexibility in terms of configuration by incorporating matching transducers and consistent voicing. The center channel speakers are available in three sizes, and for the first time, Polk has introduced a height module (R900), which can be placed on the top of the floorstanding speakers or wall mounted. The height module features a toggle switch, which tunes the speaker for the application, whether that be on-speaker or wall mounted.

The Polk Audio Reserve Series includes a five-year warranty standard and is available at select specialty audio/video dealers and custom integrators worldwide.

www.polkaudio.com | www.soundunited.co

Polk Audio designed a complete series of speakers that is ready to meet the demands of music, gaming, and home theater.

For its Reserve Series, Polk Audio introduced a height module (R900), which can be placed on the top of the floorstanding speakers or wall mounted.

ONKYO

WiSA Certified Sound Sphere Home Theater System

Directly appealing to those users who need a home-theater plug-and-play solution in more space restricted homes, Onkyo launched its WiSA Certified Sound Sphere audio system, delivering hi-res (24-bit/96kHz) wireless audio, and cinema-quality sound in a convenient WiSA Certified solution.

The Onkyo Sound Sphere includes the WiSA SoundSend transmitter and is available in 2.1, 3.1, and 5.1 system bundles, offering flexibility for consumers to purchase a format that best suits their needs and environment. Thanks to WiSA Certified speakers and the WiSA Soundsend, these Onkyo surround sound systems provide immersive, theater-quality sound experiences without the need for wires and after only minutes of setup.

By bundling with WiSA's first-ever branded product, SoundSend, audio can be wirelessly transmitted to Onkyo Sound Sphere speakers and subwoofers with extremely low latency. SoundSend can transmit up to eight high-quality audio channels at the same time, and its mobile application enables easy access and control features like MyZone, which allows users to create a "sweet spot" for ideal listening. The Onkyo systems are compatible with the latest surround audio format Dolby Atmos Height Virtualizer and its speaker sound quality matches and perfectly complements the visuals from today's high-resolution televisions. These bundles were initially launched in the Japanese market.

www.onkyo.com

Yamaha

Next-Generation Home Theater System in a Box

Yes, the dreaded home-theater-in-a-box concept is not dead yet, and in fact Yamaha proved that there's even room for expansion by simply leveraging the latest technologies, matched with convenient wired or wireless speakers. At the center of the company's vision, the integrated AV receiver remains a key component, and Yamaha did significant investments in 2021 to refresh its product range.

The latest YHT-5960U is an all-in-one 5.1-channel 80W solution—complete with a 4K/120Hz AV receiver, speakers, and wiring connections for a surround sound experience—designed to support the latest trends and specifications for home entertainment. This includes Dolby Vision, Dolby TrueHD, and DTS-HD Master Audio, with support for dynamic displays at 4K/120Hz, 8K/60Hz, HDMI 2.1, and HDR10+.

The Yamaha YHT-5960U all-in-one 5.1-channel 4K/120Hz AV receiver solution

The YHT-5960U features the iconic new look for Yamaha AV receivers, with a simplified, modern design, high-resolution LCD, and a jog dial with touch-sensitive buttons. The YHT-5960U offers five channels of surround sound, enabling users to set connections in a matter of minutes. All four HDMI inputs on the AV receiver support 4K/120Hz and 8K/60bHz, via firmware update, and couples video pass-through with extensive DSP options for all types of content, from immersive audio to gaming. And matching the current wireless connectivity standards, the YHT-5960U features the entire suite of MusicCast capabilities and app control, including Wi-Fi, AirPlay 2, Spotify Connect, built-in music streaming services, multi-room audio and voice control via Alexa, Google, and Siri-enabled devices. All at an affordable price ($649.95 MSRP).

Aventage AV Receivers

For those demanding a bit more, including support for Dolby Atmos immersive audio and the connected home, Yamaha refreshed its Aventage AV receiver range, with new models that support all the latest standards, media formats, and technologies expected today.

The all new, reimagined 2021 Aventage lineup includes the RX-A8A, the RX-A6A, and the RX-A4A, offering 11-, 9- and 7-channels, respectively. From stereo to spatial audio, and from 4K today to 8K, the range represents a new reference for Yamaha, featuring seven HDMI 2.1 inputs supporting 8K60/4K120 and Dolby Vision on each model.

Underneath the simplified, modern design, all chassis provide stability against vibrations and anti-resonance technology. The RX-A8A, the RX-A6A, and the RX-A4A models all support immersive audio with Dolby Atmos and DTS:X, enabled by the Qualcomm QCS407 smart audio platform. This SoC provides powerful quad-core audio processing for cinematic listening experiences and is designed to decode and play back a rich variety of audio content. The top two models also support Auro-3D.

In order to deliver maximum performance in surround sound modes, each receiver comes equipped with Yamaha Parametric Acoustic Optimizer (YPAO) automatic room calibration with multi-point measurement, precision EQ, and a new low-frequency mode. This included technology detects speaker connections, measures the distances from them to multiple listening positions in a room, and automatically optimizes speaker settings (e.g., volume balance and acoustic parameters).

All models also feature premier ESS Sabre audio DACs, combined with a high slew rate amplifier circuit, which more than doubles the performance of the previous generation, allowing it to respond to rapid changes of input levels with precision.

Otherwise, the new Aventage models offer the entire suite of Yamaha MusicCast capabilities and app control, including Wi-Fi, AirPlay 2, Spotify Connect, built-in music streaming services, multi-room audio and voice control via Alexa, Google, and Siri-enabled devices.

www.yamaha.com

The 2021 range of Aventage AV Receivers, including the smallest RX-A2A 7.2-channel model

The Yamaha RX-A8A Aventage 11.2-Channel Dolby Atmos AV Receiver with 8K HDMI and MusicCast

SVS

3000 Micro Subwoofer with Dual Opposing 8" Drivers

For many home theater installations on a budget and rooms of limited dimensions, achieving a convincing level of bass is always a challenge. Most standard subwoofers that ship with 5.1 packages don't deliver the quality of the cinematic experience and are often just too large for the room.

That's where highly dedicated companies (e.g., SVS, based in Youngstown, OH), delivers excellent value, with great designs that, year after year, receive CES Innovation and EISA Awards and are praised by users.

In 2021, SVS delivered a subwoofer that is probably one of its most impactful designs, given its compact dimensions. The SVS 3000 Micro subwoofer is able to reach 23Hz from a compact 10" cabinet, using dual opposing 8" drivers, connected in parallel to a single 800W RMS Sledge STA-800D2 power amplifier, originally designed for the much larger SVS 16-Ultra subwoofer.

Engineering a subwoofer capable of output of the lowest frequencies, while maintaining effective thermal and energy management in a micro-sized enclosure, has always been the challenge for manufacturers. The 3000 Micro answers the challenge with fully active dual opposing 8" SVS drivers, firing in opposite directions in unison. This cancels the mechanical energy transferred to the cabinet and solves the curse of micro subwoofers moving around a room. To help tame the power and response of the 3000 Micro is a 50MHz Analog Devices Audio DSP with 56-bit filtering, also providing abundant processing power for advanced in-room tuning and optimized frequency response curves.

The 3000 Micro also features convenient control and custom presets via the SVS subwoofer DSP smartphone app for Apple, Android, and Amazon devices. The 3000 Micro subwoofer is priced at $799.99 and comes in Piano Gloss Black or Piano Gloss White finish.

www.svsound.com

A side-by-side comparison of the SVS 16-Ultra subwoofer and the new 3000 Micro with dual opposing 8" drivers in a compact 10" cabinet that fits anywhere

Theory Audio Design

First In-Ceiling Loudspeakers at CEDIA Expo 2021

Theory Audio Design was founded in 2018 by Paul Hales, a 30-year loudspeaker engineer, AV industry veteran, and owner of highly regarded audio manufacturer Pro Audio Technology (PRO), targeting premium loudspeaker and electronics solutions for the modern home, combining professional features with sophisticated contemporary design, at a much lower cost.

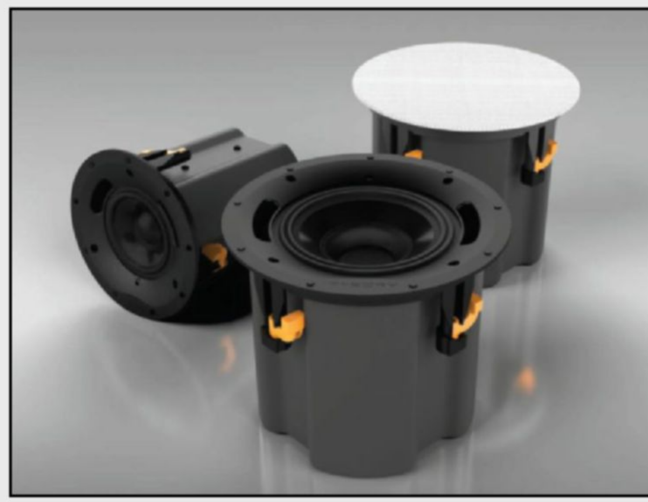
Following the successful introduction of a series of soundbars, on-wall and subwoofers models, Theory now announced the ic4 and ic6 high-output in-ceiling loudspeakers, providing dealers, integrators, architects, and designers with two premium audio options for virtually any residential or commercial installation.

Theory's product ecosystem is unique in that it offers many of the advantages of professional audio products—effortless dynamics, high efficiency, and rugged durability—but with the sonic delicacy, detail, and resolution of the best high-end high-fidelity loudspeakers. The designs are also matched with Theory's full-system design approach with automatically tailored DSP, improving the performance at a tremendous value.

As the first in-ceiling loudspeakers in the Theory product line, the ic4 and the ic6 provide installation flexibility. Discreet and highly capable solutions, the ic4 and the ic6 are ideal for residential or commercial installations, surround sound or distributed audio, and even outdoors when paired with Theory's all-weather grilles. The ic6 model can also be installed in open-ceiling and exposed environments when integrated into one of Theory's upcoming pendant housings. Available in paintable matte white or matte black, and black gloss (with matching grilles) finishes, Theory's pendant housings allow the ic6 to seamlessly blend into any décor.

Theory's new in-ceiling loudspeakers offer up to 112dB of output thanks to the advanced polymer compression drivers and carbon fiber woofers (4" in the ic4 and 6" in the ic6). For large distributed installs, the ic4 and the ic6 include a switch-selectable 70V/100v transformer, which can be driven direct by Theory's forthcoming ALM 10-channel Dante/AES67 enabled loudspeaker controllers. The estimated price for the ic4 is $500 US MSRP and the ic6 is $650 US MSRP.

www.theoryaudiodesign.com



Available worldwide from September 2021, the ic4 and the ic6 in-ceiling speakers take Theory Audio Design beyond the walls of the media room with premium architectural solutions for surround sound, distributed audio, outdoor, and commercial installations.

Perlisten Audio

THX Certified Dominus Loudspeakers

Perlisten Audio is a new audio company founded by an experienced team of industry professionals, which brought to market a full line of THX Certified loudspeakers for home theater. All the new Perlisten speakers feature state-of-the-art drivers, including beryllium tweeters, TeXtreme woofers and carbon fiber midranges with advanced waveguides, designed using advanced Comsol acoustic modeling.

Intended to be a star announcement at the CES 2021 show and any actual in-person trade show presentation, which unfortunately hasn't happened so far, Perlisten Audio is probably the greatest story in home cinema for 2021, and certainly the most important product introduction, already with a complete range of tower, bookshelf, center, surround, and monitor speakers, as well as an outstanding subwoofer.

The new Verona, WI, company was also the first audio company to include THX Certified Dominus—the highest performing THX level—in its product line, aiming to bring a truly cinematic experience

THX Certified Dominus is the highest performing THX level, aiming to bring a truly cinematic experience to the home, and bridge the gap between large home theater speakers and those used in commercial movie theaters.

to the home. Its first two products to be launched and certified are the full-range four-way S7t Tower and the D212s Subwoofer.

Dominus is the newest and the largest performance class of THX Certification, meant to bridge the gap between large home theater speakers and those used in commercial movie theaters and exhibits. Home theater owners with rooms up to $184m^3$ of space and up to a 6m viewing distance (or up to $6500ft^3$ and up to 20', respectively) can fill their entertainment space with the superior audio quality they should expect from THX Certified products. This translates to quieter whispers, bigger booms, and everything in between at a greater distance for the enjoyment of premium content in the comfort of home.

Specifically, for the THX Dominus testing, the Perlisten Audio speakers are subjected to 120dB sound pressure levels where they must keep distortion to a minimum. Dominus speakers must also be sensitive enough to reach the cinematic THX Reference Level with THX Certified Dominus, and certain THX Certified Ultra power amplifiers. The 92dB sensitivity requirement for THX Certified Dominus loudspeakers is the most sensitive of any THX loudspeaker category.

Residential Cinema

There is a reason why Perlisten Audio focuses on residential integration and home cinema applications. Its founding team includes professionals with extensive experience at some of the world's most respected audio companies, mostly acting in cinema sound and custom integration. Perlisten Audio differentiated its portfolio by implementing proprietary designs and the most advanced technologies that enhance the listening experience, bringing home theater to a level typically associated with leading movie theaters across the globe.

The Perlisten S7t Tower (SRP: $7,995/each and up, depending upon choice of finish) serves as a prime example of the company's ground-up approach, with all drivers designed by the company's US-based engineering team, and is the flagship product in the Perlisten Audio portfolio.

The design features the company's proprietary Directivity Pattern Control (DPC) array that controls the mid/high frequencies in what is essentially an MTM arrangement centered by a 28mm Beryllium dome and dual 28mm Thin Ply Carbon Diaphragm (TPCD) ultra-lightweight TeXtreme domes. All Perlisten S-Series speakers essentially share the same DPC-Array to benefit from its unique timbre matching technique. These speakers can also be configured as bass reflex or acoustic suspension to meet the demands of any listening room.

The same (TPCD) TeXtreme carbon fiber that is used in the Perlisten midranges is also the key to the performance of the S7t Tower four 7" woofers, resulting in extended dynamics and impact, with a frequency response that can play to 20Hz in a room. TeXtreme diaphragms are 30% lighter than standard carbon fiber of the same thickness while offering much increased strength and rigidity, by using a unique process of stacking multiple layers of the exclusive material composite. The unique weave is unmistakable and distributes break-up modes without sharp peaks in the response while extending bandwidth.

This speaker design is the result of 18 months of painstaking simulations using Comsol software, validated with Klippel measurements. In that unique array and waveguide, the two 28mm TPCD midranges are placed below and above the 28mm beryllium tweeter, allowing the crossover point to extend more

Perlisten Audio debuted THX Certified Dominus technology in its two first products, the flagship tower speaker, S7t, and the D212s subwoofer, with an additional eight models to be introduced to the North American market throughout 2021 and 2022.

than an octave lower than typical, and without the beaming caused by larger diaphragms. The unique waveguide "orchestrates" the directional characteristics of the array, presenting a coherent wave front, while optimizing early reflections and sound power.

The Perlisten speaker designs combine these world-class materials and transducers with unique cabinets that aren't just pretty on the outside. The attention to detail on the inside continues with a 50mm front baffle, seven different braces, strategically placed acoustic fiber to prevent internal resonances, and 1/8" of thermoacoustic damping material on all of the interior walls. The design also benefits from the finest crossover components including Mundorf capacitors.

The Perlisten S7t Tower speakers are designed to be powered by the company's 1.5kW and 3kW amplifiers, all controlled by a 32-bit ARM processor and able to run innovative diagnostics that include real-time temperature status, current and voltage monitoring, fault detection, and performance diagnostics—all evaluated at more than 1000 times a second. And the electronics are complemented with bass management to define the crossover points and behavior of the subwoofers, using a dedicated app that is able to control up to 10 subwoofers with 10 PEQs, time delay, phase, six standard EQs, 24dB low-pass crossovers, and polarity.

As the reference subwoofer in Perlisten Audio's debut selection, the D212s (SRP: $6,495 and up, depending on choice of finish) is simultaneously flexible and powerful, with a dual-woofer push-pull configuration and equal emphasis on massive power output and ultra-low distortion low-frequency reproduction.

The company employs advanced finite element analysis (FEA) to reach a high level of optimization requirements using two 12" woofers with 60mm peak linear excursion. These proprietary carbon fiber drivers produce extreme levels of bass without sacrificing total harmonic distortion (THD), resulting in bass response that is consistently accurate and powerful.

The Perlisten D212s was designed to deliver the ultra-low distortion and detail subtlety required by a THX-certified home cinema installation, and is powered by a built-in 3kW smart amp, controlled by a 32-bit ARM Cortex M4 processor, a Texas Instruments 48-bit DSP engine, and a dedicated iOS and Android app, with the ability of local direct control on a 2.4" LCD color touchscreen. It also features balanced (XLR) and unbalanced inputs and outputs.

www.textreme.com | www.thx.com | www.perlistenaudio.com

Pro Audio Technology

Three Loudspeakers in One and Modular Multichannel Amplifiers

Just in time for CEDIA 2021, Pro Audio Technology (PRO) announced a new loudspeaker model, the SR-28212ai, which the company calls its most acoustically ambitious and flexible loudspeaker ever. And amidst worldwide supply chain issues, PRO also announced the MA amplifier series to provide its integrators with a reliable and available solution for PRO and third-party systems.

The new PRO SR-28212ai speaker features a 1.7″ advanced polymer compression tweeter, two ultra-high output 8″ midrange drivers, and two 12″, 2000W woofers inside a distinctive, aimable enclosure. By leveraging the bass-management capability of PRO's ALC model amplified loudspeaker controllers, a single SR-28212ai loudspeaker system serves as three loudspeakers in one: a very high-output, very full-range loudspeaker for LCR or surround applications, an LFE (main) subwoofer, and a subwoofer for surround and Atmos speaker bass management—all at the same time, and all within a single 6″ deep enclosure.

Designed with residential surround sound systems (including DCI) in mind, the SR-28212ai can be configured with straight-firing axi-symmetric, or 40° asymmetric waveguides, which aim the sound at a 40° angle to the loudspeaker's main axis. This allows the sound to be "aimed" at the reference listening position while the speaker is mounted flush in a flat architectural wall. The SR-28212ai is also the first PRO product to utilize PRO's new Theorem Waveguide, which offers on- and off-axis dispersion of critical high frequencies.

Fully aimable using PRO's pivot brackets, the SR-28212ai provides a very wide dispersion, is incredibly efficient (>100dB/W in every passband) and offers massive output at over 130dB. The low-frequency section is 102dB sensitivity (with 4000W, 2000W AES, power handling); mid frequency is 106dB (1000W, 500W AES, power handling); and high frequency is 108dB/watt.

The SR-28212ai loudspeaker offers full-range performance, variable coverage waveguides, and built-in subwoofers, ideal for high-end home theater and video wall installations. A single SR-28212ai loudspeaker system can serve as a full-range surround loudspeaker, an auxiliary LFE system subwoofer and a surround speaker bass-management subwoofer, simultaneously.

In depth-constrained systems where the speakers cannot be aimed, the SR-28212ai can also be ordered with a 40° waveguide to aim high frequencies at 40° when mounted flat. To further increase utility, the mid-tweeter module can be inverted to bring the tweeter level down or locate it in the center of the enclosure for horizontal use, such as under or over video walls, with either a 0° or 40° waveguide.

Alternatively, an installer could invert the mid-tweeter baffle and mount the speaker upside down, placing the subwoofer section near the ceiling for use as a surround sub to augment Dolby Atmos speakers while keeping the midrange drivers and tweeters in the correct orientation and at the proper height. Extremely flexible, the SR-28212ai 40° asymmetric waveguides can be rotated in 90° increments to fire in any direction: left, right, up, or down.

New Modular Multichannel Amplifiers

Pro Audio Technology (PRO) also announced the MA-4242, the MA-9999, and the MA-9900, a new line of modular amplifiers to power any PRO or third-party audio systems.

"Earlier this year it became clear that the worldwide shortage of digital and other electronic components was going to impact our ability to deliver our amplified loudspeaker controllers timely," said Paul Hales, president and product designer for Pro Audio Technology. "Knowing we had many loyal customers with projects in the pipeline, we got to work designing a new amplifier platform that could be manufactured consistently while we wait for the supply chain to normalize. The MA series amplifiers are the result of that initiative and will allow our integrators to complete all of their open and upcoming audio system installations despite global supply chain uncertainty."

PRO's Modular Amplifiers pack four channels of up to 1000W each into a fan-less 2U chassis. When combined with PRO's ALC-3316 loudspeaker controller/processors, installers can use the MA amplifiers to quickly and easily power, apply high-resolution DSP filtering, and calibrate any PRO sound system.

www.proaudiotechnology.com.

Pro Audio Technology's new line of modular multichannel amplifiers

The Complete *Linear Audio* Library on USB

Linear Audio, the bookzine created in 2010 by Jan Didden, is now available on a USB with *The Complete Linear Audio Library*. We asked three *Linear Audio* and *audioXpress* authors to review it and share their thoughts.

Review by Gary Galo

In September 2010, Jan Didden launched a new audio publication called *Linear Audio*. Jan Didden is a familiar name to readers of *audioXpress* and its predecessors, having authored numerous articles over several decades, and in recent years serving as *audioXpress*' Technical Editor. Volume 0 totaled 169 pages, and in his opening editorial Jan described the new publication as a "bookzine." Indeed, it was far larger than a typical magazine, filling an important void between the mainstream audio periodicals (many of which have vanished), and the often highly theoretical *Journal of the Audio Engineering Society*.

The first issue was a great sign of things to come—over a dozen down-to-earth technical articles on a wide range of audio topics. The final issue of *Linear Audio* was published in April 2017. Jan noted that after 14 issues and two books, it was time to move on, and that back issues would continue to be available on the Linear Audio website. Those who were fortunate to purchase each issue amassed an invaluable technical reference library.

The entire series has now been released on a USB flash drive that also includes some very worthwhile bonus material. The articles in *Linear Audio* are of such uniformly excellent quality that singling out "highlights" is difficult. Book reviews were also a regular feature. The "extras" on the flash drive include a five-part video series by Jan about "Feedback in Audio Amplifiers," totaling more than two hours, and eight additional articles, some recent and others of considerable historical interest. These include Jan's interview with Malcolm Hawksford (originally published in *audioXpress*, November 2009), a slide show for a presentation by Jan on amplifier feedback (with contributions by Bob Cordell), John M. Miller's "Combining Positive and Negative Feedback" (*Electronics*, March 1950), another article by Miller on the origin of the "Miller Effect" (*Scientific Papers of the Bureau of Standards*, 1919-1920), H. S. Black's "Stabilized Feedback Amplifiers" (*Bell System Technical Journal*, January 1934), and 21 pages of hand-drawn notes related to feedback in audio amplifiers by Jan (uncredited, but Jan told me that he wrote these at least five years ago for a presentation; "En" in the upper right corner of each page indicates the English version).

I recommend copying the contents of the flash drive to a folder on your hard drive. This will make the files and folders easier to access, and provide a safety copy. A file called "start.pdf" loads the main index, with links to the entire contents (the *Linear Audio* issues are searchable). The PDF files for each issue of *Linear Audio* and each bonus article, along with the five MP4 video files, can also be loaded individually.

Considering the original selling price of around $30 USD per volume, the cost of the flash drive is very reasonable, especially when you consider the additional extra material. The *Linear Audio* website is still worth a visit, with several books on audio design still available, including *Baxandall and Self on Audio Power* (a collection of papers by Peter Baxandall and Douglas Self), and *High Fidelity Circuit Design* by Norman Crowhurst and George Fletcher Cooper. There are also corrections to some of the articles in *Linear Audio*, additional project documentation, and numerous free downloads related to *Linear Audio* articles (https://linearaudio.net). Anyone with a serious technical interest in audio will welcome the availability of *Linear Audio* in electronic form.

*****


ic21
CONFERENCE: OCTOBER 23-29
EXHIBITS: OCTOBER 27-29
ORLANDO, FLORIDA
“INFOCOMM IS A HIGHLIGHT IN MY YEAR. IT’S ONE OF THE ONLY EVENTS WHERE I CAN GET HANDS-ON WITH ALL THE LATEST PRODUCT AND CONNECT WITH MANUFACTURERS. IT’S AN EVENT THAT SHOULDN’T BE MISSED!”
MATT SCOTT
President
OMEGA Audio Video
infocomm
infocommshow.org

Review by Bob Cordell

I miss *Linear Audio*. Many readers are aware of the monumental effort of Jan Didden and his numerous authors in making the *Linear Audio "bookzine"* an outstanding contribution to the science and art of audio. That collection is now available to everyone on a USB stick, along with a generous dose of extra material.

As an enthusiastic subscriber to *Linear Audio* from Day One, I am quite familiar with most of the articles in the 14-volume series. It is a true collection of audio expertise. I was honored to contribute a couple of articles, and I experienced how seamlessly Jan made it to publish technical material. I have never seen such a high concentration of experts in one publication. Every serious audio designer should have this *Linear Audio* USB stick—even those who subscribed to *Linear Audio*—will enjoy the convenience and extra material included. Those who are not "serious audio designers" also owe it to themselves to get this stick to learn more about audio design. The material is an outstanding blend of educational and sophisticated audio design.

A typical issue includes subject groupings such as circuit design, loudspeakers, test and measurement, and tutorials. The latter are educational for newbies and the experienced alike. Many issues included a guest editorial by persons well-known in the field, including John Atkinson and Gary Galo. Most issues included book reviews, another asset for those who would like to delve more deeply into a subject.

Each of the 14 volumes averages about 200 pages—with no advertising! This makes for almost 3000 pages of pure expert text. A typical volume contains 10 to 12 articles, so the set includes about 140 articles created by good writers who are very experienced and know what they are talking about.

Authors include those who are well-known in the field with professional backgrounds and many gifted audio DIYers. As editor, Jan recruited such audio luminaries as Nelson Pass, Sigfried Linkwitz, Richard Burwen, Bruno Putzeys, Barrie Gilbert, and Malcolm Hawsford, to name but a few.

Linear Audio was not just a gift to the readers and the audio community. It was a gift to the authors. It provided a valuable platform for them to present material dear to their hearts in a way that did professional justice to their work.

The extras on the stick include some great archive material on audio and negative feedback and some very enlightening discussions of feedback and error correction. These are precious. The five-part video by Jan, titled "Masterclass—Feedback in Audio Amplifiers," is a treat to watch, and very educational. I watched it from start to finish. As one who has delved into the depths and discussions of negative feedback over the years I can tell you than Jan really nailed it, especially for those less experienced in applying negative feedback. It was a pleasure to watch, and Jan is one of the few who can provide great, intuitive explanations of some of the more difficult and controversial aspects of negative feedback. But he didn't stop there. He gave an insightful overview of various forms of error correction in power amplifiers. Jan also provided one of the most elegant and balanced explanations of Quad's current dumping scheme that I have seen.

In a related supplemental piece, he re-published a great interview with Malkolm Hawksford, developer of the Hawksford error correction scheme frequently referred to as HEC. There is something in these extras for everyone, from newbie to expert.

If you love audio electronics and related subject matter, you will not be sorry you got this convenient collection of top-notch material.

****

Review by Dimitri Danyuk

A few weeks ago I received a thumb drive from my friend Jan Didden. This particular USB stick contained a bookzine collection—all 14 volumes of *Linear Audio*. In case you missed it, *Linear Audio* was compiled and published by Jan Didden from 2011 to 2017 with contributions from many names in the industry, big and small.

I have all the hardcopies of *Linear Audio* in my library, but I immediately grabbed my laptop, plugged in the USB drive, and opened the content.

It felt like a journey back in time to my college years in the late 1970s, when I would visit the university library every month to ask for the newest issue of *Wireless World* magazine. I still remember the excitement of turning the crisp pages full of valuable and innovative technical information. I particularly remember the audio sections; audio was flourishing back then, before it was absorbed by the computer industry. Browsing the contents of *Linear Audio* in its digital format gave me the same excitement.

This new format has none of the limitations of the printed copies—the graphics and diagrams are clear and easy to zoom in on, I can search for specific information, and I don't need to lug around 14 volumes everywhere I go. The immense wealth of experience, knowledge, and expertise in the audio field was contained right at my fingertips.

The contents of the volumes are extensive, containing information on loudspeaker design, digital audio, solid-state and hollow-state audio, microphones, phonograph records reproduction, and more. The articles within are accessible to a wide audience—some are targeted toward entry-level DIYers, some require very solid knowledge of electronics, some contain a little too much math for comfort, and others, well, others may have been pulled from the trash bin. Nevertheless, I can't think of another comparable source of information when it comes to audio technology. The Publisher provides a refreshing balance of articles written by both well-known experts in the field and garage audio enthusiasts, ensuring that anyone who is interested in audio technology will find something useful. This unique collection of audio-related articles closes the gap between *audioXpress* and the preprints of the Audio Engineering Society conventions papers. It looks as though the *Linear Audio* was inspired by *Wireless World* and *audioXpress*.

Unfortunately, there is a sad but logical interruption in the publishing of *Linear Audio*. Recent developments in Class-D technology—amplifiers with digital audio input and digital post-filter feedback (e.g., from Axign) and GaN output devices—leave almost no room for the audio enthusiast to improve upon the electronics. Analog audio has entered the state of being a fashion industry. Even so, I commend and appreciate Jan Didden for his herculean effort to encourage authors and manage a publishing house in order to produce all these volumes of *Linear Audio*; I cannot imagine the vast amount of work it required. I have thoroughly enjoyed turning (and now scrolling!) through its pages, and will always insist that anyone interested or professionally involved in audio technology should own this collection.

The Complete Linear Audio Library. $99 USD Available at: https://diyaudiostore.com/collections/books/products/the-complete-linear-audio-library

A Tale of Speakers in a Room

Richard Honeycutt shows us why understanding the strengths and weaknesses of different speaker designs and modeling/comparing the best candidates will ultimately lead to the best results for any space.

By
Richard Honeycutt

It was the best of speakers, it was the worst of speakers. The year was 1969. Central Auditorium* had recently opened for community events, and a sound system had been included in the construction. Since most events were expected to be speeches, plays, and meetings, the design goal for the system was to make the sound loud enough for all listeners.

The room being 36′ wide and 80′ long not counting the stage area, the architect recommended using 14 small speakers spaced evenly along the walls beside the audience area, as illustrated in **Photo 1**. Each speaker is represented by a blue square with an × inside. The two long olive-green boxes represent the seating areas. The wide areas at the top represent lower and upper stage areas. The wide area at the bottom is the balcony seating. The architect did not give any thought to supporting the balcony audience with sound reinforcement. The speakers he chose were the type commonly used in school classrooms in that era (**Photo 2**). As shown in **Photo 3**, they did achieve his limited goal.

As years passed, the uses of the venue changed, as did the tastes of the patrons. These changes revealed the deficiencies of the original speakers. Increasing use of recorded tracks to accompany singers and choirs revealed the lack of bass response. Presenters of speeches, and facilitators of meetings, becoming accustomed to good sound reinforcement systems in most venues, could no longer be counted upon to practice good vocal projection, so limited speech intelligibility in the auditorium began to generate more complaints.

Sound Problems

Audiences who were exposed to excellent auditoriums gradually noticed that the sound at

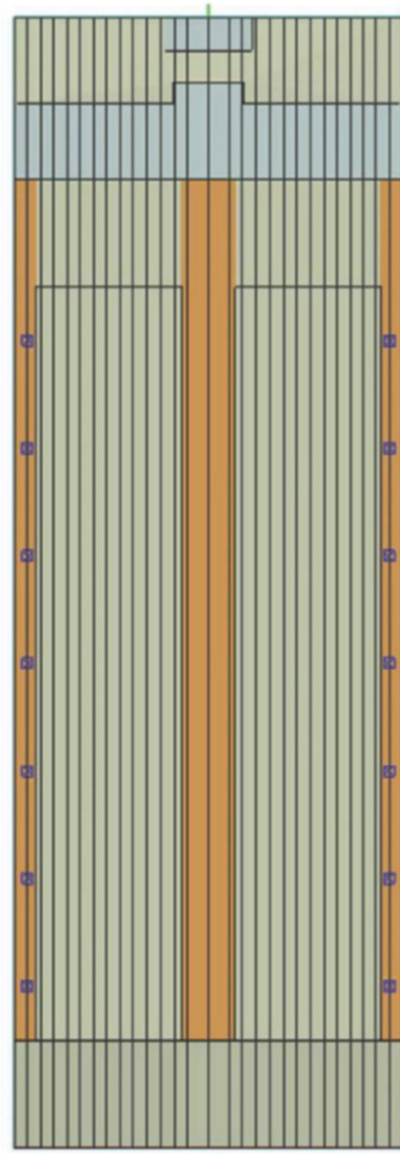


Photo 1: This plan view of Central Auditorium shows the positions of the original 14 speakers.

Photo 2: The architect chose sloped-front wall-mount speakers.

Central Auditorium did not seem to originate at the person talking or singing, or at the musician on stage; rather, the sound came from all around. As the demands of program material and the sophistication of audiences increased, the auditorium manager came to understand that improvement was needed. Wisely, the manager persuaded the board of directors to add a technical director to the staff in order to provide competent advice on getting the most out of the sound system.

Jim the tech director brought with him an understanding of the improvements in sound system technology, including measurement methods and system requirements for producing wide-frequency-range sound reinforcement with good intelligibility and proper acoustical imaging. He joined the staff in the mid-1980s. It took him about 20 years of carefully educating the board of directors before he finally convinced them of the need to upgrade the speaker system. When they finally assented, he considered hiring a consultant for guidance, but eventually chose not to do so because of cost considerations. Applying his own education and experience, he set out to select a speaker system that would perform as needed:

- The frequency response would be reasonably flat from about 60Hz to 15kHz or better.
- The speaker location would be such that the acoustical source location would correspond as nearly as possible in the horizontal plane with the location of the sound source.
- A direct-sound SPL of 85dB (flat-weighted, slow response) would be available at all seats without driving the speaker into clipping.
- The SPL variation from the best seat to the worst seat would be under 6dB.
- Most of the sound would be directed to the seating areas, with little "spill" onto the side walls.
- The sound reaching the stage from the speaker system would be minimized in order to minimize feedback squeal. (Performers would be able to hear themselves acoustically or through dedicated stage monitor speakers.)

In order to set a standard for comparison, Jim measured the performance of the current system. He found that its frequency response was limited to 120Hz to 8,000Hz. Maximum SPL without clipping was over 90dB, but with sounds having strong energy below 200Hz, the steel speaker cabinets or their mounting hardware caused annoying buzzes and rattles. The SPL on the side walls was actually over a decibel higher than in the best seats, indicating excessive "spill." This, of course, was the factor responsible for the poor acoustic imaging mentioned earlier.

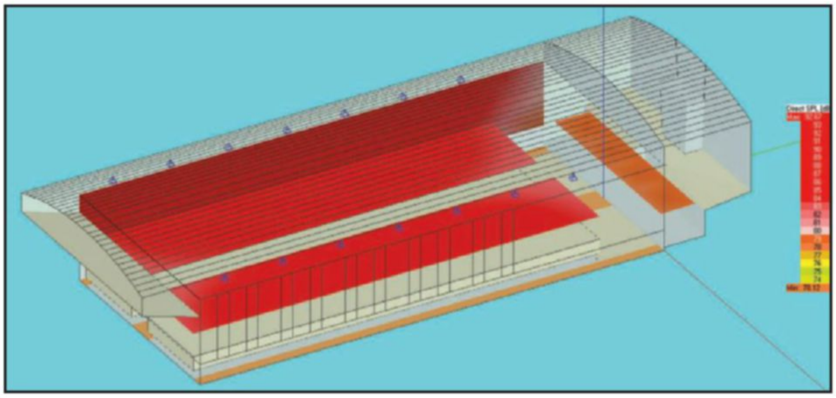


Photo 3: The wall speakers provided even coverage, as shown in this map of direct SPL on the seats and walls.

Evaluating Viable Options

As Jim was methodically evaluating options, it happened that the auditorium was rented for a celebration including a contemporary music concert and a speech by a nationally prominent politician. Knowing that the existing system could not meet the needs for this celebration, he rented a portable speaker system consisting of two "speakers on sticks" (stand-mounted speakers), and placed them 2' in from the side wall on each side of the auditorium front as indicated in **Photo 4**.

While these did not provide the evenness of coverage that the original system did, they did produce the required frequency response and SPL without buzzing or rattling. Subjective evaluation of the speech intelligibility indicated that the rented speakers were essentially equivalent to the original speakers in this respect. Fortunately for Jim, the board members were impressed with the "miracle" he had wrought using the rented speakers, and over his objections, bought a pair of similar speakers that graced the front of Central Auditorium for

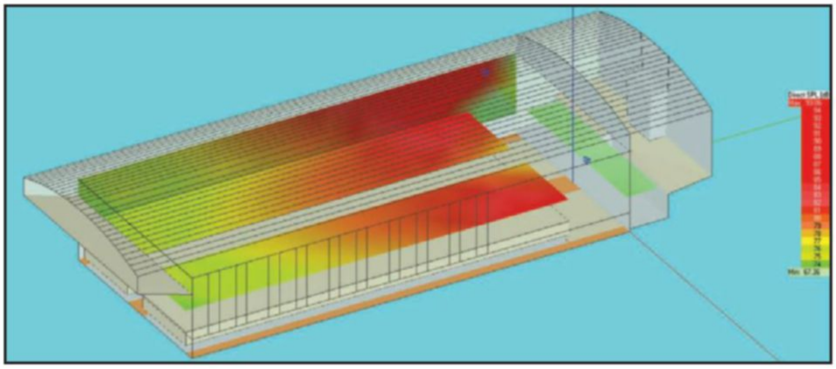


Photo 4: The "speakers on sticks" provided less-even coverage than the original system.

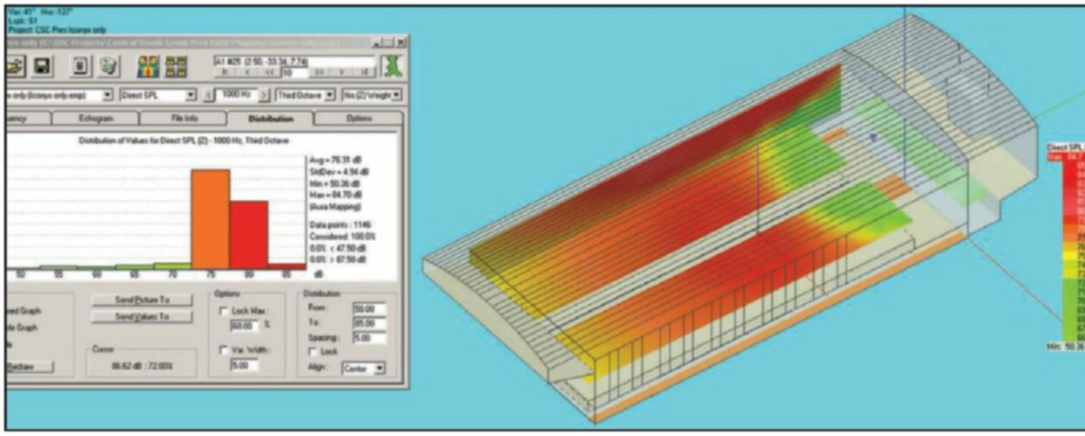


Photo 5: The steerable line array did not produce as even a coverage as the "speakers on sticks."

almost a decade. The original speakers were left in place "just in case."

New Challenges

During that time, the local music club used the auditorium from time to time for music recitals, including one memorable event featuring African percussion instruments. It was then that the issues of reverberation and echo came to everyone's attention. Each drum beat sounded like two or more, and the precise rhythms of the percussionists were turned into a jumble. Another demand was placed upon the beleaguered tech director when a children's theater group was formed.

Although by that time Jim had added wireless microphones to the auditorium' system, no one could devise a way to make sure the little darlings actually wore their mics or kept them turned on, or did not tangle the cables in their costumes and pull the mics out of place. And even when the cherubs' mics were working properly, a certain number of them could be counted upon to contract stage fright and whisper their lines too softly for the best mics to pick up. Thus, Jim was faced with a need for serious improvement in speech intelligibility.

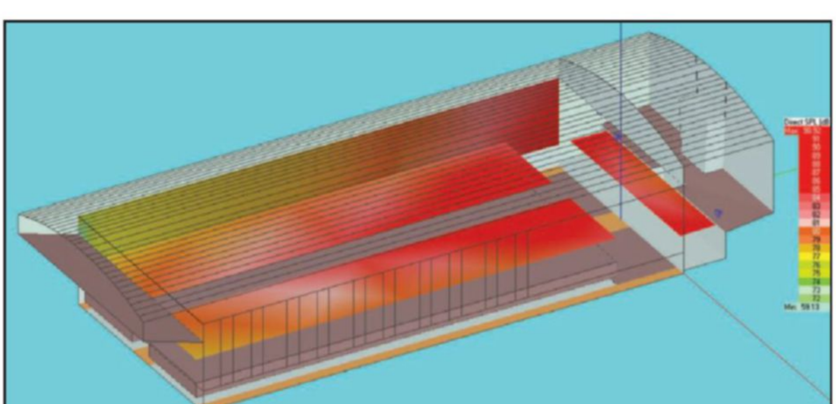


Photo 6: The horn with front-fills provided almost as even a coverage as the "speakers on sticks."

This time, Jim faced his limitations and advocated for bringing in Ron, a consultant in acoustics and sound system design. Ron's first step was to measure the performance of the room, the original speaker system, and the "speakers on sticks". He found that the reverberation time (RT) at 500Hz was 2.8 seconds when the room was unoccupied. Of course, RT dropped when the auditorium was occupied, but upon interviewing the auditorium manager, Ron found out that fully two-thirds of the events held in the room were scantily attended, so he used the unoccupied RT as his standard for design.

Despite numerous claims over the years that the directional control provided by some speakers could reduce RT, Ron had found little evidence of such claims in computer-modeling sound systems. Thus it did not come as a surprise that the RT did not vary whether measured using an acoustical source, the original speakers, or the speakers on sticks.

The Speech Transmission Index (STI) with the original speakers ranged from 0.52 to 0.56 (toward the upper end of the "fair" range), depending upon the measurement location. Echo speech as determined by computer modeling was actually better using the original speakers: a value of 0.5 was obtained. The speakers on sticks created more echo—the computed value was 0.7.

Ron immediately knew that the RT problem required acoustical treatment, and he specified 2 1/8" fiberglass to be added to the rear wall (under the balcony face), which would decrease slap echo from speakers located in the front of the room, as required for proper acoustical imaging. The wall needed to be treated from 3′ above the floor to the top. Treatment below 3′ would do little good, because direct sound from the speakers to that part of the wall would be blocked by the seats. Modeling revealed that more treatment was needed to tame the reverberation, so he also specified

fiberglass for the front 26′ of each side wall. This treatment brought the RT to his target value of 1.6 seconds. This RT target was set for current conditions in which attendance at most events is very low. However, to make sure the venue would not be too dry when treated with this absorption, Ron modeled it when 50% occupied, and the RT came out as 1.3 seconds. Central Auditorium is considered a multi-purpose venue, and 1.3 seconds is appropriate for such a venue having a volume of 69,000 ft^3.

Design Objectives

Meeting the design objectives listed earlier requires careful speaker selection. The frequency response of most speakers designed for live sound is flat enough that they will meet the first criterion. The second criterion, placement for proper acoustical imaging, requires that the speaker be placed front and center. This implies a maximum speaker-to-listener distance of about 82′, requiring the speaker to produce an average SPL of 85dB at that distance. Adding 20dB for peaks, we need the speaker to output at least 105dB at 82′ without clipping or thermal damage. The remaining three criteria relate to speaker type, location, and aiming.

As discussed in recent Sound Control articles, either horn or line array speakers could be chosen to provide directional control. Line arrays have the advantage that their SPL does not drop off as quickly with distance, but horns are better at keeping sound spill off the walls. Common line arrays can also cause spill onto the stage.

Ron was also concerned about good speech intelligibility in the auditorium. Keeping RT within the proper range would help a lot, but echo control is also important. Excess background noise degrades intelligibility, but usually this is an HVAC issue, and Ron's measurements showed that Central Auditorium did not have problems in this area.

Ron modeled three speakers for the auditorium: a steerable single-box line array, a horn speaker with lower fill speakers in the front corners, and a Constant Beamwidth Technology (CBT) line array column from JBL Professional. As you can see from the SPL coverage maps shown in **Photos 5–7**, the CBT line array provided the most even coverage: the SPL variation from seat to seat was within +3.3dB. However, other factors did not allow a clear winner in the "speaker shootout." **Table 1** shows the Echo Speech and STI along with the evenness of coverage, wall spill, and stage spill of the three modeled speakers, compared with those performance metrics for the "speakers on sticks."

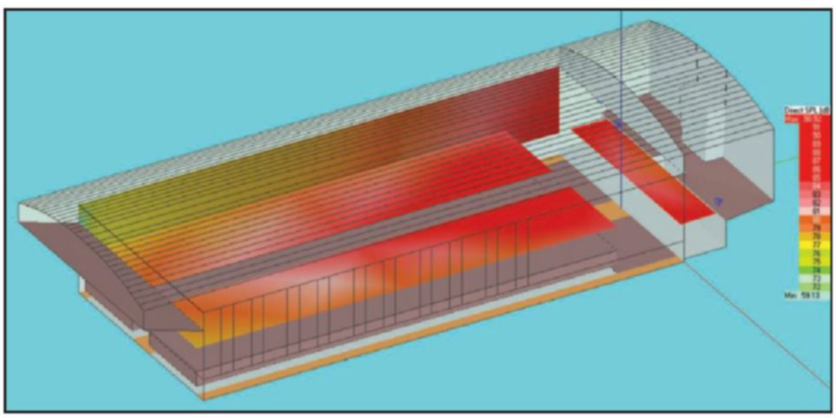


Photo 7: The CBT line array provided slightly more even coverage than the "speakers on sticks."

Speech Intelligibility is currently measured by determining Speech Transmission Index (STI). The numerical value of STI correlates with subjective intelligibility as follows:

Echo speech is a rating performed according to the method introduced by L. Dietsch and W. Kraak. The numerical value corresponds to the subjective annoyance of the echo: higher values indicate more annoyance. Values below 0.8 are acceptable; values >1 are considered objectionable.

Humans with normal bilateral hearing can acoustically localize a sound source horizontally within about 1 degree. Vertical localization is poorer: more like 5 degrees. This means that in order to avoid the distracting effect of the sound seeming to come from the wrong place, horizontal alignment of the speaker with the source is more important than vertical alignment. Thus, high speaker placements are often chosen. However, if the speaker is too high, it can create an unpleasant "voice of God" effect, or even a synthetic echo due to the sound from the speaker arriving at the listener's ears more than 50ms after the direct sound.

The CBT Speaker

The CBT speaker clearly shows the most even coverage, although the other two speakers under consideration would have benefitted from rear-fill speakers fed by delayed signals; in fact the benefit would have allowed them to surpass the CBT speaker in evenness. However, the venue manager vetoed the use of rear-fills for aesthetic reasons. The horn with front-fills provided the best intelligibility and the least wall spill. Why does wall spill matter? In wide rooms, sound hitting the side walls can cause annoying echoes. However, that is not a concern in narrow rooms such as this one. But sound arriving at the listener's ears laterally adds to the "apparent source width." This can

Model	Evenness	Echo Speech	STI	Wall Spill	Stage Spill
Sticks	+6dB	<0.7	0.62 to 0.77	+4dB	-19.9dB
Steerable line	+9dB	<0.6	0.65 to 0.79	+1dB	-26.7dB
Horn/front-fills	+6.6dB	<0.6	0.89 to 0.96	-10.6dB	-18.5dB
CBT	+3.3dB	<0.7	0.54 to 0.61	0dB	-17.9dB

Table 1: The Echo Speech and STI along with the evenness of coverage, wall spill, and stage spill of the three modeled speakers is compared with those performance metrics for the "speakers on sticks."

be advantageous for choirs and orchestras, but it can be annoying for lecturers, actors, vocal or instrumental soloists, or small musical groups. A Wagnerian soprano with a 36′ wide mouth is a frightening thing to contemplate. So Ron specified the horn with front-fills as the best speaker system choice for Central Auditorium.

It is important to note that the optimum choice of speakers is highly dependent upon the shape of the room. A wide room that is short from front to back often calls for some sort of line arrays. Long, narrow rooms such as this one usually benefit from horns mounted high up in front. Sometimes a central cluster is best, or in other rooms, a single horn with front-fill speakers mounted fairly close to the floor works better, and often long, narrow rooms benefit from rear-fill speakers in what is called an "exploded array." So copying the speaker design that works in a different room is not a good idea. Understanding the strengths and weaknesses of different speaker designs and modeling/comparing the best candidates will ultimately lead to the best results.

**Author's Note:* *The name Central Auditorium is fictitious—this article is based upon a combination of venues on whose sound systems the author has consulted.*

About the Author

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.

Ultra-Linear: Why It Works And When It Will Not (Part 1)

The Fundamental Questions

This article reviews the story and the principles of ultra-linear electronic circuits, looking in detail at several tube types. This article is the first in a two-part series in which Morgan Jones not only shows how ultra-linear works, but more importantly, predicts when it won't.

By
Morgan Jones

First, there was the triode, but although it was low distortion, it was inefficient (even by thermionic standards) because the anode couldn't swing close to 0V before grid current caused severe distortion. By contrast, the pentode could swing close to 0V without grid current and this (very roughly) doubled the power an output stage could deliver to the loudspeaker, which was a good thing. But the pentode produced unpleasant distortion compared to the triode, so engineers looked to combine the virtues of the triode (low distortion, low r_a) with the efficiency of the pentode. The prolific British inventor Alan Blumlein patented his distributed loading solution in 1937, but nobody took much notice (World War II was imminent), and it took time before the world could once again afford such frivolities as audio, so when David Hafler and Herbert I. Keroes in the US used exactly the same idea in 1951 and called it "ultra-linear," the technique stuck even though the name was lacking. But despite considerable speculation at the time, nobody knew why the technique worked...

This article not only shows how ultra-linear works, but more importantly, predicts when it won't. In Part 1, we will review the basic tenets of ultra-linear operation and introduce the detailed operation with several tube types. In Part 2, we will formulate some pertinent questions and answer them.

Pentodes

Assuming that the reader has a passing acquaintance with thermionic technology, we begin by investigating some slightly less familiar characteristics of the pentode and its close cousin, the beam tetrode.

Pentode output characteristics plot I_{g2}—in addition to the main anode curves—show that at low anode voltages I_{g2} is catastrophically high, falling to safe normality once anode voltage exceeds g_2 (**Figure 1**).

Although useful, the curves shown in Figure 1 are often taken to mean that I_{g2} becomes low and constant, rather

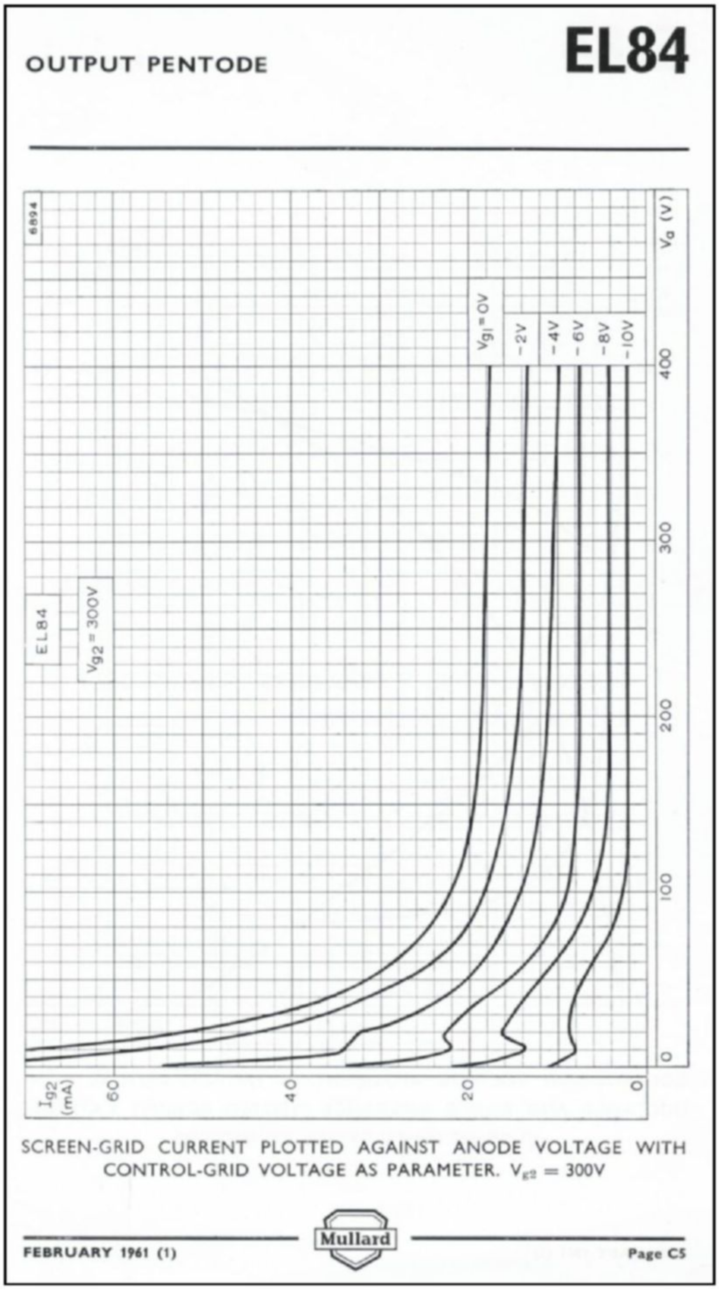


Figure 1: I_{g2} curves for Mullard EL84

than being a near-constant proportion of I_a. A useful pentode specification is its I_a/I_{g2} ratio, akin to h_{FE} for the junction transistor, and for exactly the same reason—the base or g_2 is designed to control/accelerate electrons to a final collection point (collector, anode) but a proportion of the electron flow is lost to the interposed control electrode, making it useful to quantify the defect (**Figure 2**).

A logarithmic current scale was used for I_a to avoid concealment of aberrations at low currents, but the general trend is clear; I_a/I_{g2} ratio is reasonably constant with I_a (arguably more constant than the defect h_{FE}).

Beam Tetrodes and Aligned Grids

Electrons thermally jostled from the hot cathode surface have random direction, but although nothing dissipates this original low-velocity component, it is negligible compared to the velocity imparted by accelerating electrodes. Thus, whether attracted by an anode or a screen grid, electrons leaving the control grid region take the shortest path to their destination, so adjacent paths are virtually parallel. If g_1 and g_2 grid wires have the same pitch and are vertically aligned, they cause the electron stream to divide into thin sheets, and because g_2 lies in the shadow of g_1, it intercepts minimal current.

This alignment principle can be applied to any multi-grid valve, but is especially popular for beam

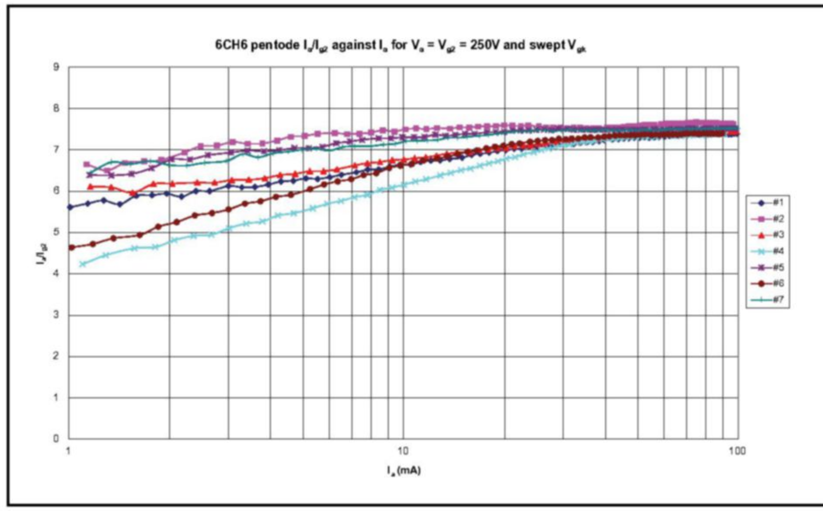


Figure 2: Measured I_a/I_{g2} ratio against I_a for seven 6CH6 pentodes. All curve tracer measurements were taken using Ronald Dekker's excellent pulse mode µTracer3+, with the raw data exported to a spreadsheet for further processing.

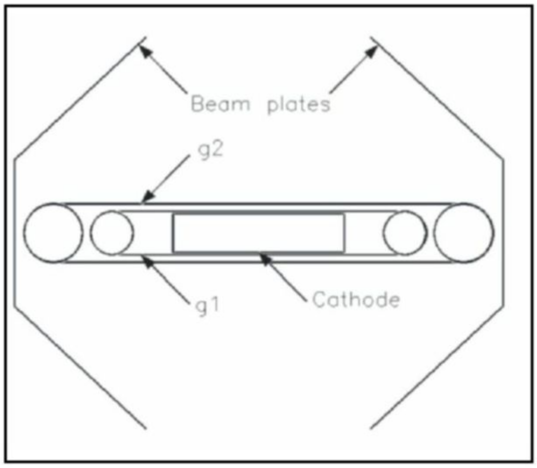


Figure 3: The beam tetrode cross-section is viewed down the cathode axis (enclosing anode not shown). Note that the cathode grid structure is planar and matched to the beam plates whereas a non-beam device could use a circular cathode/g_1/g_2 structure and would avoid localized heating of the anode.

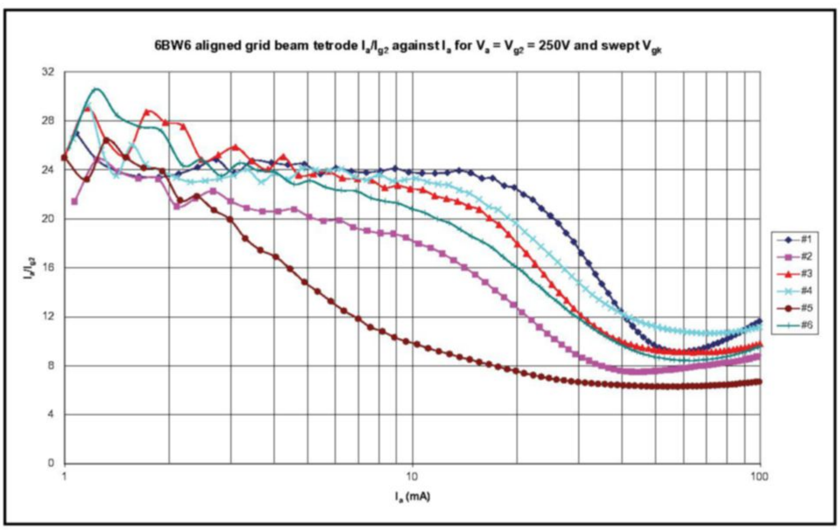


Figure 4: Measured I_a/I_{g2} ratio against I_a for six 6BW6 aligned grid beam tetrodes. Note the wide variation in transition current between high and low I_a/I_{g2} ratio.

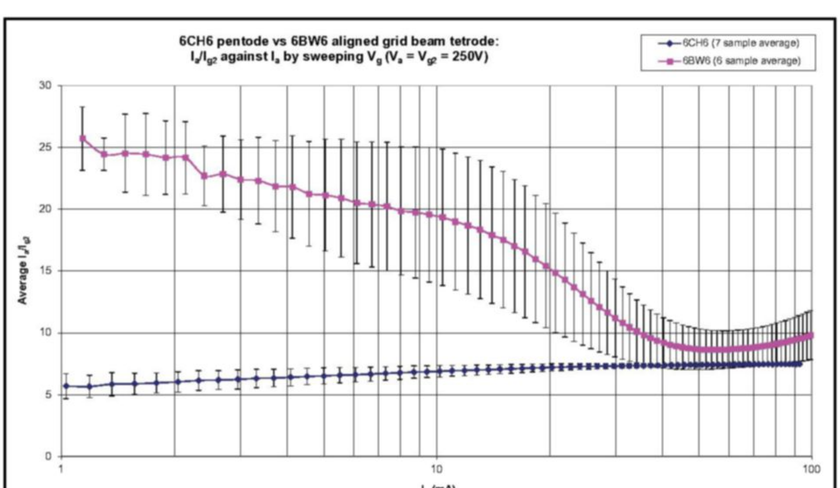


Figure 5: The aligned grid beam tetrode has demonstrably superior I_a/I_{g2} ratio at low currents but significant variation with I_a. The large 6BW6 error bars are due to the sample variability of 6BW6 transition current seen in Figure 4.

tetrodes because not only does reducing g_2 current increase device efficiency, but dividing the stream into multiple sheets further increases electron density, assisting the return of secondary electrons to the anode. A further enhancement encloses a planar cathode/grid structure within a pair of beam forming plates to concentrate the previously omnidirectional stream into a pair of diametrically opposite high-density electron beams composed of multiple sheets (**Figure 3**).

Unfortunately, the I_a/I_{g2} ratio of aligned grid valves falls as current increases because increasing current requires more electrons within the same volume, and like charges repel, so the thin sheets thicken, gradually intercepting g_2 wires, and increasing g_2 current. Worse, although machines can repeatably wind g_1 and g_2 grid wires to the same pitch, vertical alignment of the two grids is set by hand assembly, and as misalignment increases, the thickening electron sheets are intercepted at progressively lower currents. If we define transition current as that occurring at the average of initial (high) I_a/I_{g2} ratio and final (low) I_a/I_{g2} ratio, we observe that this current is determined by the quality of grid alignment, with perfect alignment achieving the highest transition current (**Figure 4**).

Taking an average of multiple samples allows the 6CH6 to be directly compared with the 6BW6 (**Figure 5**).

The aligned grid beam tetrode has superior I_a/I_{g2} ratio at low currents, and this was a valuable property because reducing I_{g2} significantly reduced quiescent dissipation when in Class B power stages. Having measured general purpose valves, it was time to measure an audio valve, so 16 JJ Electronic EL84 were measured and averaged (**Figure 6**).

The 6CH6 pentode had an I_a/I_{g2} ratio that varied from 5.68 at 1mA to 7.47 at 93mA (an increase of 31%), but the much smaller variation of the EL84 pentode was more usefully plotted as a percentage. This reduced variation of I_a/I_{g2} ratio will become significant later.

Electrical characteristics having been determined, some worn-out devices were sacrificed to science and dissected, except for the 6CH6 for which external inspection was sufficient to confirm pentode construction and dissimilar grid pitches. Fully assembled, it is difficult to check grid alignment, and anode removal easily disturbs remaining alignments, but aligned grids require identical pitches, so turns were counted over a measured grid length using the vernier scale of a travelling microscope, allowing average grid pitch to be accurately determined.

Table 1 shows that all but one of the measured beam tetrodes had aligned grids, but none of the pentodes. Note that despite published explanations of beam tetrode operation invariably including aligned grids (and that beautiful RCA pen and ink sketch), alignment is not mandatory. The Mazda 30FL1 is demonstrably a beam tetrode, works perfectly well, yet does not have aligned grids.

Ultra-Linear

If we imagine the output transformer primary as a set of windings that could be tapped at any point, we see that for pentode operation g_2 would be connected to the center tap (0%); whereas strapped as a triode, it would be connected to the anode (100%) as shown in **Figure 7**.

What would happen if we were to tap at an intermediate point? This question was asked in 1951 by David Hafler and Herbert I. Keroes [1], and an amplifier they dubbed ultra-linear became synonymous with an output stage that had been patented by Alan Blumlein [2] in 1937 (thereby pre-dating the 1951 Keroes patent [3]).

Blumlein's 1937 distributed loading patent had noted that the optimum g_2 tap was between

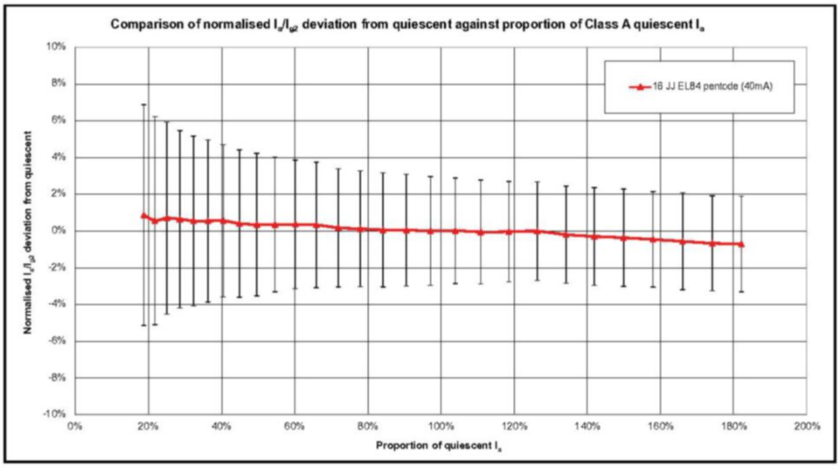
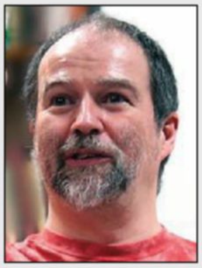


Figure 6: JJ Electronic EL84 has much lower variation of I_a/I_{g2} ratio.

About the Author



Morgan Jones began professional life as a broadcast engineer at the BBC. Upon leaving, he wrote *Valve Amplifiers*, first published in 1995 and now on its fourth edition. For a day job, he taught Broadcast Engineering, but now (semi-retired) he designs low-noise test and measurement equipment and thinks a nanoamp is a large current. When not in the lab, he enjoys making accurate swarf.

Valve	Type	g_1	g_2	g_3	Uncertainty (1Φ)
Osram KT66	Aligned grid beam tetrode	0.890	0.892	–	±0.1%
GEC KT88*	Aligned grid beam tetrode	0.701	0.703	–	±0.14%
Brimar 6BW6	Aligned grid beam tetrode	0.594	0.593	–	±0.15%
Mazda 30FL1	Beam tetrode	0.154	0.424	–	±0.3%
Marconi N78	Pentode	0.301	0.859	1.696	±0.1%
Mullard EL84	Pentode	0.386	0.880	2.980	±0.15%
Mullard EL34	Pentode	0.569	1.003	3.480	±0.16%
*KT88 anode came away without internal disturbance allowing g_1 and g_2 alignment to be directly confirmed.					

Table 1: All pitches in millimeter (mm) per turn

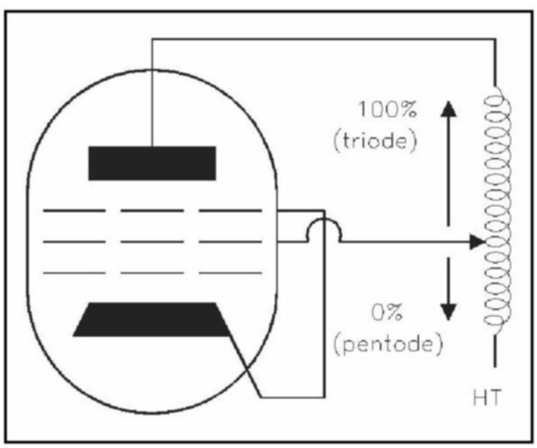


Figure 7: Ultra-linear connection via a variable tapping

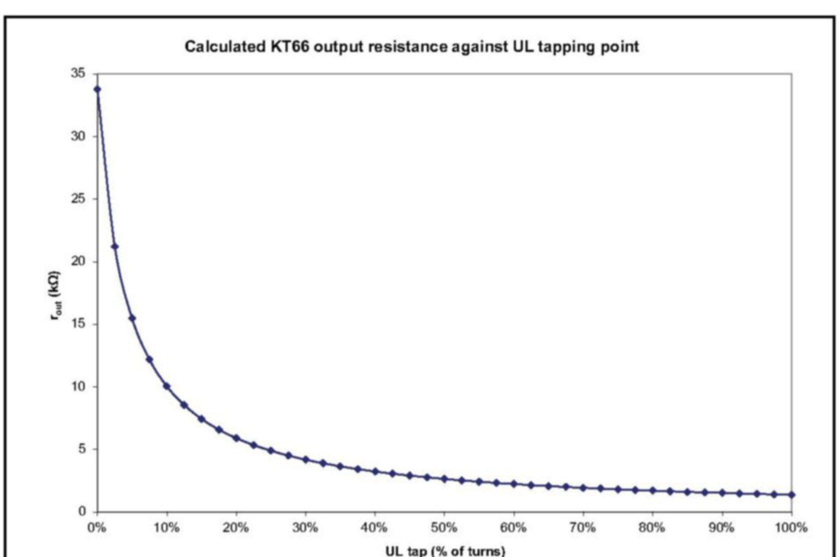


Figure 8: KT66 anode output resistance calculated using the D.T.N. Williamson and P.J. Walker equation

25% and 50% of anode voltage, but in 1955, Fritz Langford-Smith and A. R. Chesterman [4] reported detailed investigations using the KT66 that defined tapping ratios in terms of impedance and declared that 20% gave lowest distortion. Outside Australia, tapping ratios expressed in terms of primary turns (voltage) were preferred, so an undated Keroes promotional pamphlet [5] referring to the KT66 declared minimum distortion at 40-43% turns ratio (equivalent to approximately 20% impedance), and GEC KT66 datasheets from 1960 onward stated 40%, although they neglected to qualify that number.

By contrast, Mullard datasheets for EL34 (January 1960) and EL84 (January 1961) explicitly defined tapping ratios in terms of primary turns, unequivocally stating that 43% gave minimum distortion and 20% maximum power for both their valves. In 2000, Stephen Spicer [6] noted that Harold Joseph Leak (founder of H J Leak & Co., Ltd.) reduced turns tapping ratios in 1961 for the company's EL84 and EL34 amplifiers from 49% to 25% but cautioned that his tapping measurements were subject to ±2% experimental error, and this author's 300Hz measurement having a reduced uncertainty (1Φ = ±0.18%) modified Spicer's 49% value to 50%, which is not only consistent with Spicer's transformer construction diagrams, but seems a more likely value.

Ultra-linear output stages became almost universal in the final days of thermionic supremacy because they combined the efficiency and low input capacitance of the pentode with the linearity and low output resistance of the triode such that they measured very well.

However, the only published analysis at the time capable of predicting a result reasonably matched by measurement was that by D.T.N. Williamson and P.J. Walker [7] who usefully showed that comparing open-circuit devices made the effect on output resistance calculable, and that even a smidgen of feedback to g_2 significantly reduced output resistance:

$$r_{out} = \frac{\dfrac{\mu}{g_m\left(1+x.\dfrac{\mu}{m}\right)}}{1+x.\dfrac{\mu}{1+x\dfrac{\mu}{m}}}$$

where:
r_{out} = output resistance
μ = tetrode amplification factor (μtetrode)
g_m = mutual conductance
x = fraction of output voltage fed to screen
m = triode amplification factor (μtriode)

The author used his μTracer3+ to measure a bogey NOS GEC KT66 at the Quad II operating point of $V_a = V_{g2} = 314V$, and forced $I_a = 62mA$ by setting $V_{gk} = -24V$, obtaining values of; $\mu_{tetrode} = 189$, $g_m = 5.6mA/V$, and $\mu_{triode} = 8$. Having obtained typical device parameters, the equation could be investigated numerically (**Figure 8**).

The graph begins at tetrode mode (0%) and ends at triode (100%), but we see that even a 20% turns tap (x = 0.2) reduces r_{out} to 5.9kΣ from the tetrode value of 34.5kΣ, even if it doesn't reach the 1.4kΣ of triode, and a 40% turns tap gives 3.2kΣ.

The early investigators had limited test equipment and no digital computers, so it is little wonder that they concluded that empiricism was the way forward, and it was perhaps the success of the Williamson amplifier that set their investigative path onto the KT66 beam tetrode without first confronting the physics of aligned grid valves. This author wanted firm answers to the following questions:

- Does g_2 contribute power to the load, or is it purely negative feedback?
- What does the g_2 current waveform look like?
- How do we ensure that ultra-linear works well?

We will address these questions in Part 2 of this article and come to some interesting conclusions.

Project Files

To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html

References

[1] D. Hafler, H.I. Keroes, "An Ultra-Linear Amplifier," *Audio Engineering*, November 1951, pp. 15-17.

[2] A. Blumlein, "Improvements in or relating to Thermionic Valve Amplifying Circuits," British Patent # 496,883 (1937).

[3] H.I. Keroes, "Audio Transformer," US Patent # 2,680,218 (1950).

[4] F. Langford-Smith and A.R. Chesterman, "Ultra Linear Amplifiers," *Radiotronics*, Volume 20, May 1955, pp. 53-61.

[5] "Theory and operation of the Ultra-Linear circuit," Keroes Enterprises, Undated, but post-August 1955.

[6] S. Spicer, *Firsts in High Fidelity*, Audio Amateur Press, 2001, pp. 251.

[7] D.T.N. Williamson and P.J. Walker, "Amplifiers and Superlatives." *Wireless World*, September 1952, pp. 357-361 (also published as "Amplifiers and Superlatives" in the *Journal of the Audio Engineering Society*, April 1954, Volume 2, #2, pp. 75-80)

Cascodes, Folded Cascodes, and Current Mirrors (Part 1)

Cascodes and Folded Cascodes

By
Morty Tarr

Cascodes, folded cascodes, and current mirrors are circuit topologies that are used to provide specific functions for an amplifier design. These topologies improve the performance of the devices, but also add complications. In this first article, the author discusses cascodes and folded cascodes.

As designers, we have many devices from which to choose. Some devices are ideally suited to certain tasks. Other tasks require performance that is difficult to achieve. While many excellent devices exist today, ideal devices do not exist, except maybe in Spice simulations.

Cascodes, folded cascodes, and current mirrors are circuit topologies that are used to provide specific functions for an amplifier design. These topologies improve the performance of the devices, but also add complications to the design.

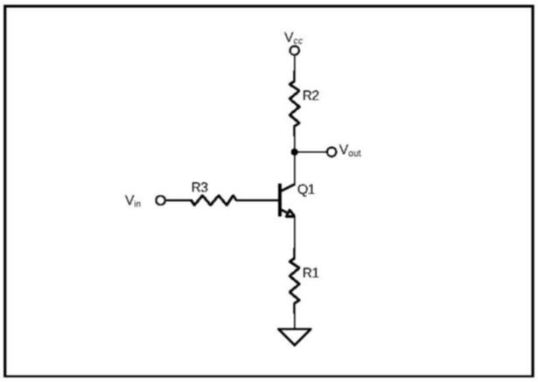


Figure 1: This schematic shows a basic NPN voltage gain stage.

Why Would We Need a Cascode?

Figure 1 shows a basic NPN voltage gain stage. Transistor Q1 is operating in common emitter mode, where the base terminal is the input, the emitter is connected to ground, and the output is taken from the collector. The gain is equal to R2/R1 (inverting) where R1 includes the emitter resistance of Q1. Since the emitter resistance of Q1 varies with current, it is important that R1 be larger than the ~26Ω/mA resistance of the emitter.

There are several factors that limit the bandwidth and linearity of the amplifier in Figure 1. In a common emitter gain stage, the capacitance between the base and the collector terminals of the transistor feeds the output signal (the collector voltage) back to the base (the input terminal). As the frequency increases, the impedance of the base-collector capacitance decreases. This contributes to the frequency response roll-off at high frequencies, especially since we like high input impedance for our gain stages. The collector-base junction capacitance varies with collector-base voltage. Higher voltages correspond to lower capacitance. This is the Miller Effect or Miller Capacitance.

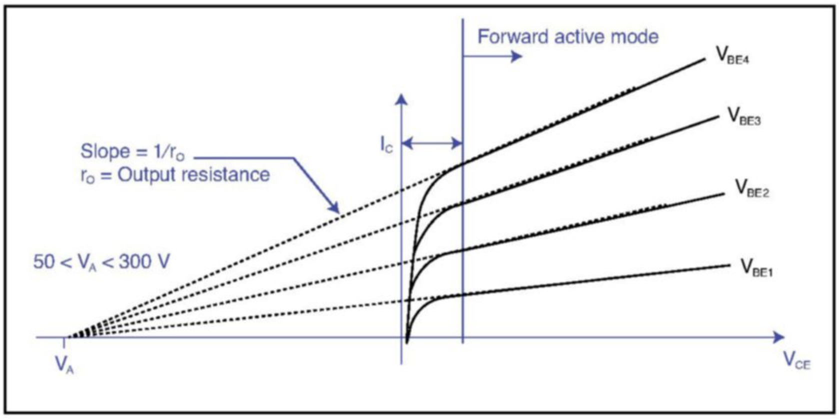


Figure 2: As the collector voltage of a transistor increases, the base width becomes smaller and the gain increases. (Original graph by Analog Devices)

Transistors also exhibit a phenomenon named after James Early, who discovered it. In a bipolar transistor, the width of the base region is modulated by the base-collector voltage. The width of the base affects the gain (β, or H_{FE}) of the device. Smaller base widths correspond to higher gain. As the collector voltage of a transistor increases, the base width becomes smaller and the gain increases. This is illustrated in **Figure 2**.

The Early Voltage (VAF in Spice) is an extrapolation of the constant V_{BE} curves to a point on the horizontal axis. Larger (more negative) values indicate a smaller Early Effect.

If the effective gain of the transistor can change with signal level, the stage will create distortion. A typical goal of 0.01% distortion is 1 part in 10,000. This can be easily exceeded by the Miller Effect and Early Effect. These effects can be mitigated via feedback, cascodes, or a combination of the two.

Cascodes are not new. Cascodes were and are used in tube circuits, particularly in high-frequency designs. In this article, we'll focus on solid-state devices.

BJT Cascodes

Let's start with a basic bipolar junction transistor (BJT) cascode. **Figure 3** shows a basic cascode circuit. Q1 is the input transistor and Q2 is the cascode transistor. The voltage gain is given by R2/R1 and is inverting just like a single transistor gain stage.

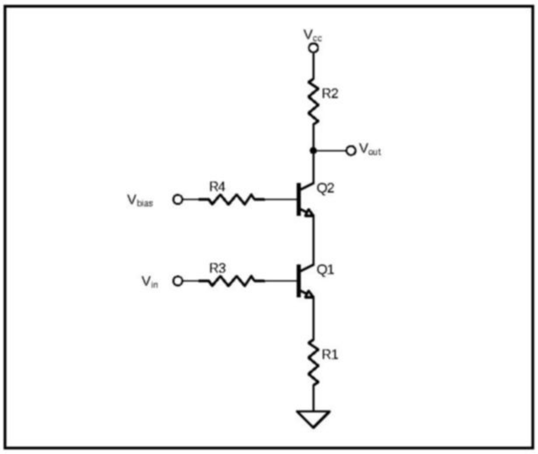


Figure 3: Here is a schematic for a basic cascode circuit.

Gain transistor Q1 is operating in common emitter mode, with the base as its input. This is where our signal is applied. Cascode transistor Q2 is operating in common base mode (the base is connected to a DC bias voltage), its input is the emitter terminal.

Here's how it works. As voltage V_{in} increases, the current in Q1 increases. Since the only place this current can go is through Q2, the current in Q2 increases. V_{BE} of both Q1 and Q2 also increases due to the higher current, so the bias at the collector of Q1 changes, but only slightly. The current in Q2 is then impressed on R2, rising current driving voltage V_{out} toward ground. As V_{in} decreases, the voltage at V_{out} increases in a similar but opposite manner. This is pretty much the same as if we didn't include Q2 and R4.

Adding Q2 provides several advantages. With Q2 in place, the collector of Q1 is held at an almost constant voltage. This minimizes the effect of the base-collector capacitance (the Miller Effect) thereby increasing the bandwidth of the gain stage. Holding Q1C at a constant voltage also minimizes Q1's Early Effect.

As an example, if we bias the base of Q1 in Figure 3 at a constant DC voltage, and increase the voltage at the base of Q2, we are increasing the collector voltage of Q1. As we do this, the gain of Q1 will increase with a corresponding effect on the collector current of Q1, even though we did not change the DC bias voltage V_{in}.

This is not an effect we want. If the gain of the stage changes with signal voltage, distortions will result. We'd like the gain of the stage to be independent of the signal, at least until we get close to clipping.

The cascode transistor Q2 prevents the collector of Q1 from moving more than a few tens of millivolts, thereby minimizing the Early Effect as well and the feedback capacitance. Note that there is no requirement for Q1 and Q2 to be the same type of device. Q1 can be a low voltage high β device, and Q2 can be a high voltage, lower β device. The β of Q2 determines only how much base current it draws through R4. With a cascode, R4 is almost always required to reduce sensitivity to oscillation. Usually 100Ω to 1kΩ is sufficient.

Devices designed for higher voltage have thicker bases, therefore, they have lower β. They also have a higher voltage at which the Early Effect becomes significant. Unfortunately, the Early Effect is not mentioned in device datasheets. Nor is the Early voltage, which is extrapolated from the slope of the constant V_{BE} curves of the device. For an amplifier stage, a higher Early Voltage is desirable, as this will minimize the gain changes with signal level.

We set up the cascode so the gain transistor, Q1, has a nearly ideal collector-emitter voltage. This is generally in the 5V to 10V range. To do this, we set the DC voltage V_{bias} at about 10V higher than the DC component of V_{in}. The base-emitter junction of Q2 will drop about 650mV, so we will have 10V − 0.65V = 9.35V across Q1. A higher voltage across Q1 makes it more linear, but then we can run into power dissipation issues if the current is significant. Q2 can be a high voltage transistor (e.g., the KSC2690 or the KSC3503). The larger packages also allow significantly more voltage across the device without running into power dissipation limits.

Please don't try to use a cascode without base resistors. The high gain and high-frequency response are very likely to cause oscillation. Also, you should not put a decoupling capacitor at the base of Q2; that is about the same as omitting R4. In practice, I test to make R4 as small as I can without oscillation and then use double that value. Generally, R4 = 220Ω is a good starting point.

Note that in the cascode configuration, Q2 provides a current gain of just under 1.00. In other words, the current at R2 is the same as the current at R1 (less the base currents of Q1 and Q2).

Although we have illustrated these effects with NPN bipolar transistors, they apply to PNP transistors as well. MOSFETs also have many similarities. In a MOSFET, the drain-gate capacitance is the feedback capacitance, C_{rss}. The equivalent of the Early Effect is the channel length modulation. So we can't really get away from these issues with today's technology. A cascode stage with N-Channel MOSFETs is shown in **Figure 4**.

MOSFET Cascodes

A MOSFET cascode has much lower bias current as the gates of the field-effect transistors (FETs) draw orders of magnitude less current than the bases of bipolar transistors. Also, the gate-source voltage of most MOSFETs is in the 4V to 5V range, where the base-emitter voltage of a BJT is in the 600mV to 800mV range. This means that the difference between V_{bias} and V_{in} has to be 4V to 5V higher than in the BJT case.

Although it may not be obvious in the case of MOSFETs, "interesting" things happen if we let the drain voltage approach the gate voltage. These effects include significant changes in node capacitances and stored charge as V_{DG} approaches zero volts. These effects can be mitigated in switching circuits, but not so much in linear designs. So keep the drain voltage higher than the gate voltage. I find a 5V difference to be a good minimum, with 10V ideal, but often hard to achieve.

Folded Cascodes

A cascode increases the bandwidth of a gain stage and its inherent linearity. It would be good to be able to have these advantages, and also reverse

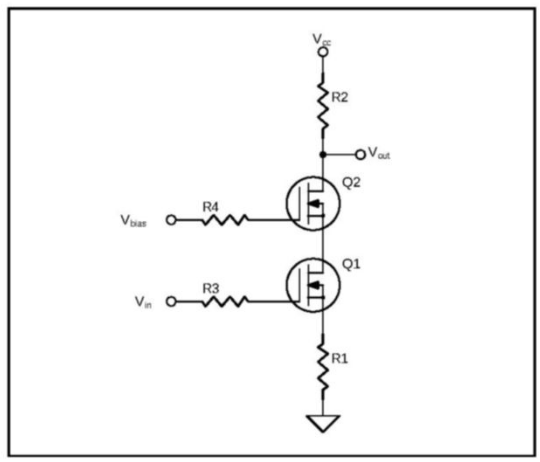


Figure 4: A cascode stage with N-Channel MOSFETs is shown.

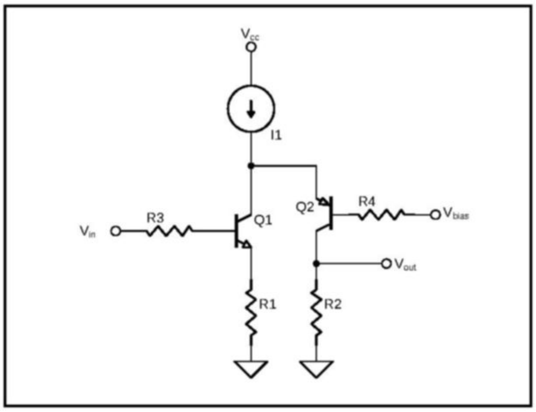


Figure 5: A BJT folded cascode is shown here.

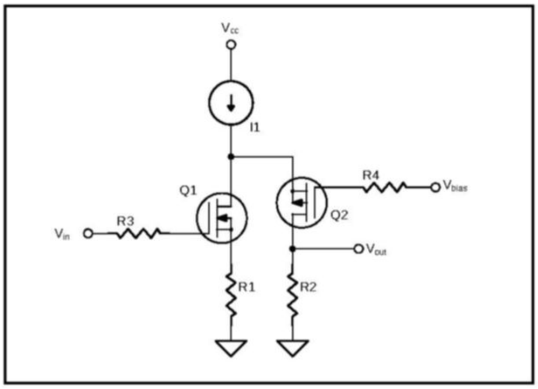


Figure 6: This is a schematic for a folded cascode that is very similar to the BJT example.

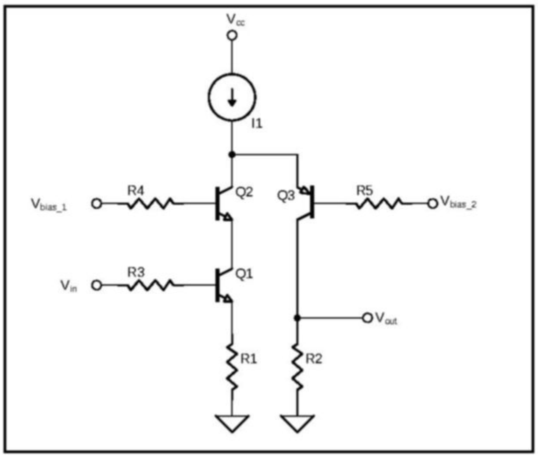


Figure 7: Here we have combined a basic cascode and a folded cascode.

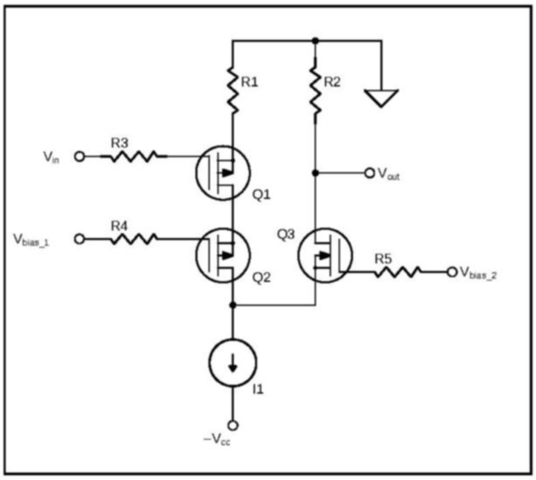


Figure 8: Input transistor Q1 is a PMOS device.

the polarity to bring the signal level back down to ground, or near ground. A folded cascode can do this. **Figure 5** shows a BJT folded cascode.

In a folded cascode, the cascode transistor is the opposite polarity of the gain transistor. In this example, Q1 is the input (an NPN transistor) and Q2 is the folded cascode (a PNP transistor). Current source I1 provides the operating current for Q1 and Q2. The current in Q2 is the current supplied by I1 less the current consumed by Q1. As V_{in} rises, the current in Q1 increases, so the current in Q2 decreases and V_{out} also decreases. The stage is inverting, just as the basic cascode.

Note that the voltage at the emitter of Q2 will change with the current in Q2. As the current in Q2 increases, the emitter voltage of Q2 will also increase as V_{BE} of Q2 increases, so if we're using a resistor for the current source, there will be small changes in the total current. For precision, I recommend a current source rather than a resistor for I1. The change in current in Q2 is equal to the change in current in Q1, so the gain of the circuit is still R2/R1 (inverting).

The emitter of Q2 is again kept at a reasonably constant voltage, so the feedback capacitance of Q1 has minimal impact on the frequency response. Here Q2 must be a different device, although it could be what we call a complementary device. In reality, NPN and PNP transistors (and N-MOS and P-MOS FETs) are not truly complementary, but that is a topic for another article.

In a basic cascode, the DC current in Q1 and Q2 is the same; in a folded cascode the DC current in Q1 and Q2 can be different. However, the AC current in Q1 and Q2 will be the same, as the current changes in Q1 are the same magnitude but opposite polarity of the current changes in Q2. Cascodes do not provide current gain. We can make a folded cascode with MOSFETs as well.

Figure 6 shows a folded cascode that is very similar to the BJT example. V_{GS} of the MOSFETs is higher than V_{BE} of the transistors, so we have to accommodate that in the biasing of the devices. Q1 is an N-Channel MOSFET and Q2 is a P-Channel MOSFET. We do not want either Q1 or Q2 to saturate as that would be really bad for linearity.

We can combine a basic cascode and a folded cascode as shown in **Figure 7**. We would do something like this if we were making an amplifier where we needed a large output swing, but we wanted to use a lower voltage device for input transistor Q1. Q2 allows Q1 to function at a lower (and almost constant) collector voltage while the output at Q3 can be a much larger voltage. Q3 would ideally be a device with a high Early Voltage to minimize the effects of β changes. An example of these requirements is the input stage of an audio power amplifier.

The input transistor is Q1, cascoded by Q2 whose emitter sets the collector voltage for Q1. The emitter voltage of Q3 sets the collector voltage of Q2, and again the voltage at Q3's emitter will change with the current conducted by Q3.

P-Channel Cascodes

Cascodes are not limited to N-type devices; a P-Channel cascode is also possible. For the schematic in **Figure 8**, input transistor Q1 is a PMOS device. It is cascoded by Q2, also a PMOS device. The cascode is then folded by Q3, an NMOS device.

Note that in each of these examples, we could have a BJT or MOSFET cascode with a JFET input, or a BJT cascode with a MOSFET input, or any combination that makes sense for the design requirements. We are not constrained to use the same or even similar devices (as some silicon designers are constrained by integrated circuit process parameters). For example, in Figure 8, Q2 could be a PNP transistor and Q3 could be an NPN transistor.

Next Month

In Part 2 of this article series, we will discuss current mirrors and line stages built with cascodes, folded cascodes, and current mirrors.

Author's Note: I would like to thank John Curl for generously sharing his time, knowledge and insight.

About the Author

Morty Tarr has more than 30 years' experience in Electrical Engineering. Morty graduated from Cornell University in 1972 with a Bachelor of Arts in Physics, and a minor in Electrical Engineering. He continued his studies at Tufts University in Electrical Engineering.

Morty began his career at recording studios in NYC and Boston. Then he started to be interested in the design of electronic equipment and made the transition from audio to electronic design.

He has extensive experience in the fields of video, audio, test and measurement, networking, and wireless (primarily Wi-Fi and Bluetooth). More recently, from 2006 to 2015, Morty led a team at Bose responsible for advanced development for a $2 billion business. He set the direction for the group and for technology strategy spanning wireless, digital, analog, and system architecture across multiple product lines. Prior to Bose, he worked for LTX developing GHz test systems, at Octave Communications developing conference call bridges, and at Avid Technology. Prior to Avid, at Data Translation, he created the first 24-bit resolution A/D for a PC for Chromatography applications, and then led the team that developed the first Media 100 video editing system.

Morty currently has 20 patents and has spent most of his professional career developing first-in-class products. Innovative ideas and new concepts come naturally to him.

You Can DIY!

Refurbishing Altec Lansing Vintage Speakers

By
Bruce Heran

Vintage Altec Lansing speakers continue to be in high-demand all over the world and driver manufacturers continue to keep replacement parts in stock, while requests for even the most obscure models persist. This article showcases how refurbishing these highly coveted speakers continues to be a rewarding experience.

There is one key element to a good sound system—speakers. As much as they can cause problems, they are something most of us cannot live without. Folks who use headphones exclusively can tune out now. Nothing that follows is likely to be of much interest. There is a bit of mystique about speakers. If you ask 10 people why they like one speaker over another you will probably get 10 different answers. I have several sets of speakers. All are excellent and all sound different from each other. While I like the accuracy and enveloping sound field nature of the MartinLogan ESLs I have, I find the sound of vintage high-end conventional speakers alluring as well.

This article is about refurbishing a pair of vintage Altec Lansing "Magnificents." I use the term refurbish in the sense that they are restored to functionality and especially made to recreate the sound that the original ones had. This concept always brings up the controversy of accuracy vs. musicality. However, I don't intend to plunge into that debate in this article.

The Magnificents

A while back I scored a pair of vintage Altec Lansing Magnificent enclosures at a thrift store (**Photo 1**). They are beasts in the best tradition of the 1950s and 1960s. I believe the ones I have were made about 1967, but records are inconclusive from that era. If you are interested in Altec Lansing history there are some sites that cover it. The best one I have found is www.lansingheritage.org.

The Magnificents are one of numerous variations that trace their origins to the really huge Voice of the Theater (VOT) speakers that were found in many theaters. They were highly efficient and capable of substantial levels of sound. The Magnificents are a smaller home entertainment variation. They are still quite large at 7.3 ft^3 internal volume and able to produce a great deal of sound. I have seen some in modest-sized theaters as they look nice and are a lot easier to move around than the VOTs. I added casters to mine as I regularly shift things around in my listening area. They save my back and muscles greatly.

Back to the enclosures, they came without the speakers (**Photo 2**). This is fairly common as the original speaker components command premium prices all by themselves. They are also much easier to ship. Huge wooden boxes that weigh nearly 300 pounds each are costly to ship. For the longest time the enclosures just sat around and begged to be used.

The lack of original drivers is problematic as the originals are scarce and pretty much priced out of reach for the average builder. The two woofers, the two tweeters with their waveguides, and a pair of crossovers can easily cost in the $3000 per speaker range and sometimes more if they are in good condition. I recently saw a set for sale at $4500. Ones of lesser quality can be refurbished, but the costs are similar in the end.

Finding the Right Speakers

Fortunately for me, I found a source of newly manufactured woofers. Great Plains Audio (https://greatplainsaudio.com) both refurbishes original speakers and builds new Altec Lansing components. They are made to meet the original specifications. I also scored a pair of the woofers from a DIYer who didn't use them and just wanted to cut his losses. They were in perfect condition and about half the going price at the time. Shipping was not cheap as they weigh about 25 pounds each. Unfortunately since then the new replacements have become nearly as costly as the originals.

Anyone who finds a pair of the enclosures would probably do well to seek alternative drivers. The Magnificents being two-way systems still needed crossovers, tweeters, and waveguides. The original waveguides were large multicellular horns. These are quite hard to find in any reasonable condition at any reasonable price. Again fortune smiled on me. A fellow audio enthusiast was familiar with the woofers and the original horn tweeters and was able to provide assistance.

To back up slightly, the woofers are 38cm in diameter (**Photo 3**), and unusually for that size, respond very smoothly to nearly 4000Hz. Based on their size, you would expect them to respond only to perhaps 800Hz to 1000Hz. Being skeptical, I measured them. They actually responded as I was told. They have a high sensitivity of about 94dB/W and better yet, they have a rather benign resonance peak. As you would expect, they are a good match for the enclosures. I am not suggesting that anyone try to get 4000Hz response from them as no crossover would work well at that frequency. But it is possible to get to nearly 2000Hz with modest slope crossovers and not run into any problems.

Obtaining suitable matching tweeters is not as simple as it would seem. First, there are many from which to choose. Second, features such as sensitivity, useful frequency response, resonances, flat bandwidth, and even the physical type and dimensions complicate the issue. With assistance of the earlier mentioned DIYer, I settled on a Radian 475PB high-frequency horn driver mounted on SEOS 15 waveguides. The combination has an effective lower response point of around 800Hz. This would make for a really nice match to the woofers. Unfortunately, the Radians have a nasty resonance peak at about 1100Hz. They are also about 18dB

Photo 1: I found a pair of vintage Altec Lansing Magnificent enclosures at a thrift store and decided to restore them to their original glory.

Photo 2: The enclosures came without the speakers.

Photo 3: Here is one of the Altec Lansing enclosures with only the 15" woofer.

more sensitive than the woofers. All this impacts the type and slope of the crossover.

It turns out that I am rather sensitive to phase anomalies that occur between drivers at crossover points. So any crossover had to have essentially no phase issues. The best match for a passive crossover was a symmetrical second-order Butterworth one.

The frequency was determined by what best can be called trial and error. With my handy audio test microphone and computer, I tried numerous values. The final one was 1725Hz with a 14dB pad on the tweeter as well as an 8Ω "L" pad to allow for fine adjustments.

This frequency was far enough from the quirks of the tweeters and well within the flattest portion of the woofer response. All well and good, but I usually bi-amp my speakers.

The Altecs are just one of the two main speaker systems I use. The others are MartinLogan ESLs. These systems are about as different as you can get. One is mellow old school and one is bi-polar new school. Their differences mandate my use of an electronic crossover. Determining the crossover frequency for the electronic crossover was similar to the passive one. Test, measure, and listen to find the best settings. The electronic crossover has a 24 dB/octave slope and has zero phase shifting and a zero sum at the selected frequency. After much trial and error again, it turns out that the best crossover frequency was at 1600Hz. This was an enlightening discovery and clearly demonstrates that there can be multiple solutions to the same issue.

In many ways I am a number-oriented individual. Precision and accuracy are things I like to deal

Photo 4: I added extra strengthening to the cabinet.

Photo 5: Here is the inside of one of the Altec Lansing enclosures with both drivers in place.

Photo 6: I added a cross brace, which has a provision for adjusting the tension it applies to the rear of the woofer.

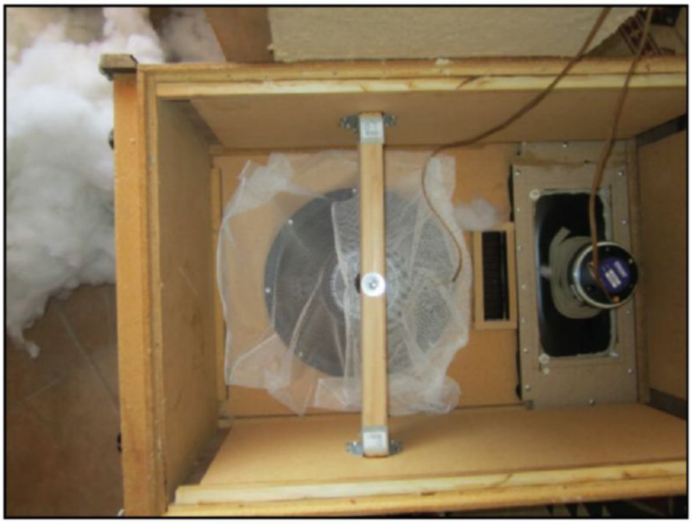


Photo 7: I put a net over the woofer to keep any pieces of the wool from encountering the woofer cone.

with. However when it comes to speakers, my ears have the final say. I like a sound that I can best describe as something that reminds me of an actual performance. This is not to say that any system can really duplicate one, but I like one that captures some of the essence of a live performance. It is what I call an experience that tickles the hairs on the back of your neck and makes you say wow. The Magnificents do that. The specifications are far different from the accuracy of the MartinLogan ESLs, but the variations don't seem to matter.

Refurbishing the Boxes

Now, getting to the task of refurbishing the speakers. Nothing is easy when you work with heavy wooden boxes the size of small refrigerators. It gets worse when you add additional weight to them. The original enclosures are pretty stout with double-walled sides made of what seems to be the equivalent of 3/4" MDF.

Unfortunately, the large surface areas are still capable of some cabinet resonances. To remedy this, I reinforced the front baffle that was only a single layer of wood and added internal bracing to the other sides. The main internal cross brace I added has a provision for adjusting the tension it applies to the rear of the woofer (**Photos 4–6**). This keeps the woofer from moving in and out and tends to damp the cabinet at the same time. I adjusted it for approximately equal tension between the two enclosures. The nuts were adjusted so the brace tensioner was just touching the speaker and then three turns of the nut tighter. Your value will depend on the thread spacing of the tensioner (a 1/4 × 20 bolt in my case). It ought to apply enough tension to insure that the movement of the speaker will not unload the tensioner completely. The original fiber damping was wool and I reused it. I put a net over the woofer to keep any pieces of the wool from encountering the woofer cone (**Photo 7**).

I needed to fabricate a mounting board for the SEOS waveguide as it was about half the size of the original multicellular horn. I wrapped some plumber's putty tape around the waveguide as there was the possibility of resonances. It doesn't need a lot; a few ounces will work. More is fine though.

I calculated the size of the cabinet vent for smoothest response at the woofer resonance point. I used the LDC7 program (*Loudspeaker Design Cookbook*, 7th Edition, by Vance Dickason ISBN1-882580-47-8) to determine the value (**Figure 1**). I constructed it out of pine boards and fastened it to an existing port hole in the front baffle.

Apparently there is a slight difference with what Altec Lansing determined to be the length of the vent

Speaker T/S Parameter and Box Performance

Input Parameters						
		Speaker 1	W8-740P	HIF17JVX	Fostex	Dayton
		notes	8 inch	notes	FE126EN	15
Sd	m^2	0.122	0.22	0.129	0.065	0.805
Re	Ohms	3.2	3.2	6.5	7.2	5.2
Fs	Hz.	67.3	28	29	83	20.8
Qts		0.24	0.3	0.45	0.3	0.65
Qms		3.85	10.53	2.48	4.8	12.08
Vas	L	17.3	22.9	64.8	8.5	311
Xmax	mm	4	12	3	0.35	14.3
Power (RMS)	Watts	40	150	30	15	600
Sealed Box Size	ft^3	0.75	0.4	1	0.25	7
Fill% (Fiberglass)		0	25	0	0	0
Vented Box Size	ft^3	1	0.4	1	0.25	7
Vent Diameter	in	3	2	2	1.5	4
Extended Parameters		Calculated Parameters				
Qes		0.256	0.309	0.550	0.320	0.687
Cms	m/N	0.0000082	0.0000033	0.0000276	0.0000142	0.0000034
Mms	kg	0.680	9.646	1.093	0.258	17.234
Mmd	kg	0.6315	9.530	1.040	0.239	16.417
BL	Tm	59.94	132.61	48.52	55.04	130.57
no	%	1.99%	0.16%	0.28%	1.46%	0.39%
SPL @ 1W 1M	dB	95.04	84.01	86.48	93.71	88.00
SPL @ 2.83V	dB	99.02	88.00	87.39	94.18	89.87
Sealed		Sealed Box Performance				
Vb (Box Volume)	ft^3	0.75	0.4	1	0.25	7
Alpha		0.82	2.02	2.29	1.20	1.57
Qtc		0.32	0.52	0.82	0.45	1.04
Fc (Box Resonance)	Hz	90.67	48.68	52.60	123.15	33.34
F3	Hz	251.52	71.26	46.49	225.27	25.76
Power (Pd)	Watts	-4512.7	3366.7	152.4	-4.5	14840.9
Max SPL	dB	#NUM!	108.8	104.3	#NUM!	118.8
Vented		Vented Box Performance				
Vb (Box Voluem)	ft^3	1	0.4	1	0.25	7
F3	Hz	54.20	38.18	41.75	89.98	25.37
Fb (Box Tuning)	Hz	57.78	34.83	37.49	87.86	23.92
Ripple	dB	-2.50	-0.26	1.41	-0.87	2.36
Power (Pd)	Watts	310.3	28300.2	492.8	7.0	41907.0
Max SPL	dB	114.1	108.8	104.3	102.1	118.8
Vent Length	in	3.5	16.0	4.6	1.4	5.5

Figure 1: Here is a ample set of computations from the *Loudspeaker Design Cookbook*, Version 7 software.

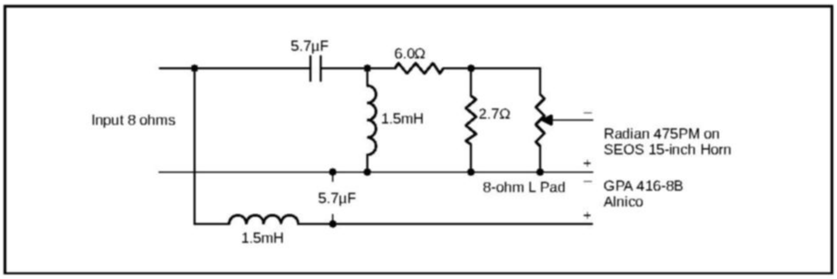


Figure 2: This is the crossover schematic.

Parts List

Audyn Caps Plus Capacitors—4.7µf and 1.0µf in parallel to obtain 5.7µf

Jantzen 18-gauge air core inductors—1.5H (DCR = 0.32Ω)

SEOS 15 horn waveguides

Radian 475PB high-frequency compression driver

Great Plains Audio GPA 416-8B—15" (38cm) 8Ω alnico woofer

Input and output connectors to match your system

8Ω L Pad 50W

Photo 8: I used wooden "card file" boxes I purchased from a department store to house the components.

Photo 9: This is the back of the crossover. The volume and port dimensions are the most critical.

and what I determined. Altec's number was just the thickness of the panel and mine had the same square area but was about 2" in total length. I figure that the original woofers most likely had a slight difference in the resonance parameter from the new ones. I had to use new longer wood screws to hold the back of the cabinet in place. Over time, several of the original screws had stripped out. The originals were 1.5" in length number 8 size. I replaced them with 2" number 10 size ones. There are around 25 of them and the back is quite secure without glue. I did seal the perimeter with plumber's non-hardening caulk rope.

Next, I cleaned up the wood surface. Over the years it had numerous blemishes and scratches. I used Howard Walnut Restore-A-Finish followed by Howard Feed-N-Wax. Together they make the cabinets look really nice. The cabinets have the original grill cloths. They were in good shape except for a section of one that had badly faded. I used a small paint brush to apply several coats of diluted dark brown RIT dye to the section. It is now essentially invisible unless you know it is there.

Passive Crossover

The next challenge was to build the passive crossover (**Figure 2**). Since I frequently bi-amplify the speakers I wanted it to be external. I used wooden "card file" boxes I purchased from a department store to house the components (**Photo 8**). Depending on your choice of active or passive crossover you could build them inside the cabinets. Just be sure to fasten them down well as the woofers will shake anything not secured well loose.

For folks who are well-versed in woodworking and want to recreate their own cabinets, the internal dimensions are approximately 22" × 16" × 36" (56cm × 40cm × 90cm). This yields an effective volume of about 7.3 ft^3 (206 liters). The area of the vent is 14in^2 × by 2" tall (35cm^2 × 5cm tall). A slight deviation in dimensions is unlikely to matter, but I would not alter any of them more than about 10%. The volume and port dimensions are the most critical (**Photo 9**).

About the Author

Bruce Heran lives in Sierra Vista, AZ, and is an avid DIYer. He has been involved in electronics for 59 years. He built his first project—a one-transistor amplifier—when he was 13. He has a BS in Biology and studied electrical engineering for two years. Electronics has always been his hobby. In May 2011, Bruce retired from his work as a project manager of a logistics contract with the Federal government. Now he is co-owner of Oddwatt Audio, an electronics company that specializes in vacuum-tube audio components. He is vice president of design and support, and all of the company's products are kits and assembled audio equipment based on his personal designs. He is an avid collector of vinyl records and vintage turntables.

Resources

V. Dickason, *Loudspeaker Design Cookbook*, Version 7, Audio Amateur Press, 2006, available at https://cc-webshop.com/products/loudspeaker-design-cookbook

Great Plains Audio, https://greatplainsaudio.com

Lansing Heritage, www.lansingheritage.org

If you use alternate speakers, I would employ a program that can compute the various parameters for you. I use the LDC7 program for such computations. It has an extensive array of tools to use for such situations. If you have trouble sleeping, read the textbook that comes with it. It boggled my mind. Passive crossover components can be a headache. I recommend that you not scrimp on the cost or quality of them as they can make or break the end result.

The Sound Test

How do the speakers sound? In short great. My system is all tube-based with the exception of the electronic crossover. That is one device that is far easier and likely more accurate than would be possible with any realistic number of tubes. Using four 20W mono block amplifiers and any sound source I choose, the sound is for want of a better word, big (**Photo 10**). I have no idea why a large speaker should sound different from a small one if the parameters are the same, but they seem to sound different. Part of it may be related to the lower distance the speaker cones have to move to reproduce deep bass notes. Part of it may be just my thoughts about their physical size and imagined bigger sound. In any case, they are awfully nice sounding.

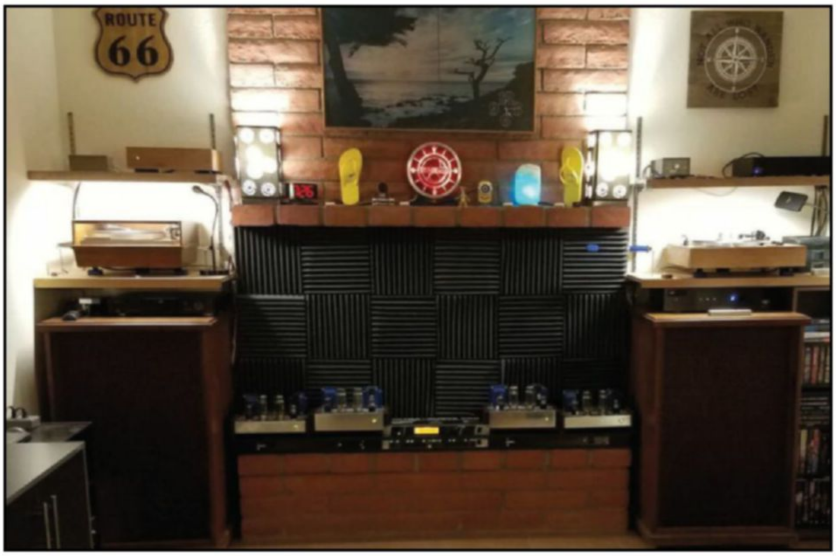


Photo 10: My system is all tube-based with the exception of the electronic crossover. Using four 20W mono block amplifiers and any sound source I choose, the sound is for want of a better word, big.

R&D Stories

The Tetrahedral Test Chamber Story (Part 2)

Consistent Loudspeaker Measurements

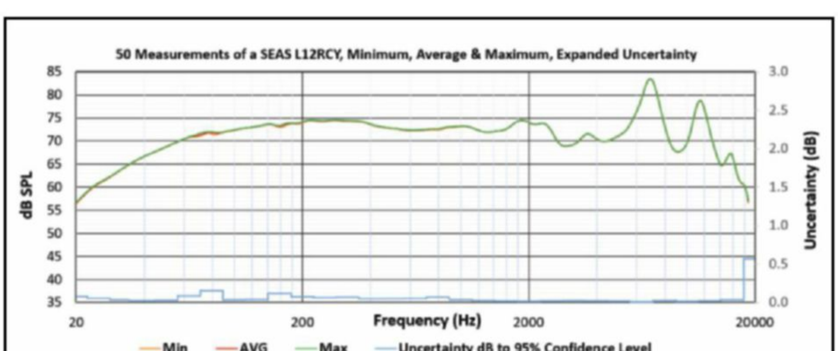
By
Geoff Hill

Starting from a simple theory and building upon his experience gained in designing chambers in Europe, US, and Asia, the TTC chamber was born. In Part 2 of this article, Geoff Hill now describes the initial launch and refinement phase, proving the chamber's measurement repeatability, and its inclusion in the international standards.

Once I had a Tetrahedral Test Chamber (TTC) design that satisfied all the criteria, I needed to bring together a team to get it into production. Alan Slaughter took on the QC and Tom Ayre translated my concept into a set of working 3D CAD drawings for the initial chamber. Alan began searching suppliers who could tackle the complex manufacture of the chamber.

We then set about building the first test chamber that could handle drivers up to 12". At this time, the Audio Engineering Society (AES) was about to hold its 51st Loudspeaker and Headphone Conference in Helsinki, Finland, in August 2013, so we loaded our TTC900 test chamber into the back of Alan's VW hatchback and headed off to Helsinki.

Eventually we arrived at Helsinki, a day later than planned after missing a ferry on our last leg of the journey. We were just in time for the conference where I planned to demonstrate the chamber. Outside the meeting room there was a series of deep cabinets stretching along one wall where several other people were setting up. Recognizing Wolfgang Klippel in whom I had confided earlier, I set up alongside him.



An SEAS L12RCY 4" driver was measured 50 times using a TTC350, the results show the consistency of the measurement to be better than 0.2dB 20Hz to 16kHz independent of the test analyzer.

Although the chamber was not formally on the agenda—there was quite a bit of interest in the idea and particularly in the consistency of results that even this prototype was capable of demonstrating repeatedly... flat 2Hz to 15kHz. Unheard of!

The next few days were very hectic, and I was unable to attend several presentations and papers that I wished to join. Many people were stopping in the hallway where we were set up and enquiring about the chamber and how it worked, even during the sessions! People were especially interested in the consistency and stability, and wanted to know what it could achieve.

This was successfully demonstrated by measuring first a bass-mid drive unit, saving the results and then changing to a tweeter using a different sub-baffle and saving those results. Then, we changed back to the first bass-mid driver and compared the results, displaying the difference curve between these measurements. Each time we made a measurement, no matter who swapped the drivers, the results were consistent over the full 20Hz to 20kHz range. As the days went by, it became obvious how stable and consistent the results were. They were within ±2dB 20Hz to 20kHz over the course of three full days.

At the AES conference, several people made some particularly useful comments. Peter Larsen, founder of Loudsoft, suggested that the minimum mic-to-baffle distance should not be less than the driver diameter and therefore to use 10cm, 31.6cm, and 100cm as standard distances as these also correspond to 20dB, 10dB, or 0dB gain steps compared to the standard 1m measurement distance.

Jorge Moreno Ruiz, Dean of Pontifical Catholic University of Peru (PUCP), suggested looking at Don Keele's work on low-frequency loudspeaker measurements, and this started me on the quest to accurately equalize a tetrahedral test chamber.

Also, during the conference, I talked to Sean Olive of Harman, who was AES President at the time. After demonstrating the TTC to him, he asked whether or not I would be showing it at the AES convention in New York. When I commented that I was too late as papers had been required two months previously, Sean said "leave that with me..."

Thanks to Sean, the TTC was submitted as an Engineering Brief to the AES 135th convention in NY and we decided to demonstrate the scalability of the concept with another chamber, the TTC350.

The AES 135th Convention in New York

So, it was back to the UK to start the process of designing a smaller chamber, the TTC350. At least this time we had a template, so while Tom worked with the 3D CAD, I concentrated on modeling the performance using Joerg Panzer's ABEC and VACS software, this laid out the underlying theory, which I then included in an AES Engineering Brief to be presented in New York.

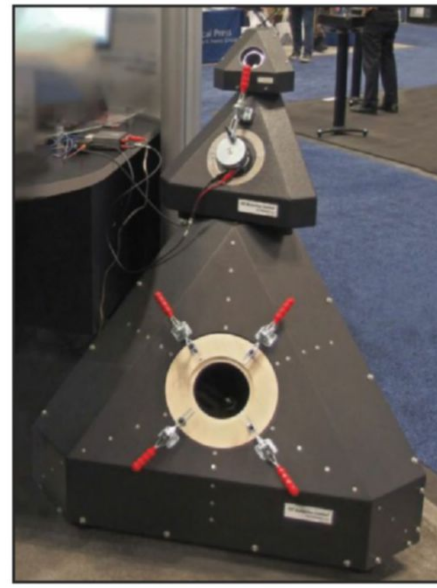
At the convention Wolfgang Klippel offered to make room for the TTC350 on his stand and I presented my EB123 engineering brief in the program and raised the concept with the AES Standards Committee 04-03—Loudspeakers and Headphones. Ed Simon and I presented independent proposals for measuring loudspeaker drivers. After seeing the Tetrahedral Chamber, Ed commented that the TTC was a "No Brainer" as it yielded better and more consistent results plus it was conceptually simpler and substantially smaller than his proposal.

At this meeting, Ed proposed the Project AES-X223 report on loudspeaker driver comparison chambers and the committee requested I produce content for a paper on suggested designs and construction methods for loudspeaker driver comparison chambers for the next AES convention in Berlin, Germany, in six months.

Subsequently, after the standards meeting many of the SC 04-03 committee members came to examine the Tetrahedral enclosure at the stand, one conversation that stood out was with Laurie Fincham of THX who asked, "Geoff, why did you get involved with all of this?" I replied, "Well Laurie, I could not afford an anechoic chamber." Laurie's reply surprised me: "No, none of us can!"

After the 135th convention, I continued work on the X223 report as requested, supporting this with an updated Engineering Brief EB143, which I previewed to the SC 04-03 committee at the 136th convention in Berlin in April 2014.

I submitted the data to the standards committee as my input to the X223 report. After much discussion, Mark Yonge, who was the AES Standards Manager at the time, felt that the importance of my engineering brief was more suited to an Information Document. Within the context of an AES Standards Committee, an Information Document goes through an extensive peer review, which can often take years and validates the proposed document. So, at the 136th convention the project X223 was changed from a report to an Information Document, with AES 73id-2019 being formally issued by the AES validating the TTC concept.



Three different sizes of Tetrahedral Test Chambers (TTCs) were presented in 2015 at the 139th International Audio Engineering Society Convention in New York.

The Final Tweaks

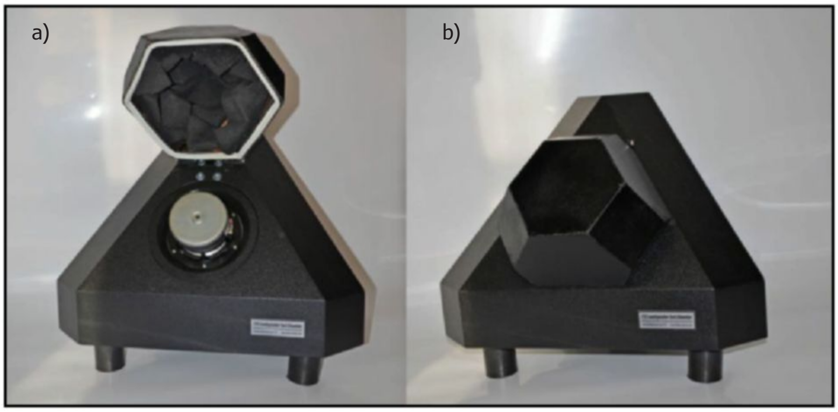
Between 2014 and 2018 there was an extensive period of refining the detailed design and manufacture of the TTCs. This included demonstrations at numerous conventions and conferences measuring a wide range of speaker drivers from 2" up to 15" to audiences in Berlin, UK, Las Vegas, NV, Los Angeles, CA, China, Hong Kong, India, France, Spain, Denmark, Ireland and back to New York. One photo shows a large chamber, the TTC1500, being used in India. Alongside is Palani Loganathan of Analogan in India. The TTC1500 is capable of measuring a 21" loudspeaker driver or a

About the Author

Geoff Hill, loudspeaker engineer, author, and inventor of the Tetrahedral Test Chamber (TTC), and CTO at Hill Acoustics, has been working in the loudspeaker and audio industry for more than 40 years. He co-founded Impulse Acoustics in 1979, his first foray into the world of professional audio design and manufacturing, focusing on horn loudspeakers and amplifier design. In 1985 he moved to Bowers & Wilkins as Production Test Engineer and shortly after to Goodmans Loudspeakers where he was Development Engineer for 13 years, designing primarily hi-fi loudspeaker drivers and complete systems. After a short stint as Senior Acoustic Engineer for Wharfedale, Geoff was hired by Ericsson as Senior Acoustic Engineer, from there he moved to Motorola, to design, build, and implement a multi-user Acoustic Measurement Laboratory and large anechoic chamber for telecommunication products. In 2005, he returned to the world of speakers as Senior Acoustic Engineer for Monitor Audio, Ltd., where for nearly six years he modernized the test and measurement facilities and equipment, designed and installed an anechoic chamber and led research efforts. In 2013, after working as technical director for other loudspeaker companies and continuing to offer technical support on the design and testing of loudspeakers, Geoff founded his own consultancy business, Hill Acoustics.

Geoff is the vice-chair of the Audio Engineering Society's AES SC-04-03 Working Group on loudspeaker modeling and measurement and a member on the International Electrotechnical Commission's IEC Working Group TC 100/TA 20/PT 60268-22. Geoff is also the author of the book *Loudspeaker Modelling and Design: A Practical Introduction* (Routledge 2018).

An example of a large chamber, the TTC1500, being used in India. Standing alongside it is Palani Loganathan of Analogan in India. The TTC1500 is capable of measuring a 21" loudspeaker driver or a complete loudspeaker enclosure and TTCs are fully scalable for any size driver.



A smaller chamber, the TTC350, is shown fitted with an Acoustic Isolator open (a) and with the Acoustic Isolator closed (b).

Project Files

To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Resources

AES73id - AES information document for acoustics—Loudspeaker driver comparison chambers

G. Hill, *Loudspeaker Modelling and Design a Practical Introduction*, Routledge/Focal Press, 2018. www.routledge.com/Loudspeaker-Modelling-and-Design-A-Practical-Introduction/Hill/p/book/9780815361336

G. Hill, "A Guide for Building Your Own Test Chamber," *audioXpress*, September 2020. https://audioxpress.com/article/practical-test-measurement-loudspeaker-measurements-at-home-a-guide-for-building-your-own-test-chamber

IEC60268-21:2018—Sound system equipment—Part 21: Acoustical (output-based) measurements

IEC60268-22:2020 Sound system equipment—Part 22: Electrical and mechanical measurements on transducers

complete loudspeaker enclosure and TTCs are fully scalable for any size driver. Another photo shows a smaller chamber the TTC350 fitted with an Acoustic Isolator open (a) and with the Acoustic Isolator closed (b).

At some of these venues, I made presentations and submitted papers to the AES, ISEAT, ALMA (now ALTI) working with: Audio Precision, Audiomatica CLIO, Etani, HolmImpulse, Klippel, Listen, Inc., Loudsoft, MLSSA, NTi, and Prism Sound (now Spectral Measurement). During this time, I published a white paper (See references/*audioXpress'* Supplementary Materials) on the TTCs, which is included in my book *Loudspeaker Modelling and Design a Practical Introduction*.

Further Validation

After an extensive peer review process, the concept has now been accepted as a part of the international IEC standards—IEC 60268-21-2018 and IEC 60268-22-2020—and the AES issued AES73id-2019 in which David Murphy of Krix, collaborated with me to mathematically prove the measurement stability achievable with loudspeaker test chambers, with the TTC being used as an example. The TTC routinely produced results with stability of better than ±0.5dB, 20Hz to 20kHz (95% confidence), calculated by internationally accepted statistical principles.

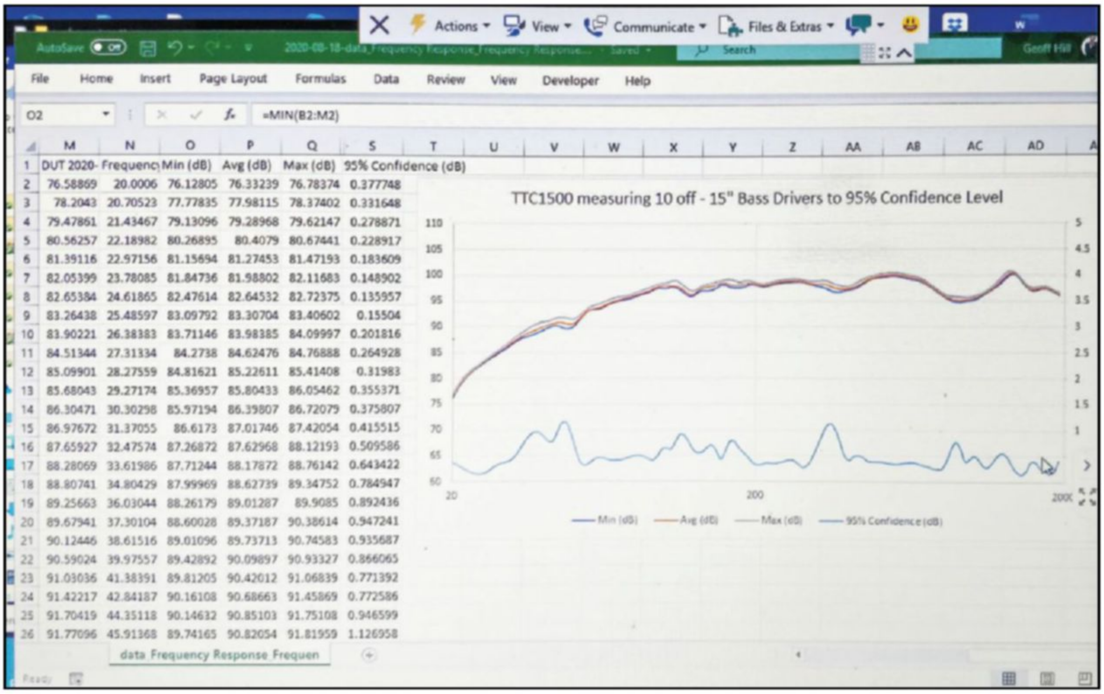
Conclusion

Starting from simple wavelength theory and Leo Beranek's wedges, tetrahedral structure, measurement theory, and building upon my experience gained in designing chambers in Europe, US, and Asia, the TTC chamber was born.

It's surprising that the concepts behind the TTC were not harnessed earlier as it is a "game-changer" in the way that we can measure loudspeakers of any size and shape. The loudspeaker test chambers are in use throughout the world, providing consistent, reliable, and accurate measurements of loudspeaker transducers and loudspeaker systems without the cost, complexity, and space requirements of an anechoic chamber, and independent of the test analyzer.

TTCs are available commercially through Spectral Measurement (formerly the Test & Measurement division of Prism Sound), which signed a licensing deal with Hill Acoustics enabling it to manufacture the chambers.

Author's Acknowledgement: During my years of developing the concept of the Tetrahedral Test Chamber, bringing it into use, and having it validated by its inclusion in international standards, I have benefited from the invaluable insights and wealth of constructive suggestions of my peers including engineers, loudspeaker



This test demonstrates a similarly tight measurement tolerance. This is especially so at the very lowest frequencies below 50Hz, which is exactly the frequency region where conventional anechoic chambers have difficulty.

measurement instrument manufacturers, and end users who reflected on my explanation of the concept and allowed me the time to demonstrate the TTC to them first hand. For that reason, I would like to thank: Tom Ayre, Dave Berriman, Graham Boswell, Peter Brooks, Vance Dickason, Laurie Fincham, Dan Foley, Su Hill, Steve Hutt, Stefan Irrgang, Don Keele, Wolfgang Klippel, Phil Knight, Peter Larsen, Patrick Macey, João Martins, Thomas Mintner, David Murphy, Bruce Olson, Doug Ordon, Ismo Pänkäläinen, Joerg Panzer, Jorge Moreno Ruiz, Gregor Schmidle, Philipp Schwizer, Ed Simon, Alan Slaughter, Bob Sturgeoner, Steve Temme, and Mark Yonge.

Effects of Negative Feedback

By
Richard Honeycutt

In this Hollow-State Electronics article, Richard Honeycutt uses LTSpice modeling to determine how different amounts of negative feedback affect the output from a hollow-state power amplifier.

Negative feedback was invented by H. S. Black of the Bell Telephone Laboratories in 1927. It involved feeding a portion of the output of an amplifier out-of-phase back into the input, with the result of improving frequency response and reducing distortion. It also reduced the amplifier's gain. A downside of negative feedback's effect on distortion is that rather than a gradual increase of distortion as an amplifier's input voltage increases, which is sometimes called a graceful onset of distortion, an amplifier using negative feedback rather abruptly increases the distortion after the input voltage is raised above the level that causes clipping. This abrupt increase in distortion is not pleasing to the ear! Therefore, many audiophiles who love hollow-state equipment believe that no negative feedback should be used in good amplifiers. Some of these folk are willing to ignore the negative feedback of unbypassed cathode resistors, but not overall negative feedback.

We have discussed negative feedback in previous Hollow-State articles. In October 2016, we examined general negative feedback theory. In July 2021, we looked at how negative feedback is used in hollow-state amplifiers. Now we will use LTSpice modeling to find out how different amounts of negative feedback affect the output from a hollow-state power amplifier.

Circuitry

We will begin by focusing on amplifiers used for music listening and studio or control room monitoring, leaving musical instrument amplifiers for later. In the early 1940s, D. T. N Williamson designed what was to become the most popular hollow-state amplifier topology for the succeeding two decades. Proceeding from the belief that most amplifier distortion came from nonlinearities in the output transformer, Williamson included the transformer in the negative feedback loop of his

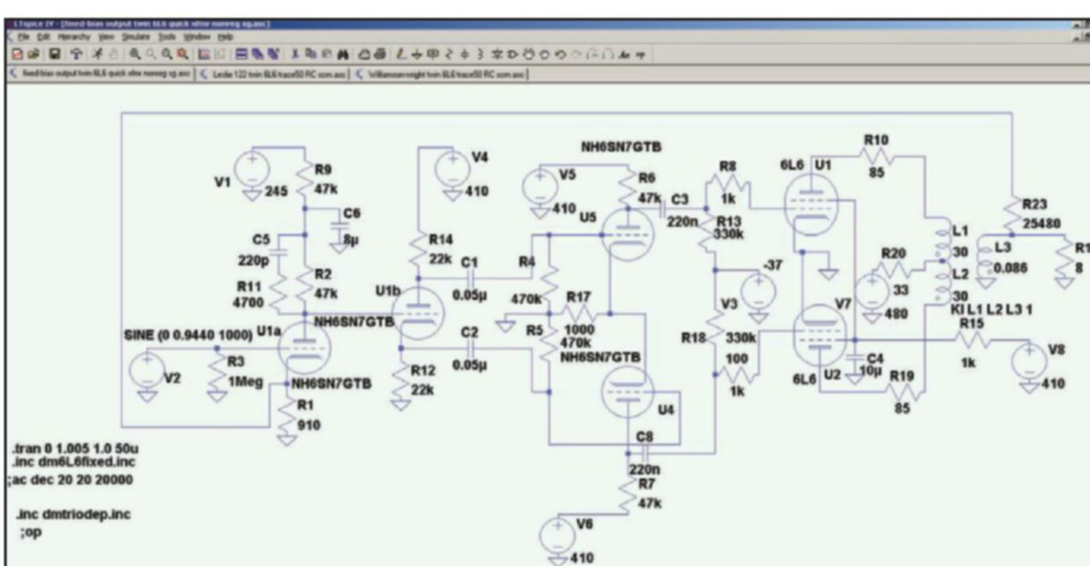


Figure 1: This LTSpice model of a Williamson-topology amplifier incorporates a typical amount of negative feedback.

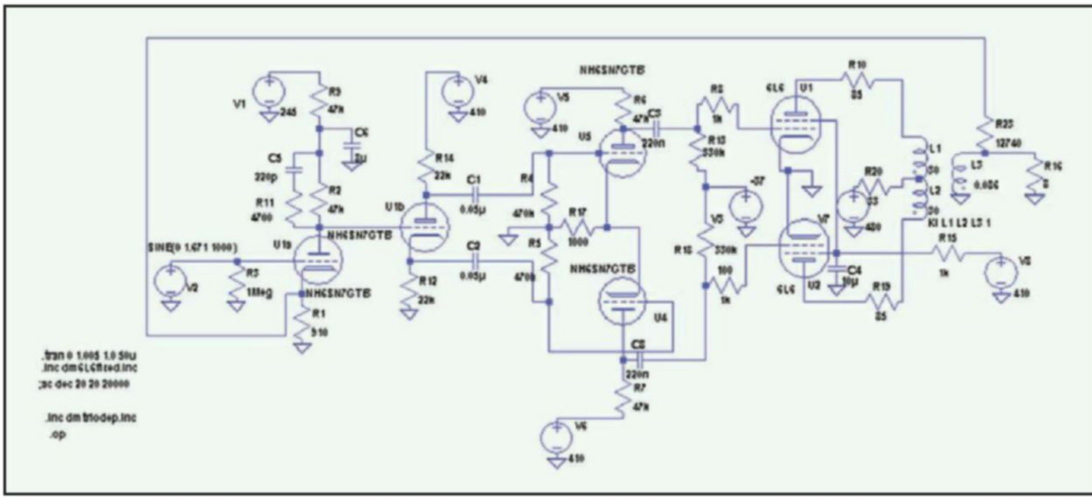


Figure 2: Cutting the value of R23 in half doubles the feedback.

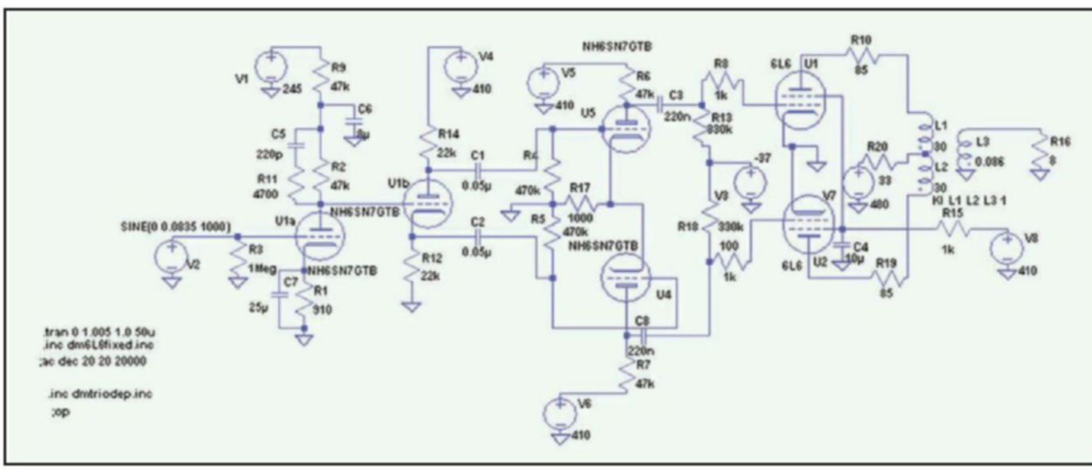


Figure 3: By omitting R23 and bypassing R1, we can eliminate the negative feedback.

four-stage amplifier. His design was created to be as linear as possible, with negative feedback applied to improve an already-excellent design. Many amateurs built Williamson amplifiers using either his 1947 original design or his later (1949) improved design. Naturally other designers introduced improvements [1], as discussed in the August 2017 Hollow-State Electronics article in *audioXpress*, and commercial manufacturers made tweaks of their own, producing a number of Williamson-like amplifiers.

Figure 1 shows the model for the Williamson-like amplifier, incorporating a "normal" amount of negative feedback. Input tube U1a acts as a common-cathode amplifier for the program signal, and as a common-grid amplifier for the feedback signal, which is derived from the top of the output transformer through R23. The feedback signal voltage appearing at the cathode of the tube is the sum of the feedback signal from R23 and the local-feedback signal derived from unbypassed cathode resistor R1. (With the resistor values shown, the voltage gain of the amplifier is 17.4, giving an output power of just under 17W at clipping, with an input of $0.944V_p$.

Figure 2 shows the amplifier with the value of R23 halved, effectively doubling the amplitude of the feedback signal. As expected, this change reduces the voltage gain, which drops to 9.9. Power output at clipping is about 19W, requiring an input voltage of $1.671V_p$.

Figure 3 shows the schematic of the circuit with no feedback: R23 has been omitted and R1

About the Author

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

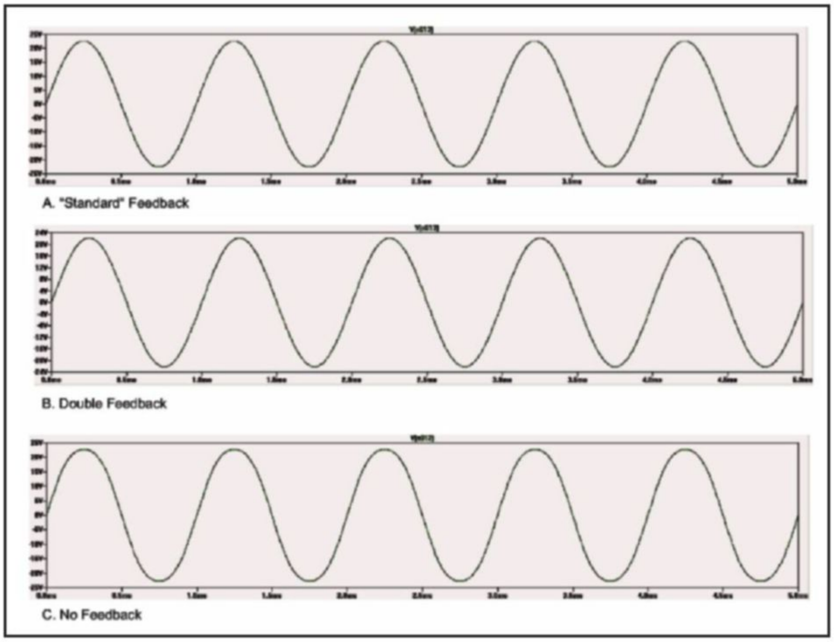


Figure 4: There is little difference in the output waveforms just below clipping with the three levels of negative feedback.

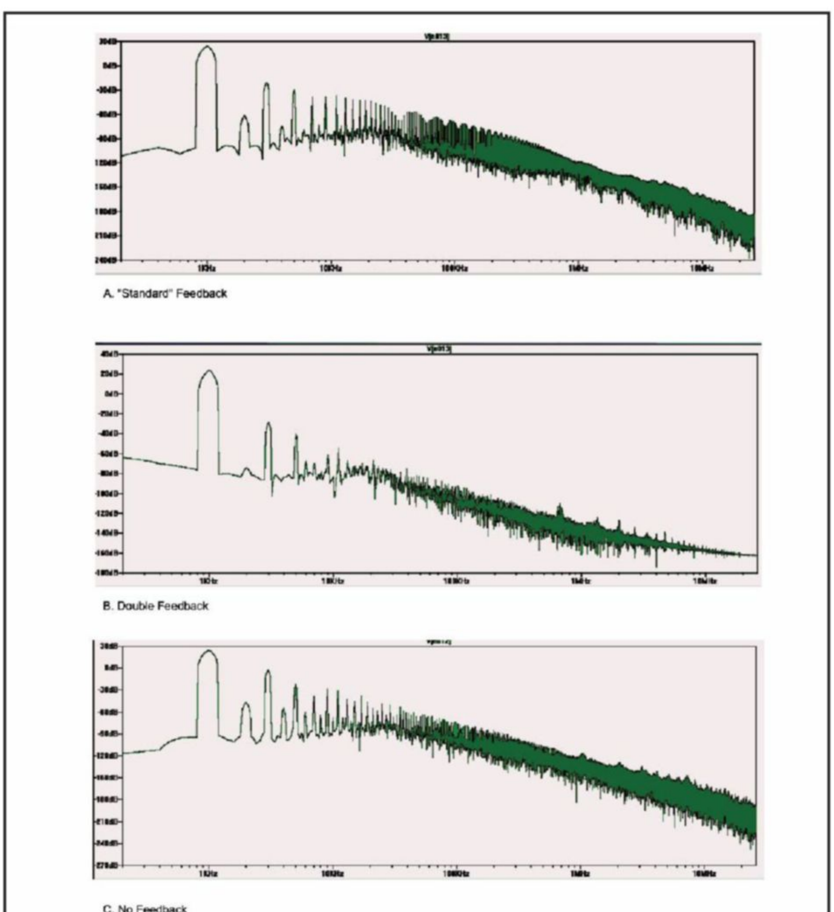


Figure 5: The distortion signature just below clipping is affected by the negative feedback level.

has been bypassed by a 25μF capacitor. Eliminating the negative feedback increases the voltage gain to 193. The clipping point is difficult to determine, because the onset of clipping is very gradual. The output power begins to show effects of compression at a value of about 10W, requiring an input voltage of 80.4 mV_p, at which level the voltage gain has dropped to 157.

Distortion

Figure 4 shows the output waveforms for the amplifier with three levels of negative feedback. You will notice that there is little visible difference. In fact, the total harmonic distortion (THD) does not differ much either: with "standard" feedback, THD is 3.5%; doubling the negative feedback drops it only slightly, to 2.5%, and eliminating negative feedback increases it slightly to 4.1%. The effect of negative feedback on THD is strongly dependent upon the gain structure of the amplifier. This circuit's stage gains are allocated so that the input stage does not distort much before the onset of clipping in the phase splitter, and the split-load phase splitter and balanced driver stage are inherently pretty linear. If the input stage were hotter (higher gain) as in a guitar amplifier, negative feedback would have more effect upon distortion below clipping.

As you can see in **Figure 5**, the fast Fourier transform (FFTs) of the three output waveforms reveal differences in the distortion signature in spite of the similar appearance of the waveforms. Note that the LTSpice graphing function automatically scales the display so that the vertical axes of the three FFTs are not the same. However, if you compare the amplitude of specific harmonics with that of the fundamental, you will notice some differences:

- Both "standard" feedback and double feedback reduce the levels of even-order harmonics. (This is one reason that some audiophiles who oppose using negative feedback give for preferring the sound of non-feedback amplifiers.)
- Double feedback reduces even-order harmonics more than "standard" feedback does.
- Double feedback reduces the levels of higher-order odd harmonics (above the fifth), compared to "standard" or no feedback.

If we compare distortion performance when the amplifier is clipping, we see much greater effects of negative feedback. **Figure 6** shows the output waveforms using the same three levels of feedback,

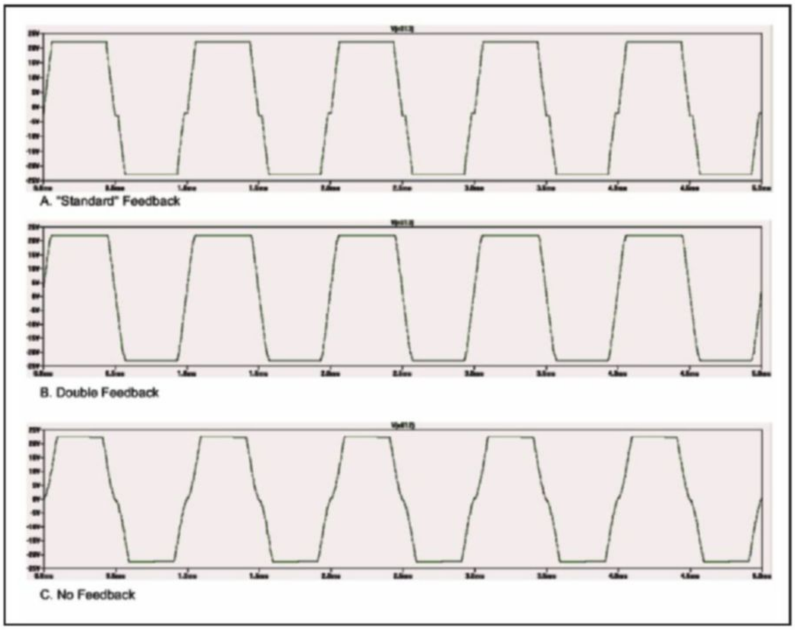


Figure 6: The effect of negative feedback level upon the distortion signature is greater above clipping.

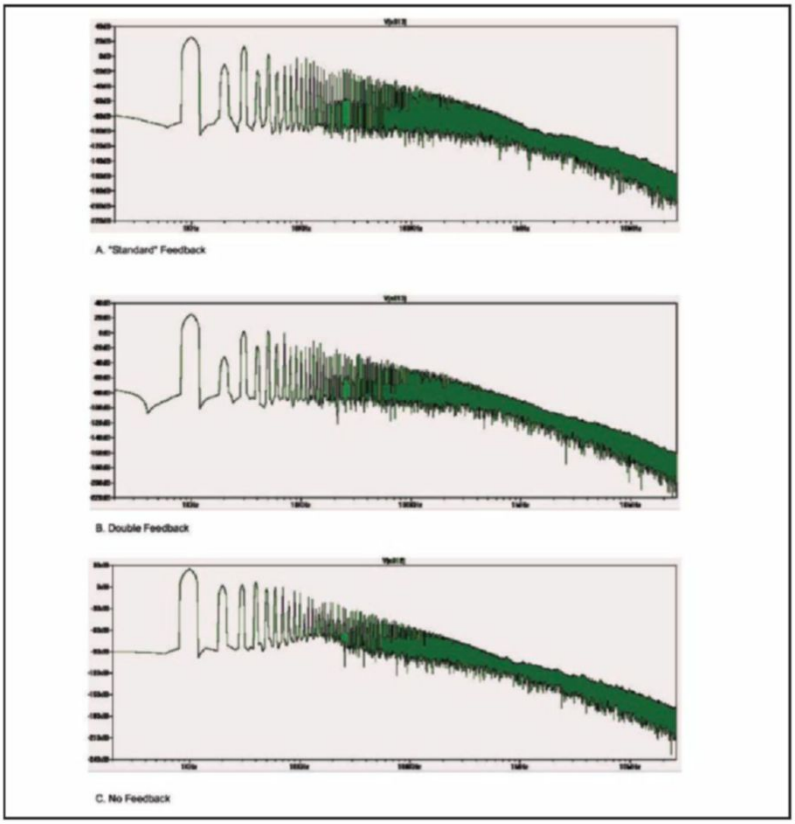


Figure 7: The distortion signature above clipping is markedly affected by the feedback level.

but with the input level increased by 10dB above the clip point. The main differences among the waveforms is the crossover distortion, which can be clearly identified with "standard" feedback, is less pronounced with no feedback, and is essentially invisible with double feedback. The corresponding THD levels are 22.6% for "standard feedback, 11.1% for double feedback, and 31.7% with no feedback.

Figure 7 shows the FFT plots for the three conditions. Comparing these plots with those in Figure 5, we notice that, as we would expect, all harmonic levels are increased by overdriving. The even-order harmonics are also larger relative to the odd-order harmonics than for the unclipped output wave, and higher-order harmonics are also increased. The largest proportional increase in higher-order harmonics appears in the "standard-feedback" FFT, as would be predicted from the more distinct crossover notch in the waveform.

Frequency Response

Negative feedback generally flattens frequency response of an amplifier. Notice that the response with no feedback droops at the lowest frequencies due to the combined effects of coupling capacitors C1, C2, C3, bypass capacitor C7, and the inductance of the output transformer. The high-frequency response droops because of Miller capacitance in the triodes and the C5-R11 network bypassing the plate resistor of U1a. (This network is included to prevent parasitic oscillation.)

"Standard" negative feedback does indeed flatten both the low-frequency and the high-frequency responses, and the "double" level of feedback flattens it even more (**Figure 8**). However, this model is not perfect with regard to predicting frequency response, because it uses a simplified output transformer model. Only the primary and secondary winding inductances and the primary winding resistance are included in the transformer

Reference

[1] Wright, T. M., "Improving the Williamson Amplifier," Electronics World, June 1961, pp. 104-106.

Resources

R. Honeycutt, "Negative Feedback," *audioXpress*, July 2021.

R. Honeycutt, "Williamson Amplifier Games," *audioXpress*, August 2017.

R. Honeycutt, "Negative Feedback in Hollow-State Amplifiers: Its Design and Effects," *audioXpress*, October 2016.

model, not the leakage inductance, interwinding capacitances, and secondary winding resistance. These approximations affect both low- and high-frequency response.

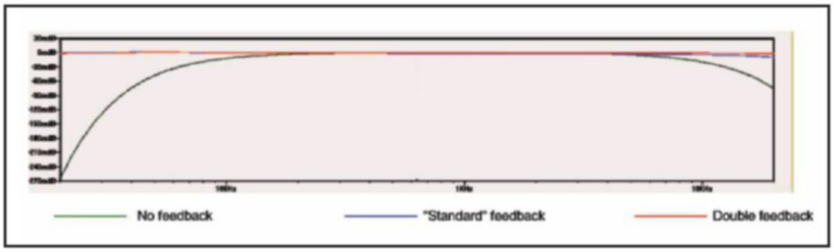


Figure 8: The level of negative feedback affects the frequency response.

Musical Instrument Amplifiers

The Williamson-like circuit we have used in this discussion is quite similar to the circuitry of many amplifiers used for home listening rooms, studio playback monitors, and control room monitors. Musical instrument amplifiers often do not include overall feedback. For example, the Fender Twin Reverb model AA270 amplifier, whose schematic is shown in **Figure 9**, only includes the phase splitter and output stage in the feedback loop. Many Ampeg and other amps do the same. Orange and a few other brands use no overall negative feedback at all, although Orange does use negative feedback in its unique tone control circuitry. Of course, very low distortion and extended frequency response are not usually required for guitar and electric bass amps.

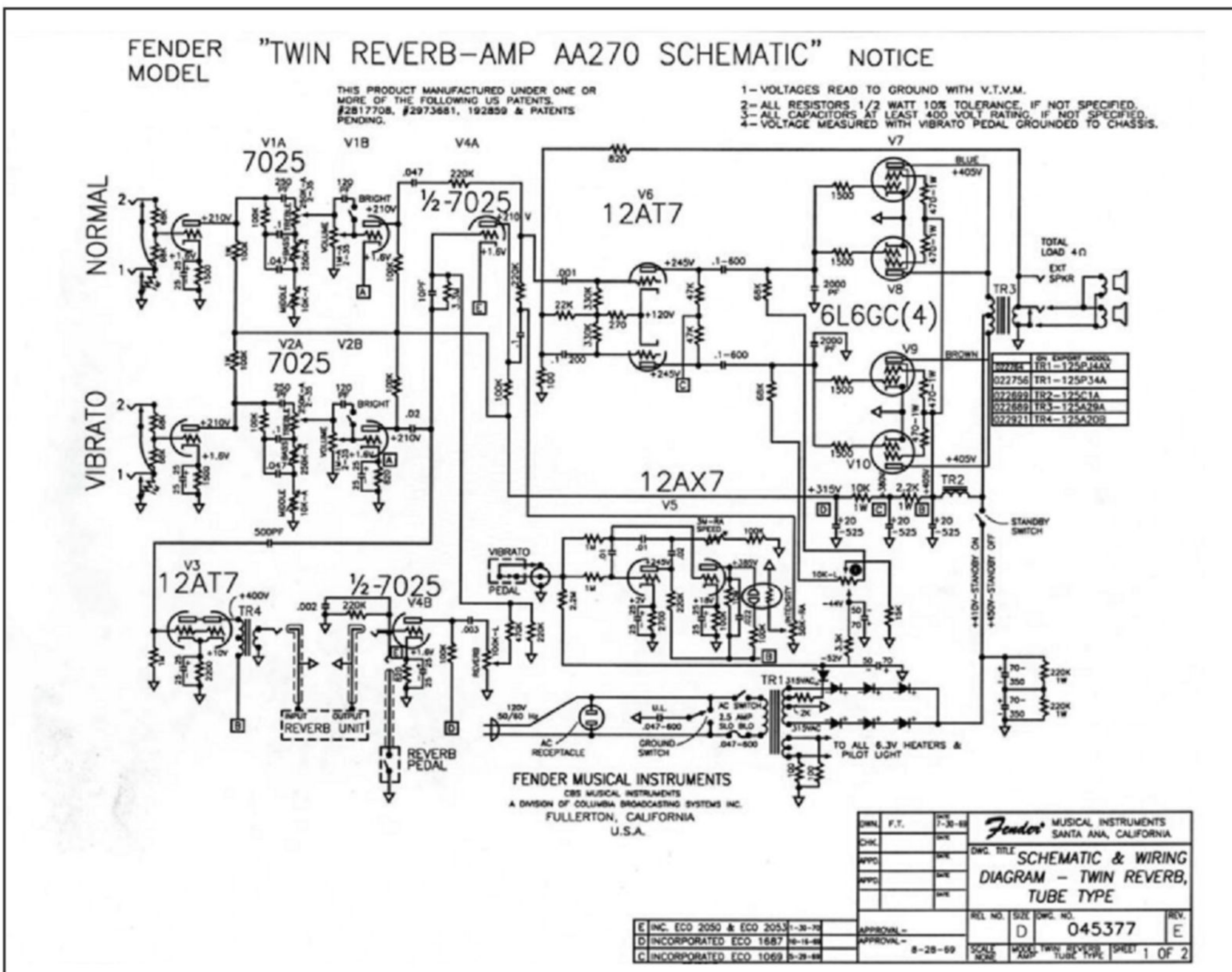


Figure 9: The schematic for the Fender Twin Reverb Model AA270 amplifier only includes the phase splitter and output stage in the feedback loop.

TRIAD

ELECTRONIC TRANSFORMERS

- Original Equipment
- Replacement
- Amateur

TRIAD MAGNETICS, INCORPORATED

460 HARLEY KNOX BLVD., PERRIS, CA 92571

www.TriadMagnetics.com