

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Market Update

Residential Integration The Home Audio Perspective

By J. Martins

R&D Stories

The Story of aptX An Epic Journey

By Jonny McClintock

Sound Control

Considerations Concerning RT Measurement

By Richard Honeycutt

Fresh From the Bench

Sumiko Phono Cartridges and RS78 78 rpm Stylus

By Gary Galo

You Can DIY!

All American Tower Speakers

By Ken Bird

Speakers

Replacing IEC 60268-5 (Part 2)

By Geoff Hill

You Can DIY!

Making a DC Power Supply with Audible Benefits Part 1: Design Options

By Frans de Wit

Audio Electronics

Triode Common Cathode Stages, the Oracle Equation, and Theodosian Objections

By Christopher Paul



ANATOMY OF A 300B

PRESENTED BY *Western Electric*

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

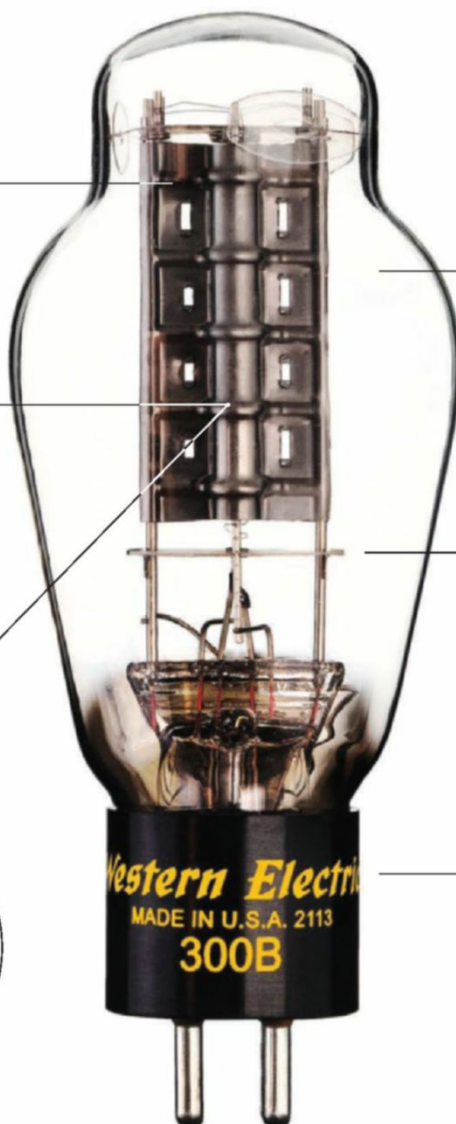
Molybdenum grid creates negative coefficient of resistance

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

Proprietary mica coating increases dielectric strength, optimizes higher voltage operation

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

Quality guaranteed with a five-year extended warranty



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

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

AVAILABLE NOW

westernelectric.com/300b



MANUFACTURING / DESIGNING TRANSFORMERS & ENCLOSURES SINCE 1917





October 2022

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, suggestions, and contributed articles to:

audioXpress, Editorial Department

P.O. Box 417

Chase City, VA 23924, US

E-mail: editor@audioxpress.com

Legal Notice:

Each design published in *audioXpress* is the intellectual property of its author and is offered to readers for their personal use only.

Any commercial use of such ideas or designs without prior written permission is an infringement of the copyright protection of the work of each author.

© KCK Media Corp. 2022
Printed in the US

More Room for Audio

This *audioXpress* issue was sent to print a few weeks away from CEDIA Expo 2022, which takes place in Dallas, TX. It was already pretty clear to me that this year's return of the most important residential integration show will be different, but I'm curious about the new trends.

Unlike certain buzzwords that seem to appear and fade fairly quickly, "Smart Home" has been an enduring but also somewhat elusive concept that involves a multitude of different disciplines and technologies, and is not always consensual—just ask the kitchen appliance manufacturers what they consider "intelligent."

A few years back, that smart home horizon meant essentially evolving from automation to connected solutions. And with the evolution of smart devices, suddenly the human interface became the center focus, evolving from touchscreens and apps to presence detection and synchronized devices in and out of the home. A frequently cited audio-related example was the possibility to move from the car when arriving home and continue listening to the same podcast or playlist—with "the connected system" simply transferring the content from the car speakers to the multiroom audio system by detecting your arrival and presence inside the home.

Then came voice interfaces, personal voice assistants, connected ecosystems, and AI, and suddenly Smart Home was no longer a concept. Instead it became a target market that appealed to a whole new range of technology players, from Amazon and Google to Sonos. While all this seemed like a great plan for further progress—and it was starting to make sense at CEDIA 2019—the global pandemic changed our perspectives.

With the whole family suddenly forced to shelter at home, the reality of multiple activities, zones, and users had to be accommodated. And it was as if everything was new, mainly because the technology we all got used to having with us (e.g., smartphones, headphones, and laptops), was what made the difference and allowed us to transform our "home" experience into an extension of everything we did outside—including work, learn, and exercise, for extended periods. Those activities overlapped with our usual home routines and home entertainment opportunities and made us all rethink our priorities. And personal devices such as headphones, headsets, and earbuds, used for all those activities, not surprisingly became the hottest consumer electronics category—something that was never accounted for as part of Smart Home scenarios. All this had a major impact on the audio industry, which was already at the center of the Smart Home (re)evolution since smart speakers with voice interfaces entered the home.

When we look at it from the residential integration perspective, the importance of the underlying technology infrastructure becomes abundantly clear. And how audio in the home will inevitably be connected to that infrastructure—wired and wireless.

But before we can dream about what the future looks like, we need to first worry about the present, because we are going through a convergence of very unique circumstances. Following a global pandemic that continues to asymmetrically affect different countries and regions, we continue to face rising logistical costs and production disruptions, resulting in cycles of supply chain constraints. That includes severe constraints in access to raw materials and also rising inflation, because of geopolitical tensions.

When I visited the first audio trade shows of 2022, the message about product availability was at the center of all conversations. While many audio manufacturers are downplaying the constraints due to the relatively small quantities involved, it's clear that products are waiting on assembly lines for components that no one had predicted would not be available. The post-pandemic situation evolved with the stock depletion at many major wholesalers that traditionally have supplied common components to audio manufacturers, and the industry in general not being able to correctly predict and manage supply and demand.

These supply chain troubles and imbalances are far from over, and will take years to bring under some form of control. And although many discussions regarding restructuring production, reshoring, and design to availability are being touted as needed solutions, those are long-term strategies.

This is especially of concern for system integrators with project designs that require quantities of certain products. There are many "sold out" items that will never be brought back into production. System integrators will need to put much more thought into what they specify.

We'll also need to accept the fact that products we considered buying before and didn't, might never be available again. Except perhaps in second-hand marketplaces.

J. Martins
Editor-in-Chief



The Team

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

Technical Editor: Jan Didden

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

OUR NETWORK



SUPPORTING COMPANIES

ACO Pacific, Inc.	45	Menlo Scientific, Ltd.	59
All Electronics Corp.	57	NTi Audio, Inc.	31
Amadeus	29	Parts Express International, Inc.	68
COMSOL, Inc.	9	Redco Audio	45
Hammond Manufacturing, Ltd.	3	SB Acoustics	63
KAB Electro-Acoustics	29	Triad Magnetics	67
Lenbrook International	47	Western Electric	2
Lundahl Transformers	37		

NOT A SUPPORTING COMPANY YET?

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

The Audio Voice

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



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





ax Cache

Every issue of audioXpress published since 2001, on USB.

Order at: cc-webshop.com





● Features

18 Sumiko Phono Cartridges and RS78 78 rpm Stylus

By Gary Galo

Gary Galo reviews the Sumiko Olympia and the Wellfleet phono cartridges, together with the new RS78 stylus, Sumiko's first 78-rpm solution, designed to meet the needs of many vintage recording enthusiasts.

26 The Story of aptX An Epic Journey

By Jonny McClintock

From film sound to radio broadcasts, studios, and finally to Bluetooth and more than 12 billion devices, the successful story of aptX technology is inextricably linked to the history of digital audio. Our author details this remarkable corporate journey.

32 Replacing IEC 60268-5 (Part 2) The Importance of Measurements

By Geoff Hill

Geoff Hill concludes his discussion about the work taking place on loudspeaker measurement standards, and the progress the International Electrotechnical Commission (IEC) has made with its standards efforts.

42 Triode Common Cathode Stages, the Oracle Equation, and Theodosian Objections

By Christopher Paul

For those readers interested in tube amplification, this article proposes a simplified equation of what might appear to be multiple mathematical factoids into a single concept, intended to help characterize and understand triode circuits.

48 Making a DC Power Supply with Audible Benefits (Part 1) Choosing the Design

By Frans de Wit

Following two previous *audioXpress* articles that explored multiple power supply circuits with potential for high-end audio applications, Frans de Wit details a high-quality power supply especially targeted toward preamplifiers, record-players, and other low power appliances.

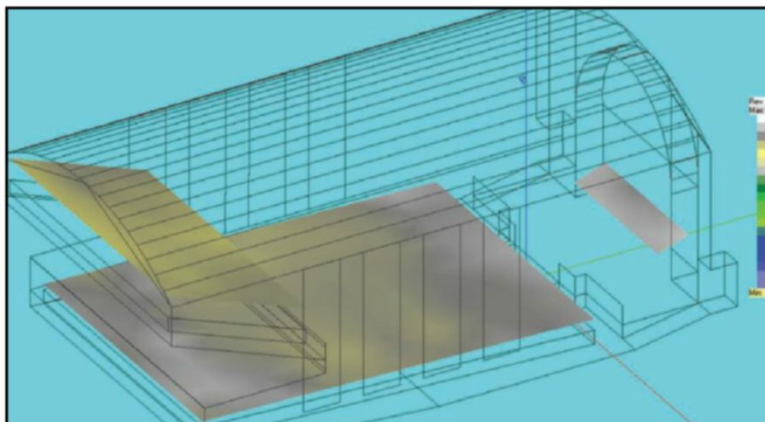
54 All American Tower Speakers

By Ken Bird

In this article, our experienced speaker builder takes us along with him on his quest to build a loudspeaker made solely from drivers manufactured in the United States.

COVER ART

Triad PDX Installation
Speaker Series
Courtesy of: Snap One



● Market Update—Residential Integration

MARKET UPDATE

8 The Home Audio Perspective

By J. Martins

Since the adoption of automation systems and home networks, residential integration has evolved gradually to connected systems and is now increasingly associated with the transition to the Smart Home. Learn more about the audio companies that are facilitating the transition.

● Departments

4 From the Editor's Desk

5 Supporting Companies

● Columns

SOUND CONTROL

38 Considerations Concerning RT Measurement

By Richard Honeycutt

In this article, our acoustics expert explains the history behind the reverberation time (RT) measurement and its importance. He then shares his preferred procedure for measuring RT.

HOLLOW-STATE ELECTRONICS

60 Old Timer, Williamson, and Ultralinear Amplifiers Families of Hollow-State Power Amplifiers

By Richard Honeycutt

The July and September Hollow-State Electronics articles examined the distortion performance of the three families of amps. This article recaps those findings and compares them to the last two families, the Williamson and ultralinear amps.

● 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



audioxpress

The Home Audio Perspective

By
J. Martins
(Editor-in-Chief)

Since the adoption of automation systems and home networks, residential integration has evolved gradually to connected systems and is now increasingly associated with the transition to the Smart Home.

Image courtesy of StormAudio

While audio products continue to be at the core of residential systems, the transition to the smart home is increasingly less audio-centric from a system perspective. Established technical standards have long started to be challenged by the need to enable complete control over IP and network distribution of signals—now increasingly converging with power distribution and management. Networked systems have evolved to Power-over-Ethernet (PoE), while at the same time control and media streaming is fully embracing wireless technology, standards, and protocols.

This transition to the smart home and connected systems obviously encouraged large technology companies to invest in residential technologies through new developments, which have included smart speakers as a way to create a connected-home hub, or through direct acquisition of other companies that made thermostats, door locks, IP cameras, or lighting control and power systems. Investments from Samsung, Amazon, and Google in the smart home space completely changed the pace of this evolution to the smart home. And even in the traditional residential integration space, we've seen organic expansion, as well as mergers and acquisitions, in an effort to build the required muscle and systems' scope required to address the smart home. Snap One, the company that resulted from the acquisition and merger of SnapAV with Control4 (which in turn had already expanded through multiple acquisitions) is a clear example. But even Crestron is increasingly resorting to strategic partnerships for its systems approach, while focusing on selling the system infrastructure.

This even motivated the Custom Electronic Design and Installation Association (CEDIA) to rethink its mission, and recently renaming itself as CEDIA Global Association for the Smart Home Technology Industry.

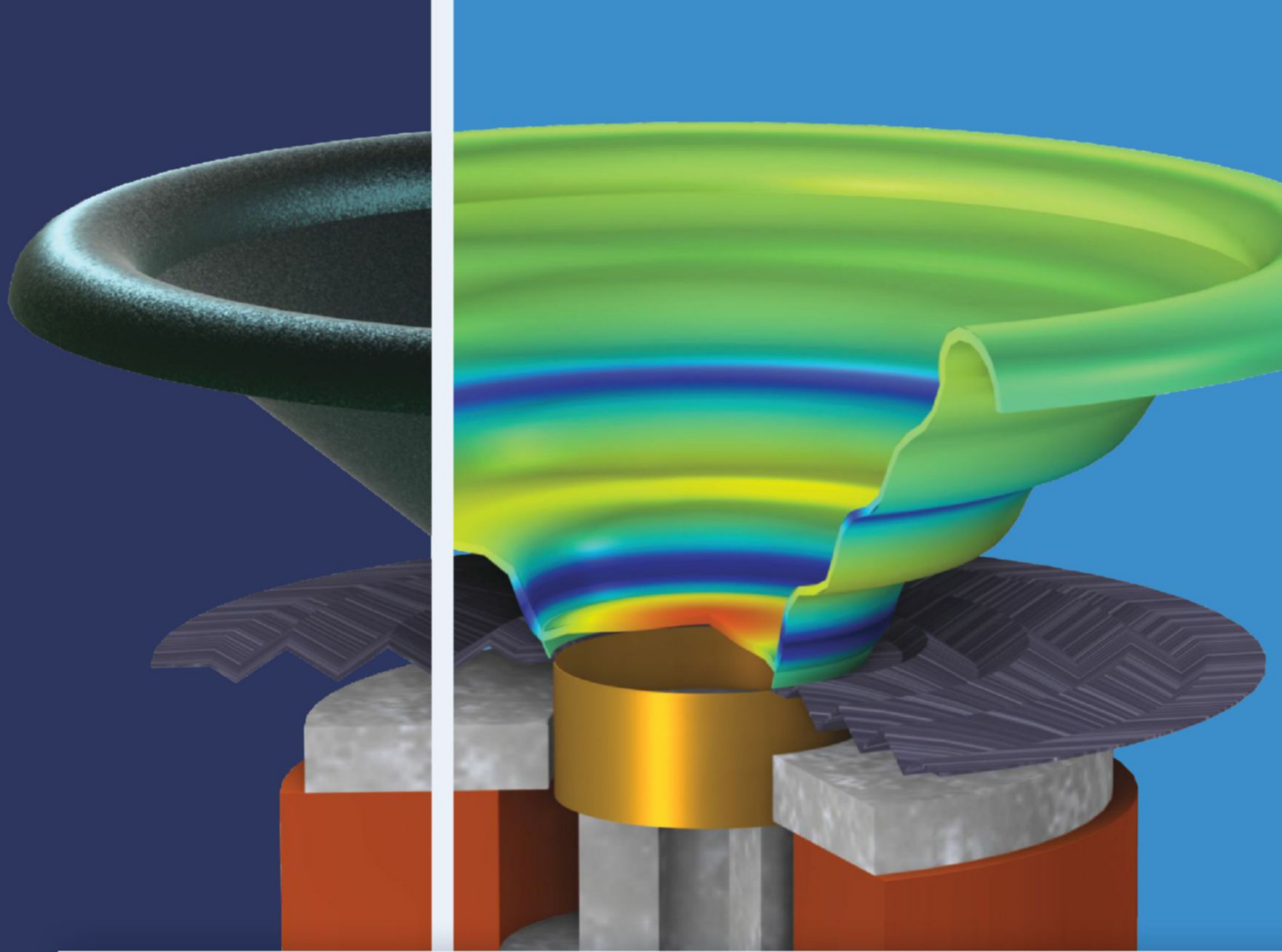
Before that, the state of the industry in residential integration—from CEDIA's perspective—remained largely distanced from related convergence discussions taking place in other relevant technology forums, such as power and lighting. But suddenly the effects of the global pandemic introduced drastic changes and accelerated existing trends. With consumers realizing the importance of investing in home technology and infrastructure, it was no longer a question of looking at technologies for technology sake, but actually responding to needs and priorities.

A New Home

Audio remains a priority. It is an essential part of residential systems, and was a category that largely benefited in terms of investment growth by homeowners. No longer just for entertainment, sound in the home is now regarded by more people as essential for well-being. As the pandemic brought increased awareness to the importance of sound and acoustics and new audio equipment was introduced to the home for different activities, working from home made quality of communications a top priority, and there was a new predisposition to invest in high-quality home audio systems, including home theater.

As highlighted in a recent CEDIA article by industry consultant Ian Bryant (former Senior Director of Strategic Partnerships at CEDIA), with the pandemic, residential integration suddenly was expanded by increasing demand in wellness, power and energy, AI and machine learning, lighting, kitchen, and bath, and connected appliances.

While power and energy remains mostly a separated discipline from what is normally the integration of low voltage electronics and systems, consumers have invested in energy management

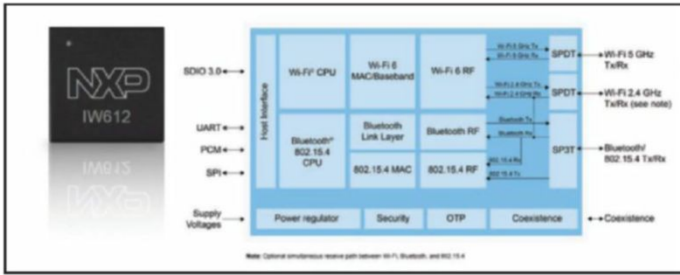


Take the Lead in Acoustics

with COMSOL Multiphysics®

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

» comsol.com/feature/acoustics-innovation



At CES 2022, NXP introduced the IW612, the first secure tri-radio in a single SoC to bridge devices in the smart home that require connecting Thread or Bluetooth products to the cloud using the integrated Wi-Fi 6 radio. Additionally, the IW612 enables communication between Matter devices, regardless of whether the devices use Wi-Fi or Thread. This allows Matter-over-Wi-Fi products to control and monitor Matter-over-Thread devices, and vice versa, for seamless interoperability.



Infineon Technologies new AIROC Bluetooth LE and 802.15.4 family of SoC devices can help companies quickly bring low-power, high-performing Matter products to market.



Synaptics launched its SYN4381 Triple Combo system-on-chip (SoC), combining Wi-Fi 6/6E (802.11ax with extended 6GHz operation), Bluetooth 5.2 with BLE Audio and High Accuracy Distance Measurement (HADAM), and IEEE 802.15.4 with built-in support for the Thread protocol and Matter application layer. The solution simplifies product development while supporting 600Mbps video and data, low-power Thread communication, and platform-agnostic device setup and interoperability.

inside the home, connected and “smart” power distribution panels, solar and wind power generation, and battery technology, creating a new important source of revenue for integrators in the “smart home” space.

As Bryant highlights, the pandemic accelerated the rise of wellness technology with consumers investing in connected fitness systems (which have in turn created or redefined entertainment and recreation spaces in homes), while the pandemic brought indoor air and water quality into system-level discussions with consumers increasingly willing to invest in better quality of life.

Another interesting trend, mainly propelled by the product companies and system integrators, is the creation of a hybrid class of “residential systems.” This was a trend originally promoted by design and furniture suppliers, inspired by the idea to introduce home/residential elements in public buildings and corporate offices, creating a “homey” environment.

On the integration side, “residential” quickly evolved to describe systems that effectively combine networked solutions that are used in both commercial and residential systems, essentially serving the upscaled, luxury projects. With the pandemic, that trend accelerated with the introduction of communication systems for work-from-home. When spaces in the home became workspaces, the industry quickly recognized that they could reuse some elements of technology, and that those integrations would later expand to work-from-anywhere environments—which include the new-generation hybrid offices.

All these trends will create an intricate environment for the residential systems integration industry going forward. Most technologies will be increasingly connected for monitoring, automation, and control purposes and this paves the way to the introduction of Artificial Intelligence (AI) and machine learning (ML) on edge devices, which are no longer only buzzwords for the industry.

Introducing secure edge-AI smart-home systems is the cornerstone for broader adoption of voice recognition and voice interfaces. And while those AI-based systems remain connected strictly for control, they need to be self-sufficient and scalable, as consumers are increasingly aware of the privacy and long-term cost concerns associated with the original connected ecosystems from Google and Amazon.

But AI/ML is also increasingly at the core of home management systems, removing the complexity for consumers and creating a new level of efficiency—even interfacing directly with service providers in the background. And for that, the industry is well aware of the need for interoperability.

The Rising Matter

In this market environment, interoperability is now the key and something that was already emerging before the global pandemic. It all started when Amazon, Apple, Google, Huawei, LG Electronics, Oppo, and Samsung got together to create a new industry standard smart home “bridge” protocol, directly promoted by leading home automation and control companies such as ASSA ABLOY, Comcast, IKEA, Latch, Resideo, Somfy, Tuya, and Wulian, with the commitment from key power and lighting suppliers such as Legrand, Lutron

Electronics, Schneider Electric, and Signify, based on semiconductor technology from Infineon, NXP Semiconductors, Silicon Laboratories, STMicroelectronics, and Texas Instruments, among others. In fact, the interoperability effort for the smart home is now supported by another 267 companies and that number is rapidly growing.

The new effort is called Matter, and I briefly discussed its origins and the technology in our October 2021 Market Update. The important thing to reinforce is that Matter promises to be an industry-unifying standard for the Smart Home that promises reliable, secure connectivity—a seal of approval that devices will work seamlessly together.

The IEEE 802.15.4 technical standard is the foundation for Matter, a collaborative breakthrough built on proven technologies and guided by the Connectivity Standards Alliance (formerly Zigbee Alliance), superseding a series of overlapping and competitive efforts. Matter builds on existing networking technologies, such as Ethernet (802.3), Wi-Fi (802.11), and Thread (802.15.4) — and for ease of commissioning, also Bluetooth Low Energy (BLE).

A recent major research conducted by Parks Associates, published after CES 2022, showed that 36% of US broadband households owned a smart home device, and 45% reported that they intended to purchase a smart home device in the next six months. The research also showed growth in the proportion of homes with smart products and the number of devices per household—with an average of eight smart home devices.

And interoperability was considered a key issue as 73% of US broadband households that own or are planning to buy a smart home device considered interoperability with other products important to their purchase decision. This aligns with the goals of industry efforts, such as the new Matter initiative.

The Audio Industry Involvement

As I wrote in an article for *The Audio Voice* newsletter, the audio industry needs to get involved in this convergence and interoperability effort in order to avoid being restricted by specifications that were not written with audio in mind, as it happened with wireless streaming and network protocols and technologies that we can no longer avoid, such as Wi-Fi or Bluetooth.

Wi-Fi standards have drastically evolved with the introduction of Wi-Fi 6 and 6E in the last two years, with hundreds of products now commercially available. The Wi-Fi 6 standard is essential for the smart home, and Wi-Fi 6E promises to reduce RF interference in the home through the opening of the 6GHz band. At the same time, Wi-Fi standards continue to evolve to improve security, which is another cornerstone of the smart home. As many semiconductor companies such as Infineon, NXP, and Synaptics are promoting, Wi-Fi 6 allows devices that are much more power efficient and enable combined integrated solutions with Bluetooth and other wireless network technologies, which is also essential for Matter.

Residential integrators have been building structured cabling using Ethernet-based technologies for whole home audio and distributed entertainment systems for years, but consumers don't actually care about what they use. Residential audio installations have been basically about building stuff behind the walls and hidden on the ceiling, with only screens and (sometimes) loudspeakers



Matter originated from Project CHIP, within the Connectivity Standards Alliance (previously known as the Zigbee Alliance) an attempt to a single, unified, application-layer IP connectivity standard designed to enable developers to connect and build reliable, secure, connected ecosystems for the Smart Home.

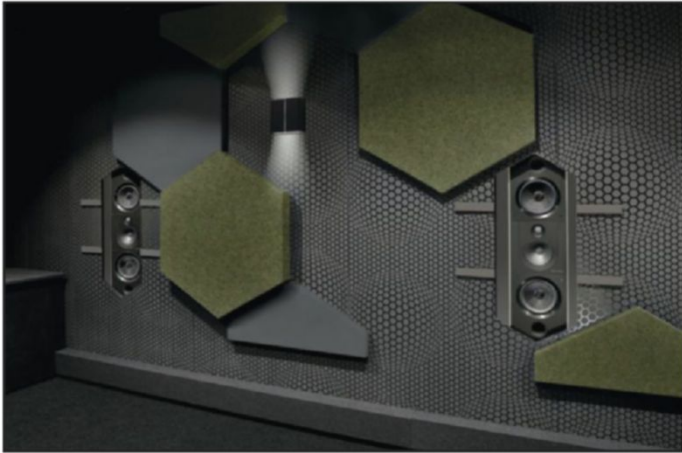
visible to the user. And when it comes to connecting those systems as part of a “smart home,” users want to wirelessly connect and control things from their own devices. And that fundamentally means Wi-Fi. Meanwhile, the Smart Home evolved so quickly that the integration business became mostly about configuring control interfaces and protecting access. And now, Matter will become another unavoidable layer where all the systems will have to coexist.

The audio industry is clearly at the end of the chain when it comes to offering interface solutions that match the content distribution services and fast broadband networks we get at home. If consumers don't understand when they play music over Wi-Fi or Bluetooth (and mostly don't care), how are consumer electronics manufacturers going to “sell” the concept of multichannel wireless audio for home theater, such as those from the Wireless Speaker and Audio Association (WiSA), DTS Play-Fi, or anything new? We still don't have a standard way to connect a wireless subwoofer from a brand to a soundbar from another brand. Now, add “spatial audio” reproduction to the mix, and suddenly the challenge seems insurmountable.

We now have soundbars that play out some sort of “3D sound” to improve the movie experience. Yet very few consumers even notice that the only reason why the soundbar is decoding all the original audio channels (and object-based metadata) is due to the fact that the TV (one of the consumer product categories with faster upgrade cycles) and the soundbar are both using the latest HDMI standard with eARC—which was specified in 2017, as part of the 2.1 standard.



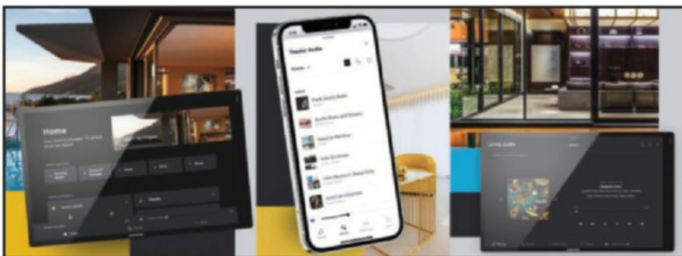
Frustration-Free Setup. A slide about the ongoing Matter efforts presented by Tuya at the Alexa Live 2022 event.



A home cinema designed with Bowers & Wilkins CI300 and CI600 custom installation speakers.



Lithe Audio, a UK-based manufacturer of wireless audio systems for residential and commercial installations, offers a lineup of all-in-one ceiling speakers featuring woven glass fiber woofer cones, titanium tweeters, and Class-D amplifiers, with support for all connectivity options, from Bluetooth 5 to Wi-Fi with support for AirPlay 2 and Chromecast, and two Ethernet ports, also available with PoE++ and WiSA bolt-on modules. To add extra bass, the 70W Wi-Fi Multiroom Micro Subwoofer can be wirelessly connected to Lithe Audio's Wi-Fi ceiling speakers and is small enough to fit under a bed or behind a couch.



Crestron, a company that is globally recognized as a pioneer in home automation, never stopped reinventing the smart home. The latest foundation in the company's strategy for residential integration is called Crestron Home OS, more than a software component, offering a full operating system specifically developed to integrate every Crestron product and third-party products and technologies as a single ecosystem. At the same time, Crestron Home OS enables the company to redefine its range of control products, which were once the cornerstone of its business.

And even though some early adopters and enthusiasts are already connecting their TVs and Apple TVs to multichannel wireless audio home theater systems, we have yet to define the best use of technology for wireless audio speakers—with WiSA being the best available option so far, but still restricted to surround sound and no support yet for immersive formats, such as Dolby Atmos or DTS:X.

If You Could Join Them...

Enter the Smart Home, where we have multiple devices that can be voice controlled. And we have connected IP cameras and smart locks controlled by Google or Amazon, lights and thermostats controlled from Apple devices using HomeKit, and Facebook pitching hardware for closely tunneled "family" video chat terminals that make sure we keep using Facebook or Instagram, while we share our home networks as part of Facebook Wi-Fi.

Are any of these devices and the latest technologies "talking" to our audio and entertainment systems? Of course! Through smart speakers, smart displays, or smart TVs that are control hubs, essentially designed to become the center of home systems. That is what the Apple HomePod Mini is all about. The problem is the vast majority of consumers will end up listening to music in the tiny mini HomePod speaker, instead of using it to control the Mark Levinson integrated amplifier and \$20,000 Revel speakers.

And why would they if that means sitting still... looking at the speakers? Younger generations will continue to look in disdain at their parents' living room hi-fi systems while they resort to wearing earbuds most of the time for their entertainment (if they are not the type to accept playing music through the smartphone speakers!...). And that right there, is reason enough why we need better standards for the Smart Home and why Matter is important for audio.

So, does Matter suddenly solve all of these problems? Well, it certainly creates an important foundation, much in the same way as Wi-Fi accelerated home networking. What is needed is to actually see how the hundreds of manufacturers that are supporting Matter address its use.

Unfortunately, many of the technology companies that have pioneered audio-centric technology efforts in the home audio space are not yet involved or have not reacted positively to the Matter initiative. Most leading CEDIA members are already involved in the Matter initiative, as are all the home automation, lighting, and home security players, but Sonos and Ikea are the only companies involved so far to represent the audio industry.

The Sonos example immediately stands out. The company was vocal in its fears that the Matter initiative could jeopardize its pioneering wireless streaming patented technology—which it has been defending vigorously in court against multiple audio companies and even the tech giants, Amazon and Google.

In a testimony to the US Congressional Subcommittee on Competition Policy, Antitrust, and Consumer Rights, Sonos chief legal officer Eddie Lazarus referred to Matter as "basically a creature of Google and Apple code." We can only hope that Sonos understands it can have a direct influence over Matter, and the new open source interoperability standard. Sonos just has to participate in its development to better defend its case—or not, and remain closed and proprietary.

And while we all wait for the Matter efforts to come to fruition, Amazon, Google, Facebook, and many other very large technology companies seem to be changing their smart home approaches. As an example, notice that these companies no longer talk about smart speakers and have been avoiding launching new hardware products. Amazon and Google are increasingly focused on perfecting their voice engine technology and applications, widening the margin that benefits their solutions against competitive non-connected efforts. While everyone waits for Matter, all these technology giants, including the Chinese companies, seem to be more focused on spreading their influence and reach.

There are some lateral initiatives in Thread technology, including the courting efforts by the Bluetooth Special Interest Group (SIG) toward DiiA, the global DALI alliance of companies from the lighting and sensor industries, trying to short-circuit Thread and Matter. The collaboration will enable the deployment of certified DALI-2 devices and intelligent D4i luminaires, with qualified Bluetooth mesh intelligent lighting-control networks, reinforcing the role for Bluetooth in that context, mainly for commercial lighting applications. This is an area that is still strongly disputed and from where Bluetooth could expand its relevancy outside its original consumer market.

Bluetooth mesh is the mesh networking capability of Bluetooth technology, designed for large scale device networks, where it competes with a number of other efforts in the IoT and building management space, most of which are well represented in the Matter alliance.

Audio and AV-Over-IP and PoE

The topic of wireless audio solutions is impossible to ignore when discussing residential integration. Supporting multiroom systems, AirPlay 2, Chromecast, and all the multiple streaming services is now mandatory for home audio systems and users expect a custom integration to offer all those options and more, in an elegantly integrated way. But the most common requirement for wireless audio is normally triggered by the need to add extra speakers or subwoofers to an existing installation, where adding wires is not desirable. But that will be a topic we will explore in detail in our next Market Update.

Another key technology trend in residential integration has to do with Audio-Over-IP or AV-Over-IP, since the acronym now also represents the growing range of network solutions that carry multichannel audio with at least one channel of video. The residential market originally adopted more or less proprietary solutions—mostly running on the physical Ethernet layer—and started to embrace Dante technology from Audinate, which was widely adopted for commercial audio and created economies of scale for the residential market as well. More recently, manufacturers started to embrace AES67, the simpler interoperability standard that conveniently bridges other audio-over-IP transport protocols, including Dante and Q-LAN from QSC.

A prime example of that trend comes from Crestron Electronics, a leading company in the residential integration market, which offers a range of DigitalMedia (DM) NAX all-in-one distributed AoIP amplifiers based on AES67. The company praises the “unmatched flexibility, scalability, and interoperability” of DM NAX for integrators, allowing

them to deliver complete control of the audio environment to users, from TV audio to streaming music, doorbells, and even paging and intercom usage to any speaker in the house. Crestron’s DM NAX systems integrate simply with legacy Crestron systems, including HDBaseT and DM NVX products and control systems, now expanded to up to 256 stereo zones (512 channels total) with interoperability with AES67 and Dante audio networking. Crestron is also able to combine native integration of music streaming services, Apple AirPlay 2, Spotify Connect, and more, all in energy-efficient units that can be continuously updated with remote firmware updates.

Going a step further, the Crestron Home OS smart home operating system was specifically developed to grow with these



The market growth in installation systems is attracting professional audio companies to experiment in new concepts that appeal to high-quality, high-power installations in both commercial and hospitality systems, but also for luxury residential integration. French manufacturer L-Acoustics is meeting the demands of that market with its latest Soka colinear source loudspeakers that are wall or flush mountable and is ideally configured with the companion subwoofer SB6i, an ultra-shallow 10cm design using two 6.5” woofers, able to extend the systems response to 33Hz.



Snap One announced the launch of the Triad PDX Series of architectural speakers. The new series is intended to provide integrators with a solution that pairs premium audio performance with the simplified installation experience made possible by Push Lock by Swarm technology. These are also the first Triad speakers launched into local stores and made available globally and mark the first Triad launch since the Snap One and Control4 merger in 2019. The Triad PDX speakers feature high-performance materials and the finest components like carbon fiber, Kevlar, Teteron high-strength polyester, dual voice coils, chemically etched metal components, and more.

installations and work flawlessly as a single ecosystem. This is key for Crestron and its network, which essentially sells the promise of a stressful smart home integration that will work as intended and will continuously evolve and actually get smarter over time.

Another important factor to reinforce the role of IP-based networked infrastructure for the home is the increasing integration and reduced clutter with lower costs. Residential integration—as with commercial installation systems—is increasingly based on devices that can be powered directly over Ethernet. Independently of what transport protocol is used for signals, Power-over-Ethernet (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.

Power over Ethernet 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 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).

It's extremely important that integrators leverage these existing trends and prepare their system's approach to scale and evolve toward the smart home convergence, which increasingly is what will determine the market segment.

The trends that we address in this Market Update all become visible when we observe a diverse portfolio such as the one from Legrand | AV, a leading vendor in the residential integration space, combining products from Legrand, Chief, Da-Lite, Luxul, Middle Atlantic, On-Q, SANUS, and Vantage brands, addressing audio, video, lighting, networking and infrastructure. The company is strongly betting on integration as the key to break into the thriving smart home systems market.

And the company also decided to phase out its only speaker and audio brand Nuvo, which it acquired in 2012, in order to enter the whole home audio market. Less than 10 years later, Legrand decided to phase out all Nuvo multiroom audio and loudspeaker product production, while promising to provide support over multiple years. The decision was directly motivated by supply chain challenges caused by the global pandemic, but also because Legrand realized it could expand its networked approach by working with third-party audio companies, instead of persisting with Nuvo's hardwired and powerline-based audio systems. And because Legrand also saw that market seriously menaced by wireless audio systems from companies, such as Sonos. Legrand even recommended that integrators explore alternative options from Bluesound (Lenbrook), Russound, Triad, or Sonos. Legrand has already established a multiyear licensing agreement with Sonos to integrate its Nuvo Player solution.



The first Lyngdorf Audio custom installation speakers, are the brand new D-500 and D-5 models for stereo, multiroom, and multichannel installations, including immersive configurations.

Lyngdorf Audio's engineers have developed the LS-1000 modular speaker system. Four LS-1000 modules combined create a line source speaker, with each module consisting of four 115mm midrange drivers with light aluminum cones, 25mm voice coils and powerful vented magnets, complemented by eight 1" soft dome tweeters. The Lyngdorf LS-1000 Left/Right models have a sound dispersion slightly beaming inwards when placed on the left or right side, away from the side wall and toward the listening area, while the Lyngdorf LS-1000 Center has the tweeters mounted centered, leading to a wider sound dispersion equal to the left and right. And there's also the Lyngdorf BW-20 (Boundary Woofer) designed to be placed near the wall boundaries.



Addressing today's wireless audio trends, Legrand also offers the SANUS WSS52 Sonos Five and Sonos PLAY:5 stands and WSSATM1 Extendable Soundbar TV mount designed for the Sonos Arc Soundbar. And its Luxul networking solutions are available as a backbone Wi-Fi mesh system for the custom installer, with remote monitoring and management of all connected network devices, while also offering a complete line of PoE+ managed switches.

As CEDIA stated in a recent article, "The core business for integrators will always be there: entertainment, infrastructure, networking, control, and AV distribution."

And yet, there's room for improvement on the core home audio technology and products, particularly speakers.

Speakers for Residential Integration

While most homeowners will select their speakers based on the concept that these can and should share the living space in the home—if for nothing else, because selecting home audio systems is a personal choice that all of us like to show and share with friends and family—the reality is that installation speakers can perform exceptionally well when integrated as part of a well-structured installation. When possible, they can be elegantly integrated inside walls and ceilings, and tuned to correctly cover all the living areas of the home, whether for background, multiroom music, for social engagements, personal or family entertainment moments. And those installations expand to home cinema/home theater rooms or designated areas, being able to deliver the full scope of signal reproduction, including immersive height channels and low frequency effect channels.

This is recognized by Lyngdorf Audio, the Danish manufacturer with ambitious plans for the residential audio space, and the brainchild of Danish entrepreneur Peter Lyngdorf. Lyngdorf Audio and its association with Steinway & Sons pianos symbolizes the pinnacle of luxury audio, extraordinary audio experiences, and the finest music rooms and private home cinemas. Lyngdorf Audio approach to product development is also focused on residential integration where its range of Multichannel Processors (MP) expanded the brand to new markets and demographics. Those products generated a lot of interest from the professional integration segment, which encouraged the brand to design a range of dedicated speakers for permanent installation. Using its experience in the development of the Lyngdorf Steinway installation speakers, the Lyngdorf Audio engineering team designed a range of seven in-wall and in-ceiling speakers for stereo and multichannel installations, to fit any type and size of room.

The new range includes the LS-1000, the D-500, D-5, and the BW-20 base models, each one bringing a unique concept on its own, for a total of eight models to suit different installations, from a modern living room with in-wall speakers next to the TV up to the largest home theater with a big screen and immersive audio.

The D reference stands for "discreet," as all models feature a wall-fixing mechanism with mounting legs and come with their own loudspeaker enclosure, making system planning and the actual installation as easy as possible. This also ensures that the sound performance is always defined and consistent, depending less on the wall construction and the available space behind the wall.



Luxury audio brand Bowers & Wilkins, now a part of Sound United, understands that the market is receptive to high-end solutions that fit the residential integration requirements. In 2022, the company introduced two in-wall subwoofers and a pair of powerful, DSP amplifiers that expand its portfolio of custom install solutions. The ISW-8 is now Bowers & Wilkins flagship in-wall subwoofer for completely installed applications. It's a no-compromise thin-profile design that is able to fit into a 2"x4" wall, while the ISW-6 is a more compact installation subwoofer with twin "racetrack" drivers arranged in a force-cancelling configuration. It is designed to work primarily with Bowers & Wilkins CI300 and CI600 speakers.



Leon Speakers offers a harmonious fusion of technology, design, and art via its Ente SoundTiles, and one-of-a-kind product that disguises a high-performance two-way speaker behind custom screen-printed artwork and a Sonos Amp, ideal for expanding multiroom audio solutions.



Now acquired by Origin Acoustics, Ambisonic Systems had recently expanded into the residential speaker market with its Landscape Ribbon Six (LS6) design, using a proprietary weatherized high-dispersion ribbon tweeter, voiced to expand an aligned woven glass-fiber woofer.

Origin Acoustics is a company that fully understands how important it is to offer a complete catalog of speaker solutions that respond to any custom-installation design. The manufacturer was founded in 2015 by the original team that, decades ago, helped promote the in-wall speaker category to completely new quality levels, producing speakers for Sonance, JBL, Bang & Olufsen, Niles



Crestron Electronics is expanding its increasingly comprehensive suite of residential audio solutions and Crestron Home ecosystem. In 2022, the company developed the DM NAX 4-Zone Streaming Pre-Amplifier for high-performance AoIP installations, in partnership with Origin Acoustics. Slimmer in form (1RU) to the DM NAX Amplifier, the preamplifier natively integrates with Crestron Home, and is interoperable with AES67, and Dante audio networking.



The Crestron Ultimate In-Wall Subwoofer represents the latest addition to Crestron's lineup of home audio solutions, joining an existing array of more than a dozen architectural and landscape speakers from Crestron, including its Ultimate or Reference lines. The Crestron Ultimate In-Wall Subwoofer was designed to create an immersive sound experience for demanding multiroom audio settings or home theater applications, and it features paintable grilles. Installation is streamlined to reduce interference with other elements present in the wall space.

Audio, and many others, joining forces with Jeremy Burkhardt, the engineering visionary who previously turned SpeakerCraft into a global market reference.

Since its appearance in the global installation and residential integration market, Origin Acoustics never stopped expanding its range of completely original designs for all needs, from background music duties to immersive channel ceiling coaxial speakers.

Just recently, Origin Acoustics introduced two new two-way coaxial speaker models to its Director Collection of premium architectural loudspeakers—the D101 and the D103DT, ideal for home-theater environments where high SPL performance is preferred. Bigger in performance, but not in size, the two new 10" loudspeakers use Origin's unique mounting system structure, which allows looking like having smaller 8" units on the ceiling. While the aesthetics will please both client and system designers, the larger-sized loudspeaker performance will stand-out in larger rooms. These new in-ceiling 10" models also allow designers to take advantage of high ceilings and extended frequencies to produce more powerful lows, as increasingly required by distributed audio and home theater applications.

Using high-quality drivers and materials, the D101 is a two-way loudspeaker with a 1" silk-dome tweeter and 10" polypropylene/butyl-rubber woofer. The D103DT features a dual aluminum tweeter configuration, specifically for a stereo signal, and an injection-molded graphite woofer.

And when Origin Acoustics feels the need to offer completely different technologies, the company offers distribution deals to specialized manufacturers, and eventually even acquires those companies. That was the case with luxury pro and residential loudspeaker manufacturer Ambisonic Systems, which Origin originally distributed, and decided to acquire in 2021. Ambisonic was founded in 2009 by industry pioneer Jeff Coombs and the company specializes in high-output, audiophile-quality sound reinforcement loudspeakers using ribbon planar drivers. The full range of Ambisonic Systems high-performance loudspeakers includes a vast line of outdoor audio solutions for any situation or location, including custom outdoor soundbars, weatherized garden speakers, triple-bandpass burial subwoofers, and more.

This reflects the fact that the industry has been embracing completely new form factors and less conventional applications for installation speakers, which fully recognize consumer trends. That includes the use of large-size soundbars to serve as the under display solution for entertainment centers—which need to be able to deliver both live TV broadcasts such as sports, cinematic content, and increasingly gaming. And since home audio systems are increasingly also used as an extension of work-from-home and communication systems, the industry has been introducing new form factors merging the best of smart speakers and compact meeting/room type soundbars. An area that Sonos greatly helped to inspire and that many manufacturers now emulate but with discreet, installation-focused designs.

I will just mention a few examples that have recently been introduced to the market. In an effort to inspire residential integrators to explore new design-forward solutions that blend performance with modern style, Leon Speakers offers a broad

range of functional concealment solutions for everything from videoconferencing systems and loudspeakers to displays, increasingly required for the aesthetics of luxury residences and “resimercial” corporate office spaces. Establishing key strategic integration solutions with brands such as Bose, Logitech, Sonos, and Samsung, Leon Speakers simplifies and streamlines the deployment of a wide variety of high-design, high-performance audio and video systems. An example would be the Tonecase FIT Universal mounting system and Horizon Interactive FIT soundbar, which mount directly to a display and are built to integrate off-the-shelf teleconferencing equipment or soundbars. The Horizon Interactive FIT includes two channels of high-fidelity Leon speakers, and complementing both products is a grille that conceals the equipment inside and blends seamlessly with the display.

Leon also introduced its Ente SoundTiles and Edge Media Frame, the first disguising two channels of high-performance audio behind custom screen-printed artwork and the second a contemporary, custom-built and finished frame that mounts and blends a TV into any space. The Ente SoundTiles is a solution that Leon promotes in combination with the Sonos Amp to create a plug-and-play zone in rooms and spaces in the home, where having the speakers invisible in artworks is ideal. Both the speakers and the Sonos Amp fit entirely in the frame.

When making the speakers completely integrated and invisible might be an approach to explore, for design and aesthetics, Amina Sound, the British manufacturer of invisible loudspeakers founded by former NXT team members and Distributed Mode Loudspeaker (DML) pioneers are known for creating the world’s first plaster over invisible loudspeaker in 2001. With an increasing need for clean, clutter-free integrated lifestyle-technology in both commercial and residential applications, the requirement for discreet, invisible audio solutions is in demand.

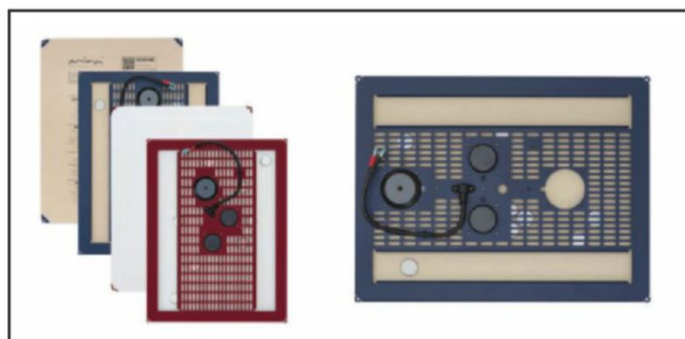
Amina’s technology interacts with the room, with extremely linear 180x180 degree all around dispersion characteristics. With the invention of Vibrational Panel Technology (VPT), Amina has been able to take the common loudspeaker design and replace the cone with a flat panel, using small vibrations to generate sound, which can be heard but not seen. This allows for seamless integration of

a loudspeaker into any wall, ceiling, or flat decorative panel with zero visual impact. Amina offers two core-technologies—Edge and Mobius—to allow integration of invisible loudspeakers into any room. Both products can be installed in either new or retrofit applications.

With cavity walls and ceilings being a staple of the construction industry globally, Amina’s Edge Series loudspeakers provide a high-performance solution in a lightweight and easy to install package where cavity walls and ceilings have been used.

Where walls or ceilings will receive a full plaster skim, for retro-fit or for where there is a specialist covering material, Amina’s Mobius Series loudspeakers provide a completely invisible alternative.

The advantages of using the technology are noticeable for immersive audio due to the incoherent-phase waveform nature of the VPT technology. This means that the acoustic energy generated by the panel interacts in a summative way with the room reflections, leading to a room-filling sound rather than the typical hot spots and tonal coloration associated with direct phase correlation. A VPT loudspeaker has the benefit of full 180° audio dispersion in almost full bandwidth, which means that it is possible to create quality sound installations in rooms and make it fully invisible. 



Loudspeaker frames from Amina Sound, the British manufacturer known for creating the world’s first plaster over invisible loudspeaker in 2001. With an increasing need for clean, clutter-free integrated lifestyle-technology in both commercial and residential applications, the requirement for discreet, invisible audio solutions is in demand.

Resources

“All Apps Aren’t Equal: Smart Home User Experiences,” Parks Associates, www.parksassociates.com/blog/article/pr-01132022

Connectivity Standards Alliance, www.buildwithmatter.com

J. Martins “The Standards We Need: Do Audio Technologies Matter in the Smart Home?” *The Audio Voice*, January 27, 2022, <https://audioxpress.com/article/the-standards-we-need-do-audio-technologies-matter-in-the-smart-home>

M. Potuck, “Sonos urges regulation of big tech, criticizes Apple’s plans to open Siri to third-parties using HomePod,” June 2021, 9TO5Mac, <https://9to5mac.com/2021/06/15/Sonos-urges-regulation-of-apple-and-big-tech>

The Digital Illumination Interface Alliance (DiiA), www.digitalilluminationinterface.org

Sources

Ambisonic Systems | www.ambisonicsystems.com

Amina Technologies, Ltd. | www.aminasound.com

Creston | www.creston.com

L-Acoustics Group | www.l-acoustics.com

Legrand AV Inc. | www.legrandav.com

Legrand North America, LLC | www.legrand.us

Leon Speakers www.leonspeakers.com

Lithe Audio | www.litheaudio.com

Lyngdorf Audio | www.lyngdorf.steinwaylyngdorf.com

Origin Acoustics | www.originacoustics.com

Snap One, LLC | www.snapone.com

Sonos, Inc. | www.sonos.com

Triad Speakers | www.triadspeakers.com

Sumiko Phono Cartridges and RS78 78 rpm Stylus

By
Gary Galo

Sumiko recently announced four new phono cartridges and one new stylus. This article reviews the new Sumiko Olympia and Wellfleet phono cartridges, and the new RS78 stylus, Sumiko's first 78 rpm solution, meeting the needs of many vintage recording enthusiasts.

Sumiko is a company well-known to audiophiles. It's been a leading manufacturer and importer of high-end audio equipment for more than 40 years. Back in the 1980s, Sumiko imported an excellent line of Grace tonearms, notably the 707 and the 747 models that were praised by the audiophile press. I owned a 747 in the early 1980s and found it to be a very fine arm when used with low-tracking force, high-compliance cartridges. The company has also manufactured phono cartridges, including several incarnations of the Blue Point moving-coil designs that are available in both high- and low-output versions. Sumiko also has a line of eight moving-magnet cartridges, all part of its Oyster series. The Oyster series goes back to the first Sumiko phono cartridges, introduced some 40 years ago. It still offers an entry-level Oyster model, a classic design that sells for \$79 US. The models reviewed here (**Photo 1**) are part of a group of four recent designs that feature interchangeable styli. The four cartridges range in price from \$79 to \$449 US, and

also include the Rainier (\$79) and Moonstone (\$299). Sumiko is now a division of the McIntosh Group.

Traditionally, audiophiles and the high-end press have favored moving-coil cartridge designs over their moving-magnet counterparts. Moving-coil designs generally have much lower internal impedance than moving-magnet types, which results in treble extension an octave or more above 20kHz. The higher moving mass of the stylus and coils often puts the high-frequency resonance point in the upper audible octave, which gives moving coils the sense of "air" that many audiophiles favor, and their low internal impedance makes them immune to the effects of capacitive loading. In past decades, many reviewers in high-end publications took a negative view toward moving-magnet cartridges, but in recent years that has changed. The resurgence of interest in vinyl has brought with it a renewed interest in moving-magnet cartridges. Moving coil designs still flourish in vinyl circles, but newer moving-magnet designs have earned a renewed respect in the vinyl community.

Sumiko Olympia Phono Cartridge Wellfleet Phono Cartridge RS78 78 rpm Stylus

Sumiko
63 95th Ave. N
Maple Grove MN 55369
510-843-4500
sumikophonocartridges.com
service@sumikoaudio.net
Olympia Phono Cartridge: \$199.00
Olympia Stylus: \$119.00
Wellfleet Phono Cartridge: \$449.99
Wellfleet Stylus: \$339.00
RS78 Stylus: \$129.99

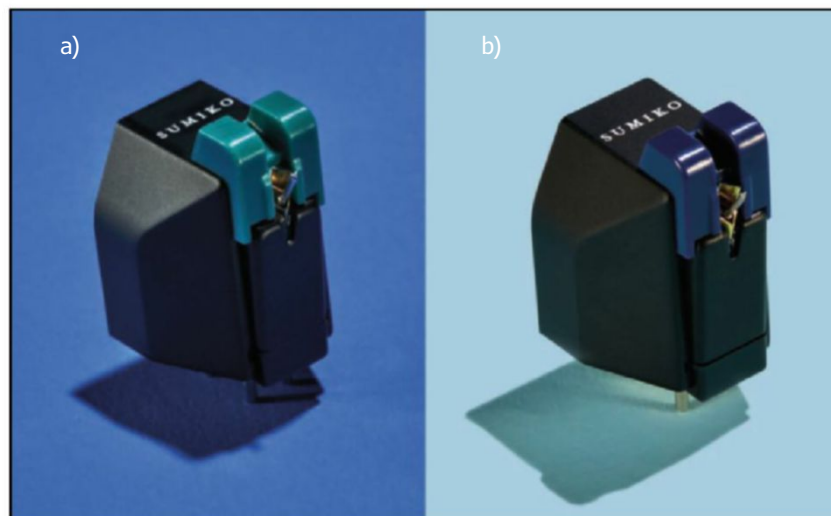
Upgradable Styli

The cartridge housings (bodies) of the Rainier, Moonstone, Olympia, and Wellfleet models are identical, so upgrades to the next level can be made simply by replacing the stylus. Sumiko notes that the housing has been designed to disperse and dissipate resonances, "making way for effortless, transparent sonics that are enlivened with intimate nuance." Listeners are said to "experience a diminutive backdrop that bolsters sonic images with a degree of aptitude reminiscent of moving coil designs."

Specifications for the Olympia and Wellfleet phono cartridges, as well as the RS78 stylus, are shown in **Table 1**. The table gives 6kHz as the -3dB point for the Olympia and RS78; and -2.5 dB for

the Wellfleet, which have to be typos. As of this writing, I've not been able to get clarification from Sumiko on this specification. I requested frequency response graphs for both models; strangely, Sumiko was unable to supply them. Both models have an elliptical stylus, 0.3 × 0.7 mils.

The main difference between the two models is in the cantilever and the bonding of the stylus. The Olympia has a cylindrical aluminum cantilever with a conventional bonded stylus. With conventional mounting, the diamond tip is bonded to a short metal shank which, in turn, is bonded to the aluminum cantilever. The more expensive Wellfleet has a nude diamond stylus, which means the diamond is bonded directly to the cantilever, without any intervening material. The result is lower mass and improved tracking. This type of mounting is considerably more expensive to manufacture than the type used in the Olympia mounting. The Wellfleet cantilever is a special aluminum pipe designed for improved resonance control. Sumiko claims "a faster, more direct, and more accurate response of the stylus's movement by the cantilever and magnet. Sonically, this increases speed and focuses central images, pulling them toward the listener with a heightened



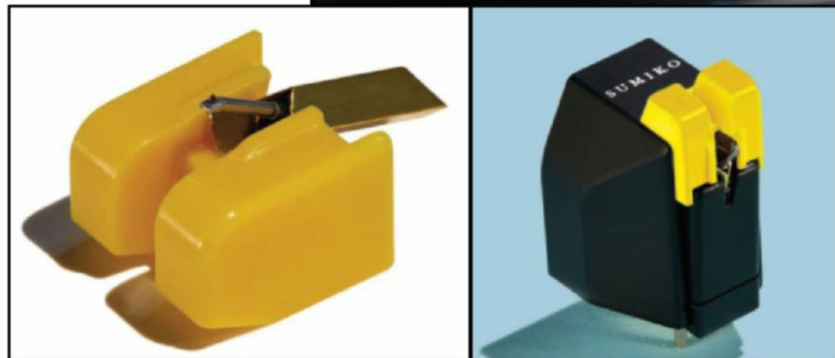
sense for detail... Listeners will hear an improvement in surface noise and sense a more distanced backdrop, facilitating a deeply suspended image." Sumiko also makes an Amethyst model which sells for \$599, and looks similar to the other four models. But, the Amethyst stylus is not interchangeable with the others. The cartridge body has a much lower internal impedance and the stylus is a nude, line-contact type. Note that while line-contact styli can deliver superior performance to conventional elliptical styli, they are more sensitive to vertical

Photo 1: The Sumiko Olympia (a) and Wellfleet (b) phono cartridges are part of a line of four that have interchangeable styli. The cartridge bodies are identical, allowing upgrades by replacing the stylus. (Courtesy of Sumiko Audio)

	Olympia	Wellfleet	RS78
Cartridge Type	Moving magnet	Moving magnet	Replacement stylus
Stylus Chip & Size	0.3 x 0.7mil Elliptical	0.3x0.7mil Nude Elliptical	3mil bonded conical
Cantilever material	Aluminum pipe	0.5mm Aluminum pipe	Aluminum pipe
Magnet	Alnico	Alnico	Alnico
Coil Wire material	Pure Copper Wire	Pure Copper Wire	Pure Copper Wire
Internal Impedance	1130Ω	1130Ω	1130Ω
Load Impedance	47kΩ	47kΩ	47kΩ
Frequency Response	12Hz-30KHz	12Hz-33KHz	15Hz-25KHz
Output Voltage	4mV	3mV	4mV @1kHz
Channel Separation 1kHz	30dB	30dB	25dB
Channel Separation at 15kHz	15dB	15dB	15dB
Channel Balance	1.5dB/1kHz	0.5dB/1kHz	1.5dB/1kHz
Compliance at 100Hz	12 x 10-6cm/dyne	13 x 10-6cm/dyne	10 x 10-6cm/dyne
Tracking Force	1.8-2.2 grams	1.8-2.2 grams	4-5.5 grams
Recommended Force	2.0 grams	2.0 grams	5.0 grams
Cartridge Weight	6.5 grams	6.5 grams	6.5 grams
Replacement stylus unit	yes	yes	yes
Frequency range at -3db	6kHz(-3dB)	6kHz(-2.5dB)	6kHz(-3dB)
Tracking ability at 315 Hz/ recommended tracking force	60µm/2 grams	70µm/2 grams	60µm/2 grams
DC Resistance	510Ω	510Ω	510Ω
Internal Inductance	450mH	450mH	450mH
Recommended load capacitance	100-200pF	100-200pF	100-200pF
Vertical tracking angle	25 degrees	26 degrees	25 degrees

Table 1: Manufacturer specifications are shown for the Olympia and the Wellfleet phono cartridges and the RS78 stylus.

Photo 2: The RS78 is a 3.0 mil conical stylus made for 78 rpm records. The stylus is only sold separately, but fits the Rainier, the Olympia, the Moonstone, and the Wellfleet models. (Courtesy of Sumiko Audio)



tracking angle (VTA). Individual VTA adjustment may be required for different records.

The Sumiko models reflect two trends in phono cartridge design that have emerged during the LP revival. First, they're less sensitive to capacitive loading than their predecessors. It was fairly common in past decades for moving-magnet cartridge manufacturers to give a specific recommendation for the capacitive load, usually between 100pF and 300pF. These days, most moving-magnet cartridges will operate within a fairly wide range of capacitive loads. The Sumiko models will perform to specification anywhere between 100pF and 200pF, which should accommodate most high-performance tonearm cables and preamp inputs. Second, the recommended tracking forces are higher than cartridges from the 1970s and 1980s. Back then, many moving-magnet cartridge manufacturers placed an emphasis on excellent tracking at very low tracking forces, with typical manufacturer recommendations in the range of 1 gram to 1.5 grams. The legendary Shure V15 series led the pack in this area. But, many audiophiles and high-end reviewers found that ultra-low tracking forces came with a price—two-dimensional soundstage reproduction, flat dynamics, and a lack of high-frequency extension and air.

These days, one look at the specifications for most moving-magnet cartridges shows that priorities have shifted. Tracking forces for most audiophile-quality moving-magnet cartridges are typically in the range of 1.5 grams to 2.5 grams, comparable to

many moving-coil designs. All four Sumiko models with interchangeable styli are rated from 1.8 grams to 2.2 grams, with 2.0 being optimum. The trade off has paid dividends, resulting in greater acceptance of moving-magnet designs in the audiophile community. Today's moving-magnet designs generally have soundstage reproduction, dynamics, detail, and high-frequency extension that are much improved over their predecessors. They also work well in many of the same medium-mass tonearms compatible with moving-coil designs. The general view in the audio industry is that tracking forces around 2 grams do not cause excessive record wear.

78 rpm Stylus

The most recent addition to the Sumiko line of replacement styli is the RS78, a stylus made especially for 78 rpm records (**Photo 2**). The RS78 features a bonded, spherical (conical) 3.0 mil diamond stylus and a cylindrical aluminum cantilever. Normally, 78 rpm records require higher tracking forces than vinyl LPs, and the RS78 is rated from 4.0 grams to 5.5 grams, with 5.0 grams recommended. The RS78 fits all four models in this line, and the stylus is available only as a separate purchase. You'll need to buy one of the four cartridges with an LP stylus in order to use it.

The description of the RS78 on the Sumiko website has a one statement that might confuse the reader. In the first paragraph the company notes that the RS78 has a spherical diamond stylus. But, in the next section it states that it's "not advisable to play 78rpm records with a diamond stylus." What they mean is "diamond LP stylus," and Sumiko later clarifies matters by correctly stating that the microgroove styli made for 33-1/3 and 45 rpm records will sound poor and can cause damage to 78 rpm records. Sumiko notes that since the four Oyster cartridges meant to be used with the RS78 are stereo, the RS78 will still yield a two-channel output. But, since 78s are mono, the same information will be heard in both channels.

Mono vs. Stereo Cartridges

There's a school of thought that monaural records sound best played with a monaural cartridge. Some cartridge manufacturers are making mono cartridges specifically for playing mono LPs. In a perfect world, I might agree with this. But, the two groove walls of a monaural record are often not absolutely identical. This is as true for 78s as it is for mono LPs. There are mechanical limitations in the cutting and the pressing process that often prevent the groove walls from being absolutely identical, and wear patterns on older

records are often dissimilar in the two groove walls. Beginning in the early 1970s, many monaural records were being mastered on stereo cutters. This virtually ensures that the two groove walls won't be identical. Most preamps made for archival use have a mono mix control—sometimes called a canting control—that allows the optimum mix of the two groove walls. Most audio transfer and restoration engineers prefer to do a stereo transfer to digital, followed by digital azimuth correction and mono sum. In short, I agree with Sumiko's decision to make the RS78 function as a stereo pickup.

My main musical interest has been classical since I began collecting recordings in junior high school. Although I am an audiophile with a serious interest in high-end reproduction of stereophonic recordings, much of my collecting and listening has always been focused on historical, pre-LP material, the era in which many of my favorite artists performed and recorded. I've been playing 78s, and building better mouse traps to get the most from them, since I was a kid. Like most 78 collectors, for many years I used a one-size-fits-all 78 rpm stylus, usually conical types with sizes in the 2.7 mil to 3.0 mil range.

Around 1980, as I become more serious about getting the most out of my vintage records, I began using truncated conical and elliptical styli in various sizes sold by Owl Audio Products, custom made for them Stanton Magnetics for their 500AL cartridge. Owl is long out of business, but at one time it stocked more than a dozen truncated styli in various sizes ranging from 2.0 mil to 6.0 mil, to accommodate vintage commercial records dating from the turn of the last century through the end of the 78 era in the early 1950s, as well as broadcast transcriptions and various "instantaneous" recordings. Many more stylus sizes were available on special order. Truncated styli aren't rounded on the bottom like conventional styli. Instead, they're squared off at the bottom, with rounded and polished corners. The truncation prevents the stylus from dragging the bottom of the record groove, which always increases noise. They allow precise matching of the playback stylus to the



groove size. Record groove sizes varied during the 78 era. The earliest 78s have a much wider groove than those made in the 1940s. I often use styli in the 3.3 mil to 3.8 mil range for records made before 1920, yet post World War II discs often play best with a 2.5 mil stylus. Having a variety of truncated styli also allows you to pick a size that plays above or below the wear area, if the record is worn. Since the demise of Owl, high-quality truncated styli have been increasingly difficult to find.

For many collectors, there's still a need for a high-quality, one-size-for-all 78 rpm stylus, with 3.0 mil usually providing a good compromise that will accommodate most records. Sumiko initially sent me an Olympia LP cartridge, so I evaluated the RS78 stylus in that cartridge body. I used my archival turntable for the evaluations, a Technics SP-15 on a custom base fitted with a Jelco SA-750L tonearm (**Photo 3**).

The preamp I used for archival work is my own custom-built Archival Phono Preamp (**Photo 4**), jointly designed by myself and D. Michael Shields, an engineer based in Minneapolis, MN. I described this preamp in an article I wrote for *Linear Audio* back in 2012 [1]. The preamp has separate bass and treble phono equalization adjustments to accommodate the myriad equalization curves found on records

Photo 3: This is the author's archival playback setup, a Technics SP-15 mounted on a custom base and fitted with a Jelco SA-750L tonearm. The SP-15 features variable pitch of up to $\pm 9.9\%$ in 0.1% increments, and the extra-long tonearm will accommodate 16" and 17" transcription discs. The Sumiko Olympia cartridge with the bright yellow RS78 stylus is mounted in the headshell.

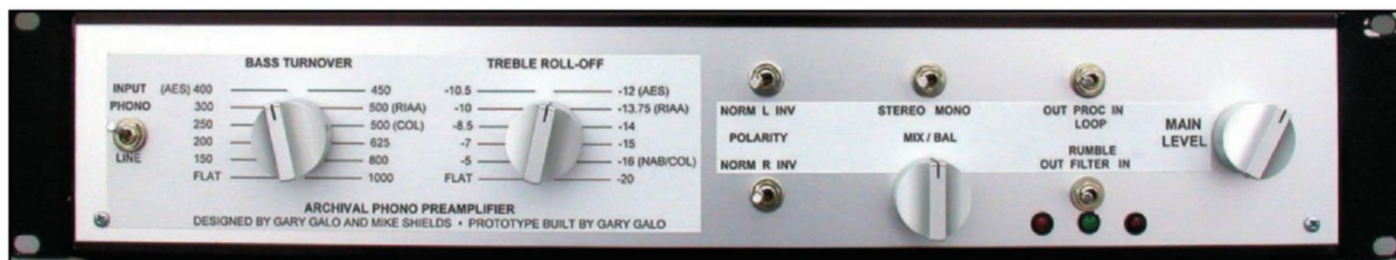


Photo 4: This the author's custom-built Archival Phono Preamp. The 12 bass turnover frequencies and 12 treble roll-off settings accommodate the myriad equalization curves found on pre-RIAA records. The preamp also includes switching for laterally and vertically cut discs, adjustable mix of the two groove walls, and a rumble filter.

Photo 5: The Suisse pianist Alfred Cortot recorded Chopin's *Ballade* in F minor for HMV electrically in 1929. The record offers surprisingly vivid reproduction of Cortot's Pleyel piano, and most connoisseurs of piano recordings consider the performance to be better than his remake of the same work in 1931.



made before the RIAA curve became the industry standard. There are 12 bass turnover and 12 treble roll-off settings, including flat positions for playing acoustical recordings. Polarity can be inverted in either channel, which is required to switch between laterally- and vertically-cut records. A mix/balance control allows adjusting the mix of the two groove walls, and the preamp also includes a 30Hz, 18dB/octave rumble filter and an external processor loop.

Records for Evaluation

I put the RS78 stylus through its paces on a variety of 78 rpm records, and I've included a selective list in this review. I've indicated the exact speeds used for each record. This is because, prior to the 1930s, "78 rpm" was just a ballpark figure. Before electric motors became standardized in disc recording and playback equipment, record speeds varied, often considerably, from the nominal value of 78 rpm. A variable-speed turntable is

essential for the serious 78 collector in order to correctly reproduce the exact pitch at which the original performance was done. Prior to 1925, all records were acoustically recorded. It was a purely mechanical process in which the performers projected into a large horn which collected the sound, feeding it to a vibrating diaphragm that cut the record groove. Western Electric began its development of electrical recording several years before the system was finally deemed ready for commercial use. It was a natural extension of their ongoing research on telephone transmission [2]. A microphone turned the sound vibrations into alternating current which, in turn, was amplified by a vacuum tube and fed to a magnetic cutter head. By 1925, the system had improved enough to be marketed commercially, and by the following year nearly all record companies had converted to the electrical process. I have noted which process was used for each of the recordings I list.

The vast majority of acoustical disc records were laterally cut but a few labels, notably Edison and Pathé, made vertically-cut discs (both companies also made cylinders, which were always vertically cut). In order to play both types of records with modern electrical equipment, a stereo cartridge is essential. For laterally cut records, the left and right channels are summed to mono "in phase" (same polarity, to be technically correct). This sums the lateral musical information and cancels the vertical component, which is primarily noise. For vertically cut records, one channel must be inverted in polarity before summing to mono. This sums the vertical musical information and cancels the lateral component which, again, is mostly noise. Note that vertically-cut records can't be played with a conventional monaural cartridge, and dedicated vertical cartridges haven't been available in decades.

RS78 Performance

Label scans for a few of the records I used for evaluating the RS78 are shown in **Photos 5–8**. I found the RS78 to be an excellent all-purpose stylus for 78 rpm records. There's a school of thought that because 78 rpm records pre-date the high-fidelity era, any magnetic cartridge with an appropriate stylus will be sufficient. Serious collectors and restoration engineers know better! There's far more information in those vintage record grooves than the original playback equipment could possibly extract. Over the many decades I've been playing 78s, I've experienced better sound from those records with every improvement in playback equipment, whether the cartridge and styli, the tonearm, the turntable itself, or the preamp.

Photo 6: The French tenor Lucien Muratore made a number of vertically cut records for Pathé. This disc features one of the staples of his repertoire, an aria from Gounod's *Roméo et Juliette*, recorded in 1918. Pathé verticals can be difficult to track, but the Sumiko RS78 stylus had no problems playing it.



The Sumiko cartridge fitted with the RS78 stylus shows the benefits of using an audiophile-quality cartridge with 78s. The RS78 offers a level of detail and definition across the frequency range of these records that lesser 78 cartridges fail to offer. Where cheaper 78 cartridge/stylus combinations may sound closed down in the treble region, the RS78 shows that the fault is not always on the records, but in the playback equipment. It's true that having custom, truncated styli tailored to the myriad groove sized encountered throughout the 78 era will yield the optimum results. But, the RS78 did a fine job on all of the records I auditioned.

Vertically cut discs can present unique problems. Pathé discs have a wide and rather shallow groove have a wide and rather shallow groove, and are notorious for being difficult records to track. The Pathé discs I tried with the RS78 played without problems, but anyone attempting to play these records should pay careful attention to tonearm adjustments. Skipping can often be corrected with careful anti-skate adjustment. Cheap tonearms and cartridges often fall flat with vertically cut Pathé discs. Owl made a special 3.7 mil glass stylus for Edison Diamond Discs (so-named because Edison's phonographs had a diamond stylus in a floating reproducer to minimize record wear). But, I've found that a stock 3.0 mil conical stylus usually works well with these discs, and the RS78 lived up to expectations. The clean treble region of the Sumiko cartridge with RS78 had no difficulty revealing the superiority of Edison's recording process compared to those of his competitors.

Shellac-based 78 rpm records are often noisy, and much of that noise is caused by the abrasive fillers that were necessary in order to ensure that the record wore the heavy steel "needles" rather than vice versa. The Victor Talking Machine Company's formula for its record compound in the 1920s was 75% abrasive filler, an equal mix of powdered Pennsylvania slate and Indiana limestone [3]. Shellac was the binder that held it all together, and was only 13.6% of the compound. Fortunately, the metal parts that were used to press shellac records often survive in record company vaults. When they're in good condition, they can be used to press vinyl records that are far quieter than the original pressings. I have many in my collection and used several in my evaluation of the RS78. The RS78 did an excellent job with these pressings, easily revealing their superiority in overall sound and lower noise, to original shellac pressings of the same recordings.

Collectors are often faced with the difficulties of tracking warped shellac records. I have a couple



of badly warped 78 pressings that I keep on hand, to test the ability of tonearms and cartridges to deal these problems. Playback of a warped 78 will usually fail for one of two reasons. First, the warp can throw the arm upward causing the stylus to lose contact with the record groove. This is not the fault of the cartridge or stylus. The inertia of the tonearm causes the arm to continue in the direction it was sent. The Jelco SA-750 arms have fluid damping in the pivot, which helps, but there are limits on to the amount of damping fluid that can be used without compromising the playback of non-warped records. Second, the cartridge will bottom out when the high side of the warp hits the cartridge body—78 rpm cartridges with a short stylus cantilever are prone to the second problem.

I tried one of my worst warp cases on the RS78 tracking at 5.5 grams, the manufacturer's upper limit. The stylus lost contact with the record

Photo 7: The vaults of many major record companies house original metal parts used to press shellac 78 rpm records. If the metal parts are in good condition, vinyl pressings can be made from them, resulting in much quieter playback than original shellac-based discs, and free of the wear sometimes encountered on vintage records. This recording was made in 1904, and was the very last record made by the Italian heroic tenor Francesco Tamagno. The modern vinyl pressing offers amazingly vivid sound, capturing a powerful voice that cut through the limitations of the primitive recording equipment.



Photo 8: Electrical recording made it possible to credibly convey the sound of a symphony orchestra. This electrical recording of the Prelude to Act I of Wagner's *Lohengrin*, made in 1930, features the Berlin Philharmonic Orchestra conducted by the legendary German conductor Wilhelm Furtwängler, capturing the unique sound of that orchestra during this period. Polydor was the export label of the German Gramophon company.

Recordings Used for Evaluation (Selective List)

78 rpm Original Shellac Pressings

Buzzi-Peccia: "Lolita" (Spanish Serenade). Enrico Caruso, tenor. Victor 88120, Recorded in 1908, 76.6 rpm. (Acoustical recording)

Chopin: Ballade in F-minor, Op. 52. Alfred Cortot, pianist. HMV D.B. 1346, Recorded in 1929. 77.5 rpm. (Electrical recording).

Gounod: *Roméo et Juliette*—"Ah! leve-toi soleil." Lucien Muratore, tenor. Pathé 64006B, Recorded in 1918, 73.5 rpm (Vertically cut; Acoustical recording).

Prokofiev: *Classical* Symphony in D, Op. 25. Boston Symphony Orchestra conducted by Serge Koussevitzky. RCA Victor set DM-1241, Recorded in 1947, 78.26 rpm (Electrical recording).

Wagner: *Lohengrin*—Prelude to Act I. Berlin Philharmonic Orchestra conducted by Wilhelm Furtwängler. Polydor 95.408, Recorded in 1930, 78.26 rpm (Electrical recording).

Wagner: *Rienzi*—"Allmächt'ger Vater." Jacques Urlus, tenor. Edison Diamond Disc 82269, Recorded in 1922, 77.0 rpm (Vertically cut; Acoustical recording).

Weber: *Oberon*—Overture. New York Philharmonic Orchestra conducted by Willem Mengelberg. Victor 74766 & 74767, Recorded in 1922, 75.00 rpm (Acoustical recording).

78 rpm Vinyl Re-Pressings from Original Metal Matrices

Bach: *Fantasia and Fugue* in C-minor, BWV 537, arr. Elgar, Op. 86. Royal Albert Hall Orchestra conducted by Sir Edward Elgar. HMV Matrix Nos. CR 346-IIA (previously unpublished) & CR 347-1A. Repressing by Symposium, 1025, 75.0 rpm. (Electrical recording).

Chopin: Ballade in A-flat, Op. 47. Sergei Rachmaninoff, pianist. Victor Matrix Nos. CVE-32510-1 & 32511-1. Recorded in 1925. Vinyl pressing made for transfer to LP and CD releases, previously unpublished, 75.0 rpm. (Electrical recording).

Giordano: *Andrea Chenier*—"Un di all'azzurro spazio." Francesco Tamagno, tenor. Gramophone Monarch 052100, Recorded in 1904. Repressing by Historic Masters HM-FT7B. 73.0 rpm. (Acoustical recording).

Verdi: *Aida*—"Pur ti riveggo...Fuggiam gli ardori inospiti." Melanie Kurt, soprano & Jacques Urlus, tenor (sung in German). Gramophone 2138c & 2139c. Recorded in 1911. Repressing by Historic Masters. 80.0 rpm (Acoustical recording).

Stereo LP Records

Dukas: *The Sorcerer's Apprentice* (in the album *The French Touch*). Boston Symphony Orchestra conducted by Charles Munch. RCA Victor Living stereo LSC-2292. Recorded in 1957, Orchestra Hall, Chicago, IL. Recording Engineer: Lewis Layton, Producer: Richard Mohr. Reissue by Analogue Productions.

Holst: *The Planets*. Boston Symphony Orchestra conducted by William Steinberg. Deutsche Grammophon 2530 102 (Germany). Recorded in 1970, Symphony Hall, Boston, MA. Balance Engineer: Günter Hermanns, Producers: Karl Faust and Thomas Mowrey.

Rachmaninoff: *Symphonic Dances*, Op. 45. Dallas Symphony Orchestra conducted by Donald Johanos. Turnabout TV 34145S. Recorded in 1967, McFarlin Auditorium, Dallas, TX. Recording Engineer: David Hancock, Producer: Thomas Mowrey. 45 rpm reissue by Analogue Productions (two 12" records).

Rimsky-Korsakoff: *Scheherazade*, Op. 35. Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo LSC-2446. Recorded in 1960, Orchestra Hall, Chicago, IL. Recording Engineer: Lewis Layton, Producer: Richard Mohr. Reissue by Analogue Productions.

Wagner: *Die Walküre*. Jon Vickers, David Ward, Gré Brouwenstein, George London, Birgit Nilsson, Rita Gorr; London Symphony Orchestra conducted by Erich Leinsdorf. Decca (British) 7BB 125-9 (5 records). Recorded in 1961, Walthamstow Town Hall, London, UK. Recording Engineer: Kenneth Wilkinson, Producer: Eric Smith.

groove once every revolution. As an experiment, I temporarily increased the tracking force to 7 grams. This time, the record played without problems. I don't recommend exceeding the recommended tracking force, but I wanted to see if the stylus cantilever was long enough to prevent the cartridge from bottoming out. The RS78 passed this test without problems. After this test, I promptly lowered the tracking force to the nominal 5.0 grams. Readers interested purchasing a turntable for 78 rpm playback should visit the web sites of Esoteric Sound, KAB Electro-Acoustics and Nauck's Vintage Records. A fine entry-level 78 turntable table is the Audio Technica AT-LP140XP. More expensive models are also carried by these dealers.

LP Performance

The editor of *audioXpress* asked me to tackle this review because of my long-time interest in the playback of vintage records. But, the capabilities of the Olympia and Wellfleet cartridges playing modern stereophonic recordings will be the main reason most music lovers purchase these cartridges. I've included a short list of some of my favorite reference LPs. These records reflect a variety of stereophonic recording techniques, from fairly simple to heavily multi-microphone techniques.

David Hancock used four spaced Figure-8 ribbon microphones—a simple but unorthodox approach—on the Dallas recording of the Sergei Rachmaninoff *Symphonic Dances*, whereas Deutsche Grammophon used a forest of microphones on the Boston recording of Gustav Holst's *The Planets* conducted by William Steinberg. Engineer Lewis Layton used three space omni-directional microphones, with a small number of judiciously placed accent mikes, on the RCA Victor Living Stereo recordings with Fritz Reiner and Charles Munch. Decca's legendary engineer Kenneth Wilkinson used the three-microphone Decca Tree as his main pickup with spot miking that was tastefully accomplished. Articles I wrote some time ago on the Rachmaninoff recording, and stereo microphone techniques in general, provide additional information [4, 5].

A good vinyl playback system should reveal the differences in engineering on these recordings, and the Olympia did an admirable job. Decca's secret was its ability to use accent microphones while still preserving the perspective of the main "tree" pickup, and maintaining a sense of depth on the accented sections of the orchestra. Deutsche Grammophon's approach sound much more like the entire soundstage is manufactured rather than natural. RCA Victor Living Stereo recordings reflect Layton's own skill at maintain a realistic

soundstage, albeit with a very different main pickup. David Hancock's Rachmaninoff LP was made in a very dry venue, but is notable for dynamic, realism of individual instrumental timbres, and a natural soundstage. The Olympia had no problem revealing the essential qualities of each of these recordings. It gives a realistic hall ambience, and reveals the differences between ambience captured naturally by the main pickup, and ambience created by placing extra microphones out in the hall (Decca and RCA Victor vs. Deutsche Grammophon). The dryness of the Rachmaninoff is a striking contrast to the other venues in which these recordings were made, but the ribbon microphones favored by Hancock have a uniquely rich, full-bodied sonic character that is unlike any condenser microphone used in the other recordings. The differences were ably revealed by the Olympia. The Olympia is surprisingly detailed and articulate for a cartridge in this price range, with a smooth treble region that should provide many hours of musically-satisfying listening.

The Wellfleet takes its virtues of the Olympia several steps further. High-frequency extension and detail are greatly improved, sound staging is more three-dimensional, with the most noticeable improvement being in depth. Bass is better defined and extended. The Wellfleet is a very refined cartridge that reveals subtleties in a recording – and subtle differences between recordings—that less expensive cartridges can't be expected to match.

There's a credible sense of high-frequency air that, while not the equal of my Denon DL-103R moving-coil cartridge, gives you a taste without the expense of an outboard step-up device, or the noise that can be introduced by the extra gain required by moving-coil cartridges. Moving from the Olympia to the Wellfleet takes you from affordable high-end vinyl playback to performance that approaches the best that can be achieved with moving-magnet design. Like the Olympia, the Wellfleet provides natural, non-fatiguing sound that should be good for the long haul.

Since my retirement from the Crane School of Music at SUNY Potsdam I've been doing volunteer work in its recording archive, transferring vintage Crane performances to digital. Next fall I will begin a series of transfers from some very well-recorded LPs made in the mid-to-late 1960s. I've decided to use the Wellfleet for this important project.

Conclusion

Although any of the four cartridges in this series can be converted to 78 rpm playback by removal of the LP stylus and insertion of the RS78, the swap can be a bit tricky. The plastic stylus mount has

About the Author

Gary Galo retired in 2014 after 38 years as Audio Engineer at The Crane School of Music, SUNY at Potsdam, NY. Since then he has worked as a volunteer in the Crane Recording Archive, transferring vintage Crane recordings to digital. He is the author of over 300 articles and reviews in over a dozen publications, and has been writing for audioXpress and its predecessors since the early 1980s. Gary is a long-time active member of the Association for Recorded Sound Collections, a frequent presenter at ARSC conferences and the author of numerous articles and reviews in the ARSC Journal. He is a member of the Boston Audio Society and a Life Member of the Audio Engineering Society.



smooth and rather narrow sides, making a solid grip difficult. I found that the safest way to swap styli is to remove the headshell and place it upside down on a table. By grasping the back edges of the stylus mount, I was able to safely pull the stylus out of the cartridge body. I wish Sumiko would consider selling the cartridge housing bundled with only the RS78 stylus, for those who don't need an LP stylus.

If you plan to use one of these cartridges for both 78 and LP playback, you might consider purchasing the low-priced Rainier, mounting it in a dedicated headshell for 78s only. The Rainier sells for only \$79, making it a very affordable solution for a dedicated 78 cartridge, free of the risks and hassles of stylus swaps.

The Sumiko Olympia and Wellfleet cartridges are excellent cartridges in their respective price ranges. The Olympia makes is a cost-effective way to enter the world of high-end vinyl playback, while the Wellfleet offers performance that puts in near the upper end of moving-magnet designs. Fitted with an RS78 stylus, any cartridge in this series will provide fine performance on 78 rpm discs. 

References

- [1] Galo, Gary. "An Archival Phono Preamp," *Linear Audio*, Volume 5, 2012, pp. 77-104.
- [2] J. P. Maxfield and H. C. Harrison. "Methods of High Quality Recording and Reproducing of Music and Speech based on Telephone Research," *Bell System Technical Journal*, Volume 5, No. 3, July 1926, pp 493-523. Available from the Internet Archive, <https://archive.org/details/bstj5-3-493>.
- [3] W. R. Isom, "Record Materials, Part II: The Evolution of the Disc Talking Machine," *Journal of the Audio Engineering Society*, Special Centennial Issue, Volume 25, No. 11, October/November 1977, p. 719.
- [4] G. Galo, "The Legendary Rachmaninoff Symphonic Dances," *audioXpress*, December 2012, pp. 24-28.
- [5] G. Galo, "Stereophonic Recording: What do Listeners Prefer?" *audioXpress*, April 2014, pp. 22-28.

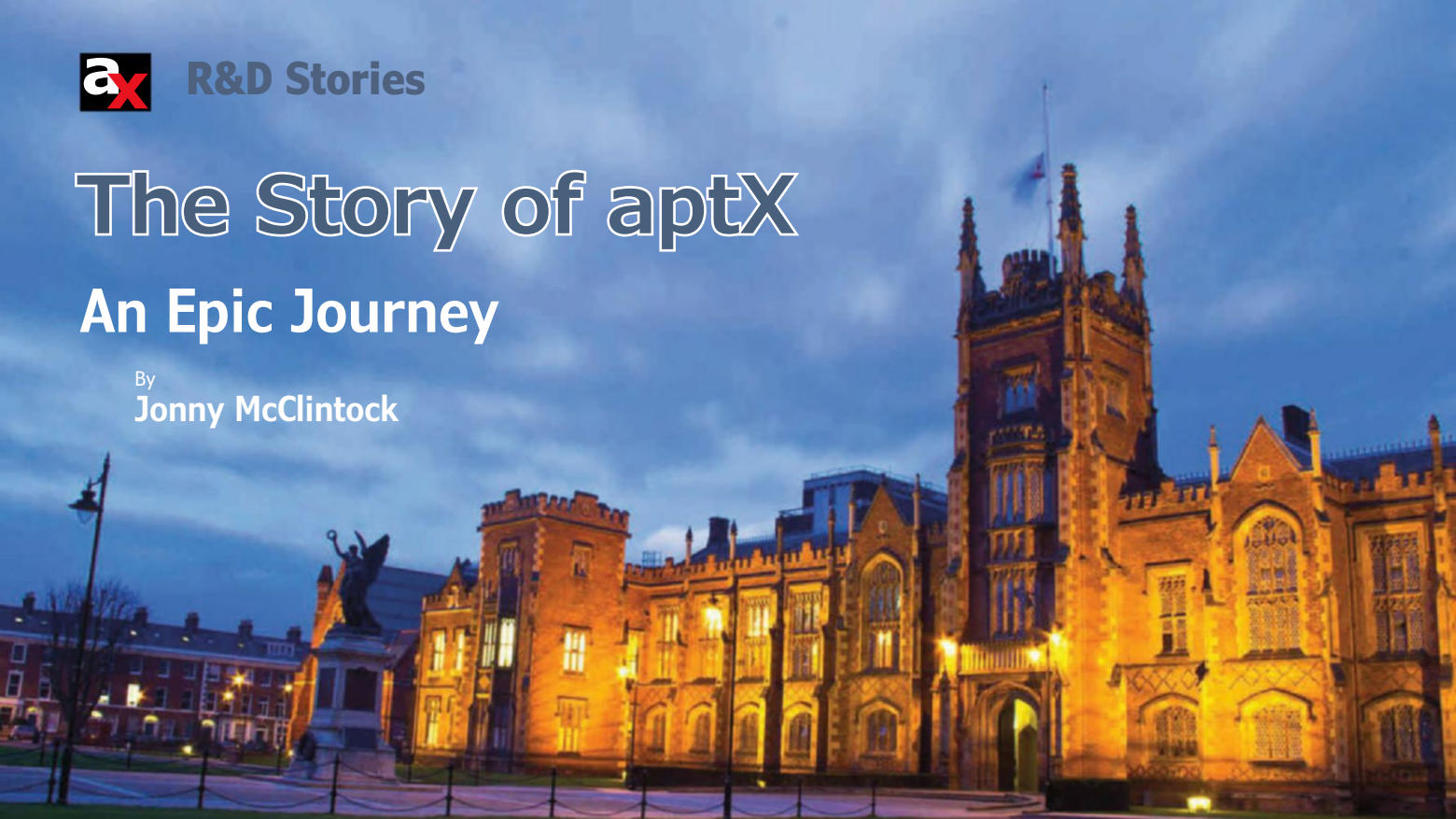
Resources

Audio Technica,
www.audio-technica.com/en-us/turntables/type/direct-drive/at-lp140xp
Esoteric Sound, www.esotericsound.com
KAB Electro Acoustics, www.kabusa.com
Nauck's Vintage Records, <https://78rpm.com/collections/audio-gear>
Sumiko Audio | www.sumikophonocartridges.com

The Story of aptX

An Epic Journey

By
Jonny McClintock



From film sound to radio broadcasts, studios, and finally to Bluetooth and more than 12 billion devices, the successful story of aptX technology is inextricably linked to the history of digital audio. A remarkable corporate journey.

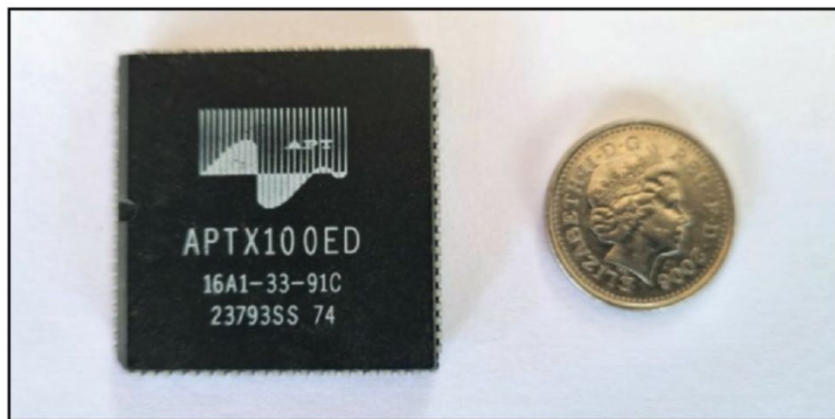
Queen's University Belfast

The following is an overview of the aptX audio codec technology, and my personal journey, now that after 28 years I have decided it was time to move on. However, before diving into the details, I must explain how aptX became interlinked with my professional career.

Having graduated from the University of Ulster at Jordanstown, Belfast, in 1986, I joined Dolby Laboratories in London and worked on the Dolby SR (Spectral Recording) noise reduction format

and cinema products. I did two stints with Dolby between 1987 and 1992, which were interspersed with time in Australia and working with Ericsson on analog Cellular Base Stations.

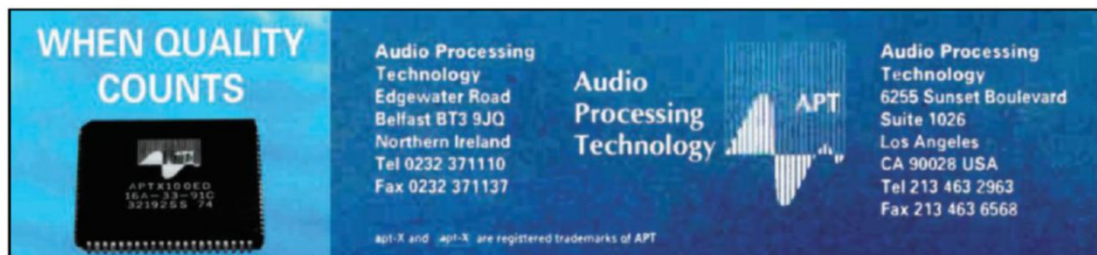
In 1993, before the Peace Agreement in Northern Ireland, I wanted to move back to Ireland. It is a superb country and the outbreak of peace offered hope to the diaspora to return home. Audio Processing Technology (APT), Ltd., headquartered in Belfast, offered me a job in customer support and that started the aptX journey. I moved into sales in 1994 and relocated to the Los Angeles, CA, office, then became the commercial director in 2000, and later in 2005, with Noel McKenna, became a co-owner via a management buyout (MBO).



aptX technology was presented to the market in 1990 as the APTX100ED, a mask programmed DSP chip, and was adopted by radio automation companies and then DTS for 5.1 surround-sound playout in widescreen cinemas.

Data Compression

The history of digital audio data compression goes back to the mid-1980s. At that time, digital was taking over from more traditional analog storage, playout, and transport mechanisms and once digital music became available, there was an obvious need to reduce storage space and data rates due to cost and space implications. Back then, only a handful of technical pioneers were capable of developing audio compression algorithms, namely James D. Johnston (Bell Labs, AT&T, and Microsoft) and Stephen Smyth



APT grew quickly throughout the 1990s and its products sold all over the world to post-production studios and broadcasters for satellite and ISDN distribution of audio.

along with places, such as the Fraunhofer Institute in Germany and Sony Corp. in Japan. Also making a significant impact were Peter Craven and Malcolm Law of Algol Applications, Ltd. Essentially, all original and innovative thinking on audio compression/decompression (codec) techniques date back to these names.

Codec architecture had three schools of thought: Adaptive Differential Pulse-Code Modulation (ADPCM), perceptual, and fractal/wavelet-based. There are five key metrics for a codec, namely compression ratio equating to bit rate or storage space, audio quality, latency, power (MIPs/Memory) and robustness (although this can be addressed via additional forward error correction—FEC—techniques). Most commercially successful codecs will generally have addressed at least three of these demands. The original aptX Codec, which used ADPCM, addressed latency, audio quality, and robustness. Although a little heavy on MIPs, it was light on memory (RAM) requirements.

There was also an over-dependency on data rates, which can be seen as a tradeoff for latency and real-time applications. Conversely, MP2 (MPEG-1 Audio Layer II using Perceptual coding techniques), which was the emerging codec from Fraunhofer at the time (before MP3 or MPEG-1 Audio Layer III), offered exceptional bit rate and MIP efficiencies but fell down on latency and audio quality. However, the value of being able to cram more music into a smaller space had its own attractions, regardless of the consequences. Elsewhere, Sony was forging ahead with ATRAC and Algol with Meridian Lossless, which ultimately became Dolby Lossless/TrueHD.

As stated, my own journey started with APT in Belfast in May 1993, almost a full week after Stephen Smyth left to join DTS, where he later developed the Coherent Acoustics algorithm. Stephen, along with his brother Mike and colleague Paul Smith, had already set up APT, Ltd. in 1988 using seed funding from QUBIS (the commercialization arm of Queen's University, Belfast) and Solid State Logic (SSL). Stephen received his PhD in digital audio data compression, which partially looked at G.722 as a two sub-band, 14-bit voice algorithm and moved that to a 16-bit, four sub-band music codec (aptX100). This investment in APT allowed the founding team to tape out a chip known as the APTX100ED using a masked ROM from AT&T.

Stephen took his team over to California and was at the epicenter of DTS' march into consumer electronics. The legacy was DTS' continued use of the APTX100ED chip for its cinema products, due primarily to the low latency of accessing stored content, which ensured synchronization between audio and film. DTS and aptX100ED were given a



In 2005, with help from Crescent Capital in Belfast, Trinity Venture Capital in Dublin and Queens University, APT went through an MBO and became management-owned by directors Noel McKenna (managing director) and Jon McClintock (commercial director).

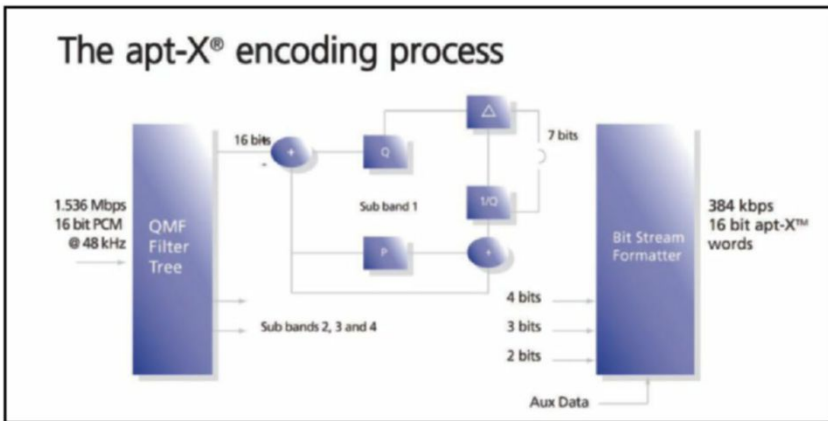


The first aptX enabled Bluetooth Stereo headphones from Sennheiser were launched at the 2009 IFA International Consumer Electronics Show in Berlin, Germany.

About the Author

Jonny McClintock has worked in the audio industry for more than 30 years, the majority of that time he has been involved in the marketing and selling of aptX. He has recently left Qualcomm and is currently the Commercial Director for AntennaWare | Wearable Antenna Technology | Made To Wear and he has a consulting role with Sonical.ai. His passion is driving start-ups.

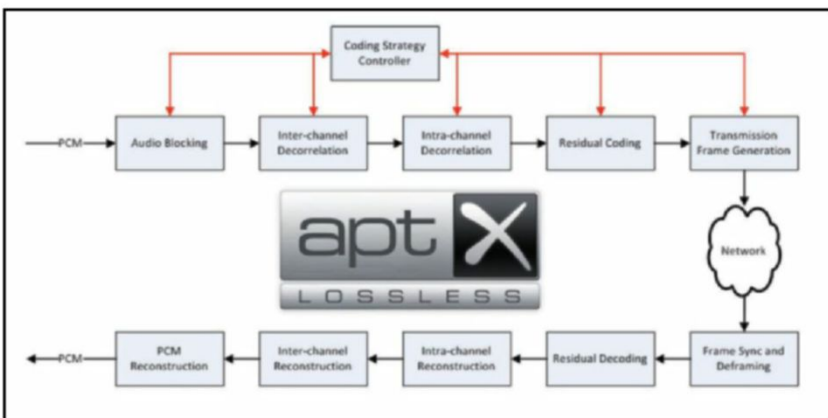




The original or Standard aptX, 16-bit, ADPCM algorithm, has proven extremely popular since its launch. In 1999, APT developed an updated new version of the algorithm to meet the requirements of new high-performance professional audio applications such as DAB, DVB and 5.1 channel surround sound. Branded "Enhanced aptX," this algorithm operated with 16-, 20-, or 24-bit word resolution and had a host of other performance benefits. Standard aptX and Enhanced aptX remain in daily use by radio stations, broadcasters, cinema surround sound, and professional music studios around the world, and is relied upon by manufacturers.



This image shows one of the original Worldnet hardware codec solutions for broadcast applications with Enhanced aptX.



In 2009, complementing the existing Standard aptX, Enhanced aptX, and aptX Live (low latency) codecs, APT announced at the NAB Show 2009 its first iteration of the aptX Lossless audio codec, supporting audio up to 24-bit/96kHz to serve broadcast applications, digital wireless microphones, and music storage and archiving applications. It was "a big deal" at the time because it promised a "hybrid" coding scheme for applications where average and/or peak compressed data rates must be capped at a constrained level.

huge boost when Steven Spielberg decided to use this format for his movies, most notably the *Jurassic Park* franchise and *Saving Private Ryan*.

Meanwhile back at APT, in Belfast, a range of products servicing the needs of the radio and voice over/ADR/stereo mix approval industries were being developed. In addition, selling the APTX100ED chips to a range of OEMs in the radio broadcast industry was progressing very well. The chip was also incorporated onto a PC sound card, which was sold to radio automation playout system providers. At the time, the company had around 30 employees, was turning over approximately £3 million per annum, and had just won the Queen's Award for Export. The legacy of Stephen Smyth and his cohort guided APT for another five years before a lack of innovation started to cause some pressure.

Toward the end of the 1990s, the pressures built further when AT&T/Lucent issued an "end of life" for the APTX100ED chip along with the diminishing cost of storage space as Moore's Law became apparent. The lack of a viable alternative to a chip combined with a saturated VO/ADR market and the radio automation system moving to Linear PCM, meant that the future was not very bright. Corporate activity with SSL and a consistent lack of ability to produce stable, cost-effective products for the radio industry were also factors in the scaling back of the company. It all came to a head in 2004 when SSL went into receivership. APT was bought out from the receivers in an MBO using VC's funds from Crescent Capital in Belfast and Trinity Venture Capital in Dublin. We began management buyout negotiations with SSL in January 2005. APT Licensing, Ltd. was incorporated in March 2005.

From Broadcast to Bluetooth

APT was always profitable and, under new management, the three key objectives were to build hardware devices that addressed the needs of the broadcast market, develop licensing, and freshen the roadmap for new audio codecs. Given that the primary usage in the radio industry was for mission critical applications (e.g., studio to transmitter links, station networking and remotes as well as outside broadcasts), a bulletproof, highly redundant range of products that ensured "dead air" never occurred was of the utmost importance.

Getting this correct while at the same time embracing the migration from synchronous to IP networks, allowed APT to get back on the road to recovery. The winning of a tender by the European Broadcasting Union (EBU) for the networking of all EU Public Service Broadcasters for stereo and 5.1 content opened the flood gates and the company

was back in the game and making a competitive comeback.

At the same time, the licensing business was being developed but wholly focused on professional applications. Sennheiser (which was familiar with aptX from digital wireless microphone applications) tabled an interesting opportunity from its Consumer Electronic Business Unit. Sennheiser asked APT to look at the emerging Bluetooth market as they believed the incumbent low-complexity subband codec (i.e., SBC) was not fit for the purpose. In 2008, Sennheiser launched its first Bluetooth device at CES and APT won the “Best of Bluetooth” award from the CES Committee.

With approximately 60 employees in 2009, a turnover of £6 million and numerous awards for innovation and growth, the expansion/development of the company was clear to see. APT had been split into two companies—APT Hardware Limited and APT Licensing, Ltd. In 2009, APT Hardware was sold to the French company Audemat (today part of WorldCast Group) and the proceeds were returned back to the initial investors.

APT Licensing, Ltd. went on to win deals with Apple, Motorola, Nokia, and Microsoft and in 2010



After selling its broadcast hardware business in 2009 to Audemat, in 2010, APT Licensing and its aptX suite of audio compression algorithms was acquired by CSR, a company with a strong portfolio of connectivity and location platforms that included Bluetooth, GPS, FM, Wi-Fi, UWB, NFC, and other technologies used in mobile consumer devices.



Having launched the www.aptX.com website with background and product information, and appointed a PR company to assist with driving the aptX brand, the message remained consistent during the CSR years.

Cambridge Silicon Radio (CSR PLC) swooped in to acquire the company. CSR was a pioneer of the integrated Bluetooth and processing platforms. Their approach boosted the roll out of Bluetooth initially for voice and later for music.

For the next few years, the CSR aptX team enjoyed success in the Bluetooth world with wins in the Android handset market and headset OEMs readily embracing the codec primarily due to its

HOLOPHONIX
NATIVE

HOLOPHONIX Spatial Sound Solution Available Now
as a macOS App

A 3D rendering of a box for the Holographix Native 128 IN macOS app. The box is dark blue with a white geometric pattern resembling a crystal or a molecular structure. The text 'HOLOPHONIX NATIVE' is printed vertically on the side, and '128 IN' is printed in large white numbers on the bottom front. The background of the box features a dark, starry space scene with a silhouette of a mountain range at the bottom.

www.holophonix.xyz

*Spiral Air*tm
Interconnects

*Coax
Is Cool
Again!*

A detailed cross-section diagram of a Cardas Golden Strand cable. It shows a central core, surrounded by an air-filled cotton dielectric, a unique spiral Litz shield, and ETI Tellurium copper connectors.

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

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

KAB
Preserving The Sounds
Of A Lifetime
www.kabusa.com



Following the acquisition from CSR, in 2018 Qualcomm launched Qualcomm aptX Adaptive, a next-generation audio codec designed to be dynamically adjustable and provide a single flexible solution for virtually all key wireless audio applications. Effectively, it combines the features of low-latency aptX, aptX, or aptX HD, depending on the content and the connectivity.



With streaming music services starting to offer lossless options, Qualcomm introduced a CD-quality lossless option over Bluetooth. aptX Lossless over Bluetooth was announced in September 2021, extending the aptX Adaptive set of technologies—supported by its latest Snapdragon Sound processors and combining codec, wireless link and RF optimizations.



Following the original Snapdragon Sound announcement, in 2022 Qualcomm announced new Snapdragon Sound optimized platforms that offer dual-mode, combining traditional Bluetooth wireless audio and the latest LE Audio technology standard, including support for audio sharing and broadcasting. For the smartphone manufacturers, these technologies are fully integrated with Qualcomm's mobile processors.

faithful reproduction of audio quality and low latency credentials. The original codec—the aptX Classic—was augmented by aptX HD, moving the audio quality from 16-bit/44.1kHz to 24-bit/48kHz. In addition, an interoperability program was introduced to ensure that each OEM followed guidance to guarantee a consistent experience for the consumer.

In 2015, CSR was then acquired by Qualcomm, which wanted to pursue a strategy of joining up handsets to earbuds to ensure an optimum experience for music, video, and gaming through an integrated approach known as Q2Q (Qualcomm to Qualcomm). This furthered the roadmap of aptX with the launch in 2018 of aptX Adaptive supporting 24-bit/96kHz and along with aptX Voice for super-wideband voice quality to complement VoLTE on 4G and 5G.

aptX Adaptive was designed to be dynamically adjustable, combining the features of low-latency aptX, aptX (Classic), or aptX HD, depending on the content and the connectivity.

The culmination of this technical journey was the ability to deliver Lossless CD quality via aptX Lossless, which is a function of improved data throughput that's the result of combining Qualcomm High Speed (QHS) and the scalability of aptX Adaptive. Introduced in 2021, aptX CD Lossless technology delivers bit for bit exactness when compared to the original Linear PCM, expanding the options available for Bluetooth wireless audio, combining codec, wireless link, and RF optimizations. Bluetooth is now fit for the purpose for music.

Nobody at the Table Went Hungry

The end of my professional journey culminated in aptX being subsumed into Snapdragon Sound, which addresses audio quality, latency, and robustness by marrying the aptX Codec with QHS Link modulation. Between aptX and Snapdragon Sound there have been more than 1,000 license agreements signed and aptX is used in more than 12 billion devices across handsets, PCs (both Microsoft and Mac), headsets, earbuds, speakers, soundbars, TVs, cars, and happily LP record players.

There are even metal and water leak detectors using aptX over Bluetooth. Adding in the professional applications for radio and voice over, with digital wireless microphones and a successful entry into the security sector, the story of the aptX brand is writ large.

Having said all that, and for the sake of balance, it is worth noting that there may be headwinds. The Bluetooth SIG works slowly, but there is a strong probability that the Bluetooth Low Energy (BLE) solution with LC3 will address low latency use cases



A personal journey over 28 years, from APT to CSR and Qualcomm

and it should be a much better codec than Bluetooth's classic SBC codec from an audio reproduction perspective.

Fraunhofer will augment the audio experience with LC3+ for higher quality again. If you add in LDAC and LHDC codecs, then this period could be considered the golden era of audio coding in terms of choice. What will need to be addressed, and what Qualcomm is achieving, is the Interop testing to ensure a consistency of experience across the three performance pillars of audio quality, latency, and robustness. This takes time and investment but will be well worth the effort in terms of underwriting brand value.

I hope that my own personal journey in the future will be as interesting, varied, and successful as the past 28 years. Having

worked in audio networking for radio before becoming involved in delivery via Bluetooth, the last piece of the jigsaw is content generation and acquisition. Considering the challenge around RF connectivity, simply because of the sheer volume of devices in the 2.4GHz space, there could be value in improving the antenna design, especially given the reduced form factor in earbuds, which will ultimately result in body blocking and range issues. Merging knowledge around codecs and RF could help in digital wireless microphone for news reporting, music, and talent use cases. If you've made it to the end of this article, many thanks for persisting. Better still, if the last few lines resonate, please feel to get in contact via my LinkedIn profile. 

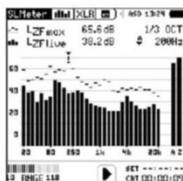
Ready for any Challenge!

Discover an Audio Analyzer you can trust for your application:

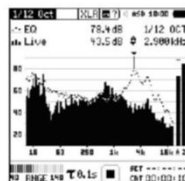
- Sound Level Meter class 1
- Level RMS / THD+N
- 1/1 or 1/3 octave RTA (up to 1/12 optional)



Distortion Analyzer



RealTime Analyzer



1/12 octave Analyzer

INTi
AUDIO | NtI Audio Inc.

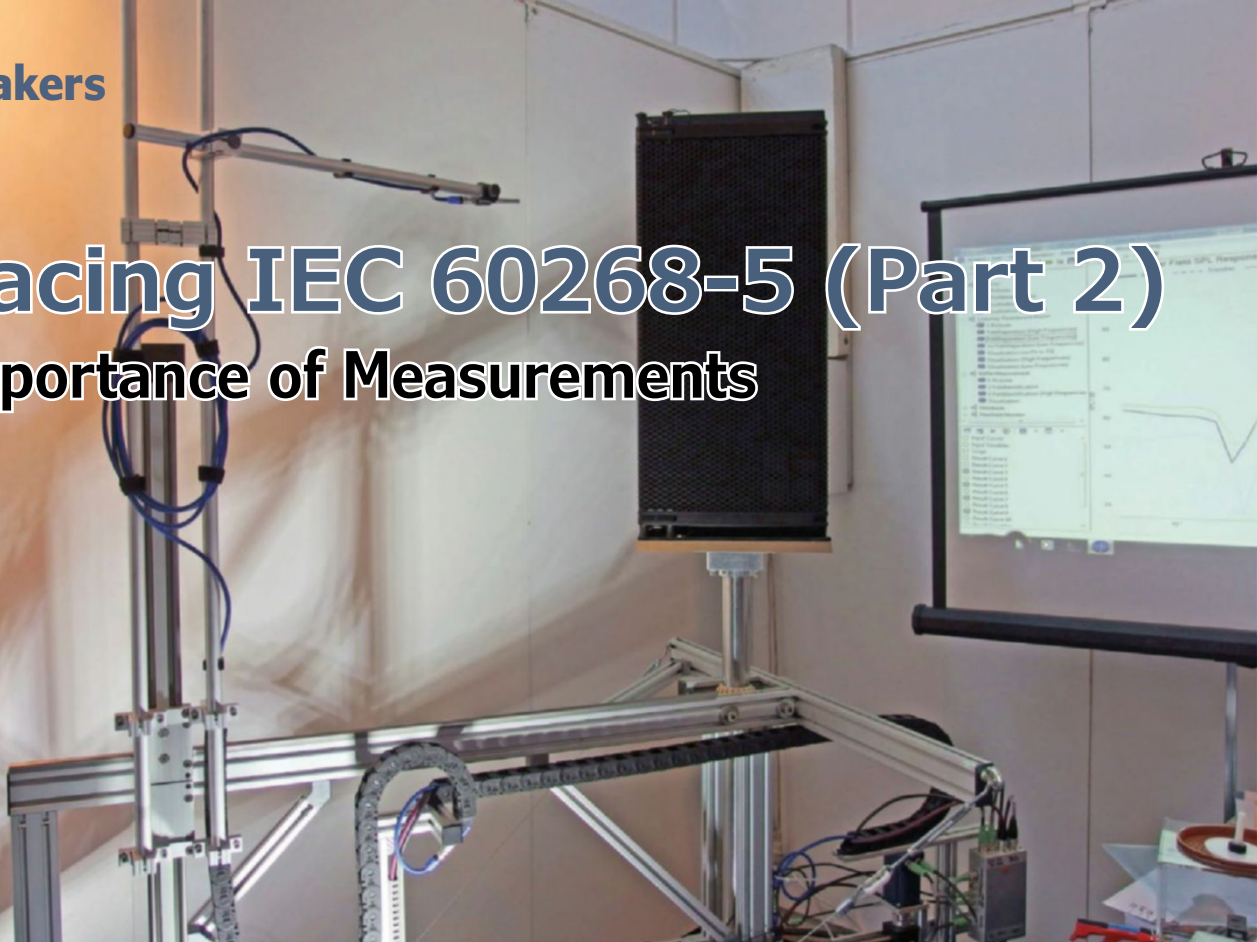
americas@nti-audio.com | (503) 684 7050 | www.nti-audio.com/XL2



Replacing IEC 60268-5 (Part 2)

The Importance of Measurements

By
Geoff Hill



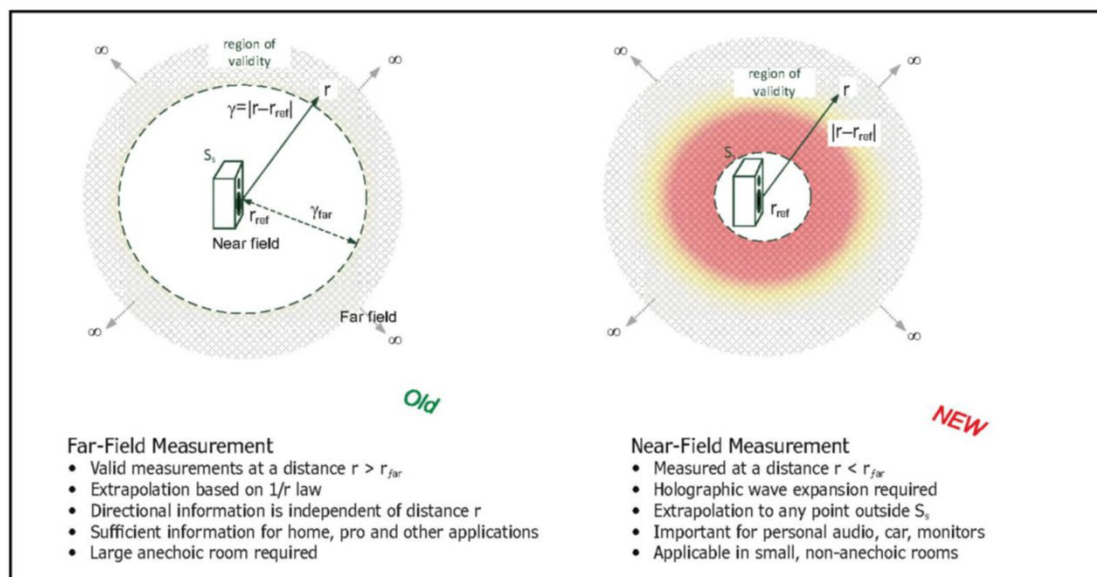
Geoff Hill concludes his discussion about the work taking place on loudspeaker measurement standards, and the progress the International Electrotechnical Commission (IEC) has made with its standards efforts.

In Part 2 of this article series focusing on loudspeaker measurement standards, I discuss how these new standards apply to measurements of sound in 3D space and in production environments, measurement of SPL using voltage sensitivity, and the measurements of microspeakers.

Measurement of Sound in 3D Space

In IEC 60268-21:2018 there is a totally new section that addresses, in my view, one of the biggest limitations to practical loudspeaker measurements—namely the anechoic chamber. These new standards address the measurement

Figure 6: Measurement of the direct sound field in 3D space enables near-field measurements, which is important for personal audio, cars and monitors and is applicable in small, non-anechoic rooms.



environments and probably the most significant change is the move away from the anechoic chamber and toward defined acoustical environment and the target environment—as in a car’s cabin. And importantly, for the first time the standards specify the positioning of both the loudspeaker and microphone, in the X, Y, and Z planes—using spherical coordinates so at last we know what and where we are measuring.

Measurements can now be made in a 3D space in both conventional anechoic and non-anechoic rooms (**Figure 6**). Using holographic wave expansion we can predict the far-field performance from the near-field response(s), allowing measurement of the radiated power over a specified angle or coverage. This subject is covered in more detail by AES56 (the AES standard on acoustics—Sound source modeling—Loudspeaker polar radiation measurements).

An example of this technique is the Klippel Near Field Scanner (**Photo 2**), which is a mechanical scanner that moves a single microphone around a loudspeaker system, or an individual transducer mounted in a baffle to effectively generate a “balloon measurement.” The main advantage of this technique is that this can be done in a normal room, while the main disadvantage there is no reduction of sound from the loudspeaker so it can be a noisy process.

Measurement of Sound in Production Environments

In IEC 60268-21:2018 and IEC 60268-22:2020, an interesting option is allowed beyond the ubiquitous IEC baffle and Type A or B test cabinets, which require the use of an anechoic chamber. A chamber such as the Tetrahedral Test Chamber as shown in **Photo 3**, may be used for end-of-line testing and relative measurements by generating a defined air load and radiation condition at the front of the transducer.

An additional chamber at the rear side of the transducer can be used to provide additional noise isolation or to consider the influence of the air volume of a sealed box in the target application.

The Audio Engineering Society (AES) issued AES73id-2019 information document on the subject of loudspeaker comparison chambers includes an appendix on measurement uncertainty, again part of these standards.

Using the concept of a measurement system’s total uncertainty budget, it is possible to accurately determine loudspeaker measurement system tolerance. This is based upon the usual 95% confidence level of the individual linear contributions of the separate deviations (e.g., a microphone,



Photo 2: Measurements can now be made in a 3D space in both conventional anechoic and non-anechoic rooms. An example of this is the Klippel Near Field Scanner (NFS), which is a mechanical scanner that moves a single microphone around a loudspeaker system, or an individual transducer mounted in a baffle to effectively generate a “balloon measurement.”



Photo 3: This is an outline of a Tetrahedral Test Chamber, with an acoustic isolator shown open (a) and closed (b).

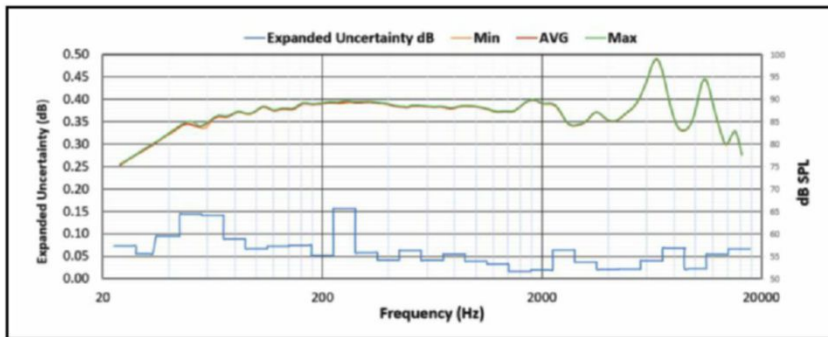


Figure 7: This chart of 100 measurements of the same driver was measured over the course of a couple of hours, also shows the measurement uncertainty to 95% as well as the results.

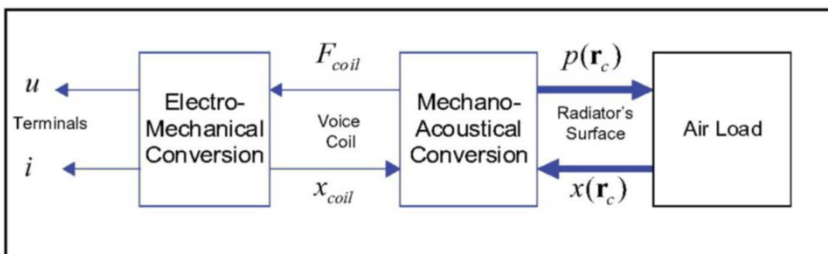


Figure 8: This graph shows the scope of the IEC 60268-22 electrical and mechanical measurements.

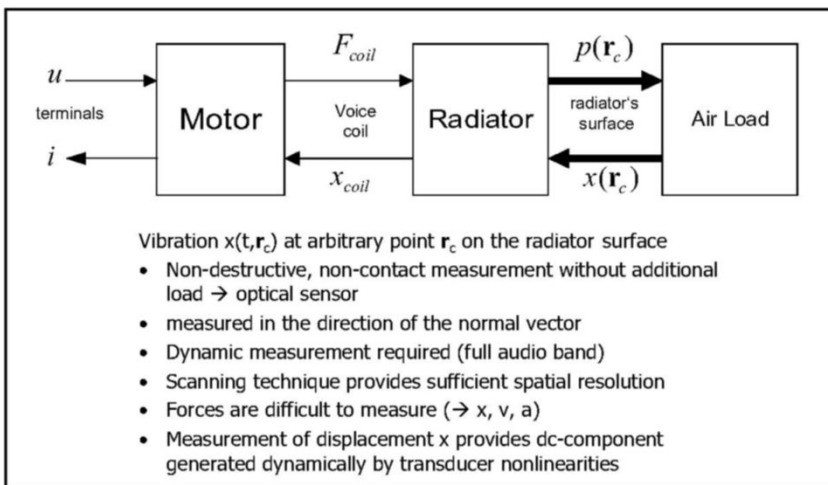


Figure 9: Mechanical measurements measure vibration $x(t, r_c)$ at an arbitrary point r_c on the radiator surface.

About the Author

Geoff Hill, 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 is Chair of the AES SC-04-03 Working Group on loudspeaker modeling and measurement and a member on the IEC Working Group TC 100/TA 20/PT 60268-22. He is also the author of *Loudspeaker Modelling: A Practical Introduction*. Geoff welcomes your comments and can be contacted via email geoff@hillacoustics.com.



analyzer, positioning, etc.). An example of such a result is shown in **Figure 7**, where the uncertainty of the measurement is plotted in addition to the usual SPL axis vs. frequency.

AES73id brings to these standards the concept of an Uncertainty budget, analogous to the gain budget in Mobile Telecoms, and develops a methodology for demonstrating this so that at last we can truly have confidence in our acoustic measurements.

Voltage Sensitivity

Another—and in my view overdue—update, is the change from Power to Voltage in the definition for sensitivity measurements. As is well known to the design community, the impedance characteristics of a loudspeaker driver unit is often far from flat—this results in a variable amount of power being absorbed or consumed by an individual transducer.

The change to voltage drive reflects the reality of modern design where power is no longer (and in truth has not for an awfully long time) been matched between components but rather, the amplifier typically has a much lower source impedance than the load. This also reflects the increasing use of DSP/amplifiers, which are effectively immune to impedance loading effects.

IEC 60268-22:2020

The IEC 60268-22:2020 standard is arguably the biggest change and focuses upon loudspeakers where access is available to the electrical and or acoustical surface. This allows us to assess the behavior of a loudspeaker in both the small signal and large signal domains, as well as consider the acoustical boundary conditions in which a loudspeaker is operating.

As IEC 60268-22 does consider the internal workings of a loudspeaker and incorporates much of the recent advances in loudspeaker theory, we have access to and knowledge of the internal functionality (if not the underlying design) of these components, we can use this “a priori” knowledge to go from causes to effects. Of course, a modern loudspeaker standard still requires the electromagnetic-mechanical parameters, indeed the large signal parameters are an extension of them (**Figure 8**).

So, it is appropriate to start with these parameters. However, the measurement methods have been expanded to include those in 60268-21 plus vacuum. Along with laser sensors and scanning of the diaphragm, together these allow the measurement of the voice coil displacement and other state variables that depend for their value upon various factors, such as displacement (x), current (i), and temperature (tv), as shown in **Figures 9–12**.

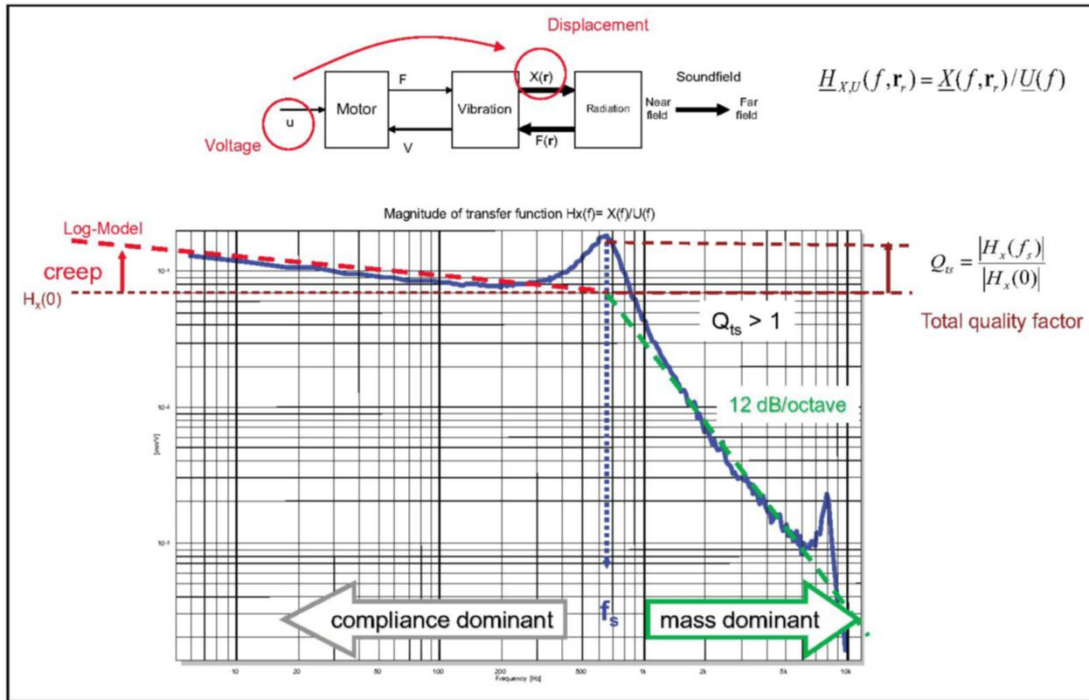


Figure 10: The measurement methods have been expanded to include the displacement transfer function.

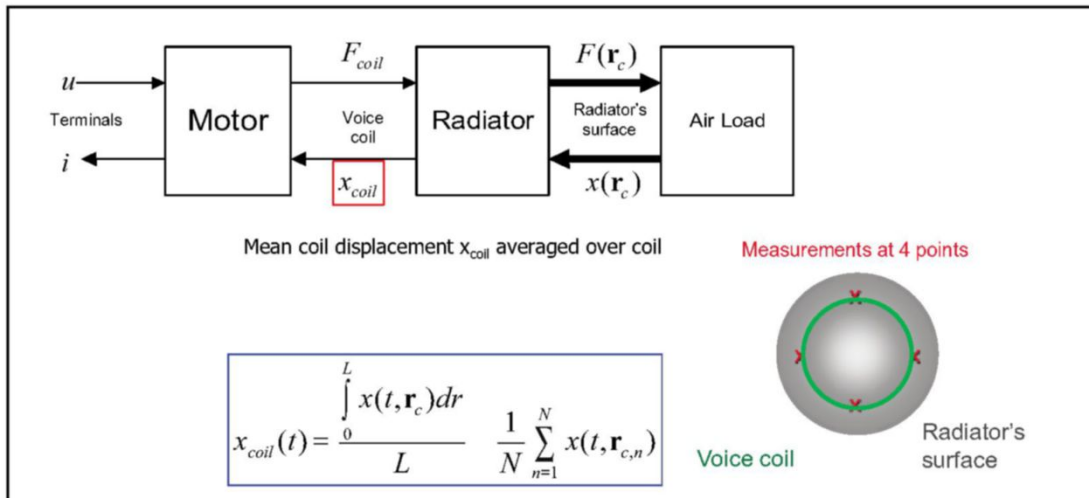


Figure 11: The measurement methods have been expanded to include the mean voice coil position.

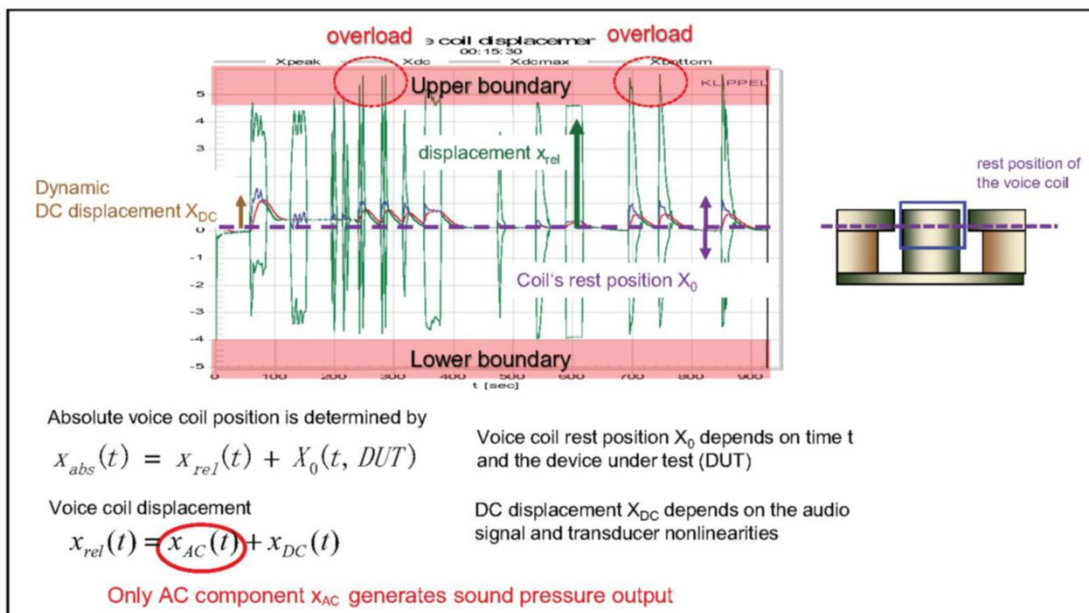


Figure 12: The measurement methods have been expanded to include the voice coil position in which only the AC component x_{AC} generates the sound pressure output.

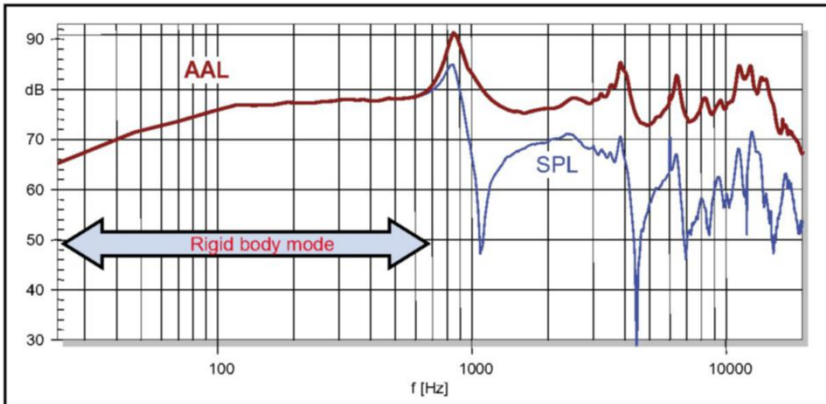


Figure 13: The calculation of the Accumulated Acceleration Level can in turn be used to understand the breakup modes in a diaphragm.

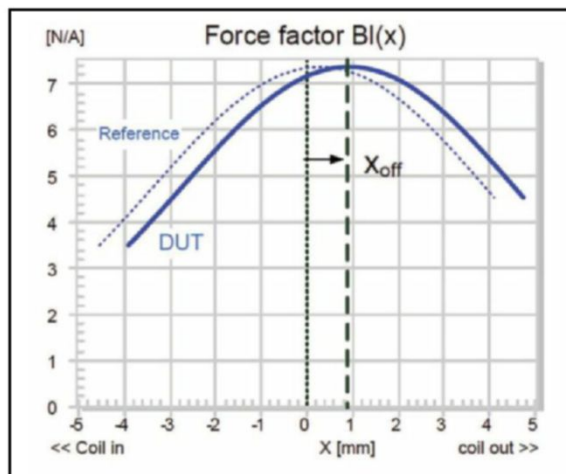


Figure 14: This graph shows a demonstration of a typical $Bl(x)$ curve versus a reference measurement.

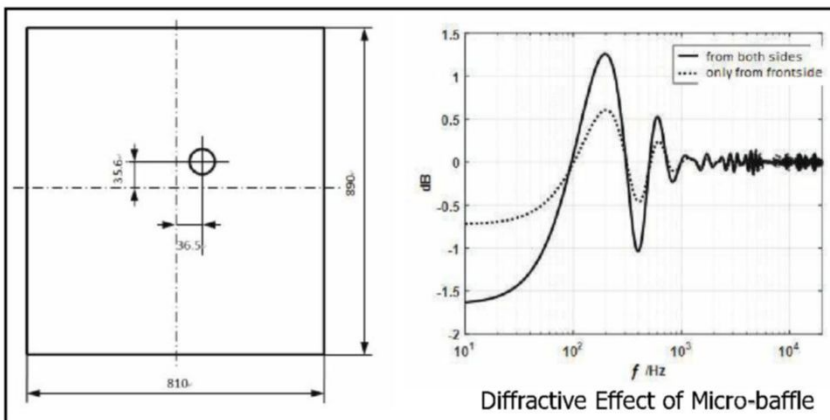


Figure 15: This graph demonstrates acoustic measurements for a micro baffle—810mm × 890mm at measuring distance=0.1m.

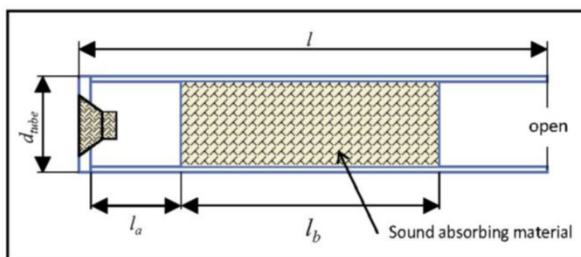


Figure 16: Measuring a plane wave tube is convenient for practical measurements, such as at a production stage.

The calculation of the Accumulated Acceleration Level (**Figure 13**) can in turn be used to understand the breakup modes in a diaphragm.

An important section is now devoted to this critical aspect starting with: the well-known parameter Bl translates to $Bl(x)$, where (x) is the displacement term is shown as **Figure 14**.

So, while in previous standards we had to use the small signal parameters from the well-known Thiele-Small set, IEC 60268-22 extends these to include state variables, such as displacement (x) , with examples of particular interest being $Bl(x)$, $Cms(x)$, $Le(x)$.

This is because Cms although mathematically useful is less useful to a transducer designer than its inverse Kms and arguably all are less useful until extended into the nonlinear or large signal domain, where compliance is inverted to become stiffness $Kms(x)$, a measurement of Cms with respect to displacement (x) , then Le , the lossy impedance rise of a loudspeaker can be predicted through the use of various models from W. M. Leach, J. R. Wright, and K. Thorburg While $Le(x)$ extends this using the displacement (x) . Modal Analysis and Rocking modes are particularly troublesome for small high throw transducers, and are added together with measurement methods. Thermal characteristics together with mechanical fatigue and methods of measurements for these are also included.

IEC 63034 Is About Microspeakers

Crazy though this may sound, until the IEC 63034 standard there was effectively nothing suited to microspeakers. These are defined as speakers where the size of the radiating element (diaphragm or another radiator) is no greater than 40mm. Of course, compression drivers and high-frequency drivers (tweeters) are specifically excluded.

We may ask why not apply IEC 60268-5, IEC 60268-21 or IEC 60268-22 to microspeakers? It is simply because their typical conditions of use are smaller than the usual measurement range. Where typically speakers are measured at 1 meter, microspeakers are usually used at 10cm distance, voltage ratings are below 5V, but they have extreme displacement for their size.

The IEC 63034 document specifies the characteristics of microspeakers as well as the relevant test methods for these devices using steady-state sinusoidal signals, sinusoidal chirp, multitone, or noise. The main characteristics include, but are not limited to, impedance, displacement, amplitude frequency response, distortion, and power handling. Considerations are in place for its small size, light weight, low handling power, short listening

distance, and difficult working conditions. These take into account the working environment, test signals, equipment, preconditioning (acclimatization, pre-loading). Typical measurements include:

- Impedance Curve (rated impedance), Small signal parameters (F_s , R_{dc} , L_e , Q , M_{ms} , C_{ms} , R_{ms} , Bl , etc.)
- Displacement (peak/bottom, DC, max), amplitude, distortion (THD, IMD, HOHD)
- Input voltage/ electrical power, environmental testing (temperature/humidity)
- Stray magnetic fields

Although an improvement over the usual IEC Baffle of 1650mm × 1350mm, a baffle that measures 810mm × 890mm is still pretty big. Especially as this is for measuring microspeakers at 10cm (**Figure 15**).

For production, a plane wave tube is introduced (**Figure 16**). Plane wave tubes have long been used to provide standardized measurement results. Fortunately, alternatives are available and have been standardized in the IEC 60268-21 and 22 standards, and also by the AES in AES73id-2018.

In the AES73id standard the concept of a defined acoustic environment has been advanced, some of these are capable of measuring at 10cm while minimizing uncertainty to below $\pm 0.5\text{dB}$ from 20Hz upward dependent upon the transducer being measured. A suitable example being used for microspeaker work is the Hill Acoustics TTC350 shown in **Photo 4**.

Also, in AES73id the concept of defining the total measurement uncertainty is further developed from a single point or frequency at which it is measured to encompass the measurement uncertainty over a given frequency range along with a methodology for general application.

This enables us to more closely define what we are measuring in an amplitude response over a range of frequencies in terms of the variability. Of course, this is not of much use until its matched with an appropriate methodology for the measurements themselves, capable of a high degree of accuracy/consistency.

This article highlights just some of the many changes to the standards that bring them up to date. Taken together these standards in effect define a solid base level with which every engineer or specifier of loudspeakers should be familiar. 📺

Author's Note: I would like to thank Wolfgang Klippel and Prof. Shen of Nanjing University for permission to include many of the figures in this article.

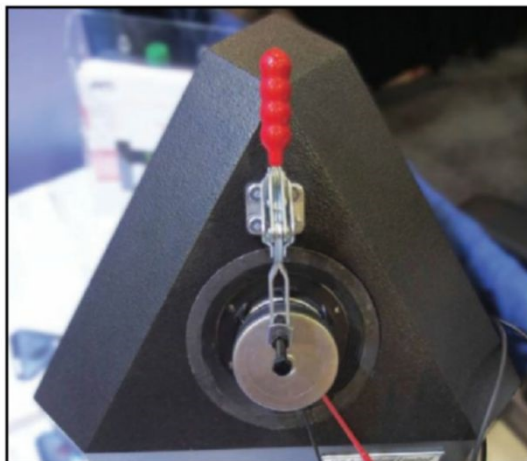


Photo 4: A suitable example being used for microspeaker work is the Hill Acoustics TTC350.

Resources

W. Klippel, "Electrical and Mechanical Measurements of Loudspeakers and Sound System Equipment: Tutorial to a New IEC Standard," 2016, www.klippel.de.

"Meter," National Institute of Standards and Technology, updated June 2021, www.nist.gov/si-redefinition/meter

"What is Metrology," NCSL International, <https://ncsli.org/page/WIM>

We are a world leading manufacturer of high performance transformers for audio applications.
Let us help you improve your sound.

"IF YOU CAN HEAR IT,
IT'S NOT OURS"

LUNDAHL
— TRANSFORMERS —

LUNDAHL
— TRANSFORMERS —

www.lundahltransformers.com
office@lundahltransformers.com

A colorful, abstract background with musical notes and the company logo. The logo is "LUNDAHL — TRANSFORMERS —". The background features a stylized face with a wide smile, musical notes, and various colorful shapes.

Considerations Concerning RT Measurement

By
Richard Honeycutt

In this article, our acoustics expert explains the history behind the reverberation time (RT) measurement and its importance. He then shares his preferred procedure for measuring RT.

When an acoustician—or a musician—visits an unfamiliar room, (s)he often claps once or a few times to get a sense of the acoustics of the room. An experienced listener can tell several things about the room from the sound of the clap including:

- The approximate volume of the room (large, medium, or small, but not actual cubic feet or cubic meters)
- The presence or absence of slap or flutter echoes caused by unwanted reflections

- The “liveliness” or “deadness” of the room, even (for a truly astute listener) the approximate reverberation time (RT)

Although we know that the effects of echoes and reverberation have been known since antiquity, we have no way of knowing at just what point in history people began using hand claps to assess their predominance in rooms.

We do know that Professor Wallace Clement Sabine developed the measurement and analysis of RT into a science during the closing years of the 19th century. His study of reverberation culminated in the “Sabine equation”:

$$RT = \frac{0.049V}{\sum S_i \alpha_i}$$

where S_i = the surface area of a particular section of the surfaces in the room, and α_i = the acoustical absorption coefficient of that particular section.

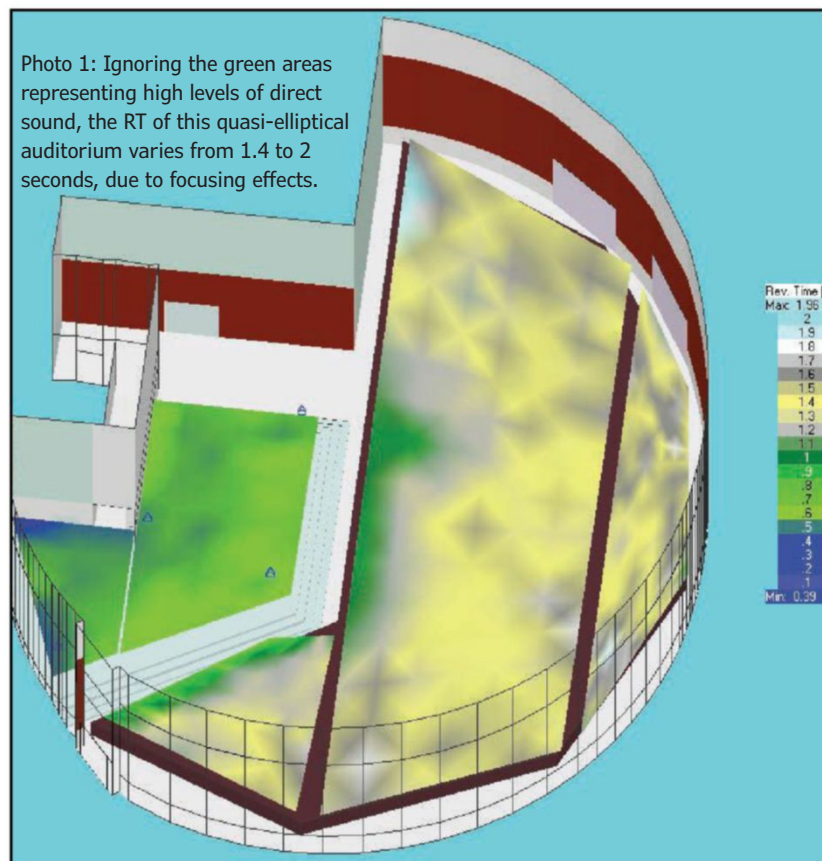
As can be seen from this equation, Sabine’s theory states that the RT of any room can be predicted from a knowledge of the room’s volume and total acoustical absorption. In order for the prediction to be accurate, two conditions must be met:

- (1) The sound in the room must be diffuse, meaning that at any point in the room there is equal probability of reverberant sound arriving from any direction.
- (2) The acoustical absorption must be evenly distributed among all surfaces: wall, floor, and ceiling.

Given these conditions, the RT is the same at any point in the room.

RT Variation within a Room

There are several possible reasons why the RT may vary from place to place within a real room. Concave wall or ceiling surfaces can focus sound creating



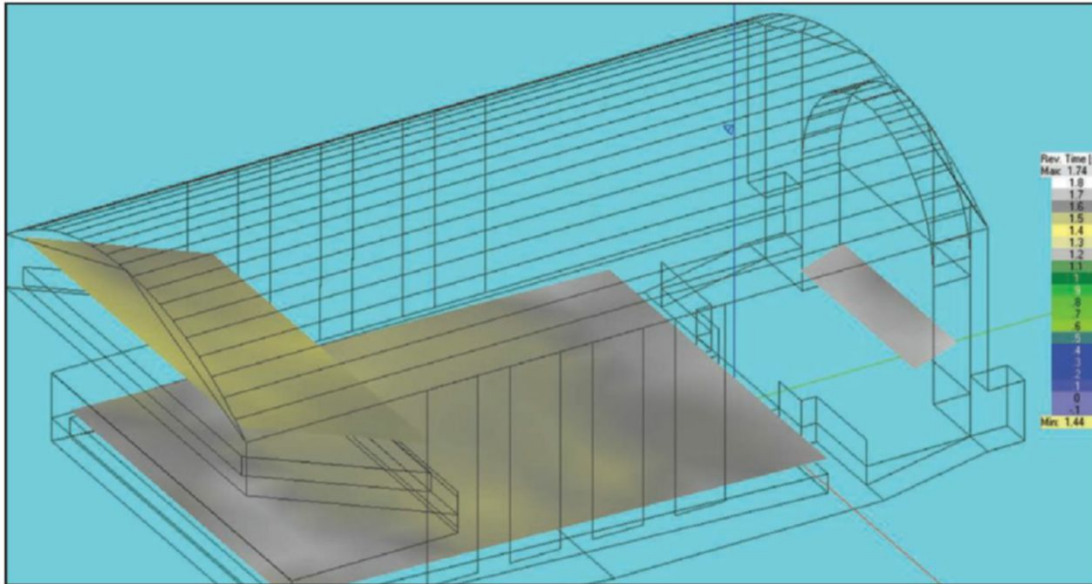


Photo 2: The RT in this auditorium varies from about 1.5 seconds in the balcony seats to 1.6 seconds in the main seats to 1.7 seconds under the balcony.

areas of higher-level direct sound. **Photo 1** shows serious variation in RT resulting from focusing effects. Architectural features, such as a deep under-balcony recess, can effectively create separate acoustical spaces within a room. As shown in **Photo 2**, the variations can be significant. The most effective sound absorber in most auditoriums is the audience, especially if the seats are not cushioned. Low occupancy in certain areas will also result in RT variations.

In general, seats located closer to the sound source—within the critical distance of the source— (talkers, singers, musical instruments, or loudspeakers) will experience a shorter RT because the initial part of the decay measurement represents direct sound rather than reverberant sound. This effect can be seen in Photo 1, with the green low-RT region on the stage where the sound source (speakers) is located.

Signal Sources

The ideal signal source for an RT measurement would have high acoustic energy (SPL > 45dB above the background noise in the room), a flat or at least known frequency spectrum, and omnidirectional radiation. The hand clap often used for informal RT assessments represents an impulse source; others include balloon pops, firecrackers, starter-pistol discharges, electric spark discharges, and electronically generated impulses.

Figure 1 shows a .wav file of a hand clap recorded in an acoustically dry space (RT $\approx$ 1.2 seconds). The multiple spikes represent different portions of the hands colliding at slightly different times (1ms to 2 ms apart), and reflections from a nearby table. The spectrum of the hand clap shown in **Figure 2** reveals that although the instantaneous sound level can be rather high, the total energy in the signal is limited due to the

shortness ($\sim$ 19ms) of the impulse. The level varies about + 12.5dB with frequency.

A clearer view of the hand clap spectrum is shown in **Figure 3** [1]. This figure also shows that at some frequencies the hand clap is reasonably omnidirectional, due to the fact that the hands are acoustically small below a few kilohertz. Another weakness of the hand clap signal is that it is essentially unrepeatable, even when results from multiple hand claps by the same person are averaged.

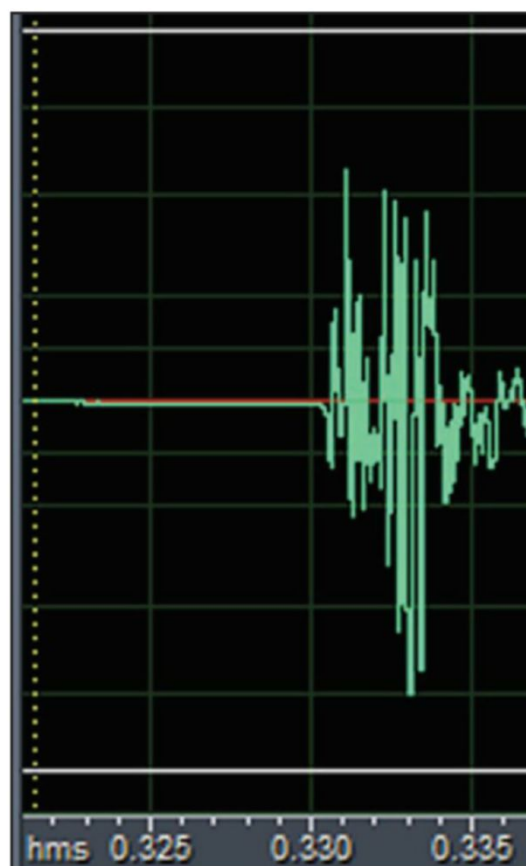


Figure 1: This hand clap recording is far from being a perfect impulse.

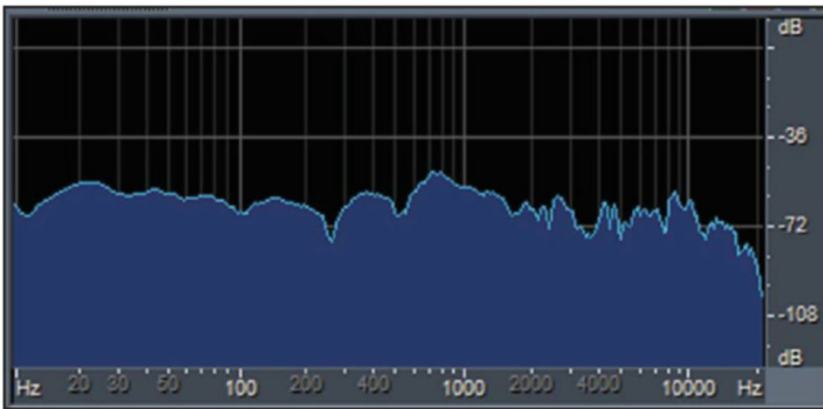


Figure 2: The spectrum of the hand clap has insufficient energy, and does not have a flat spectrum.

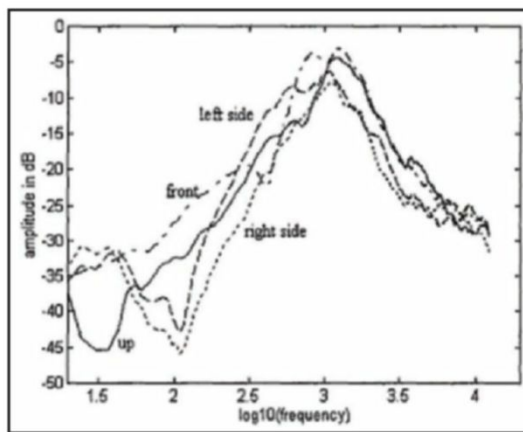


Figure 3: This detail of a hand clap spectrum does not include background noise.

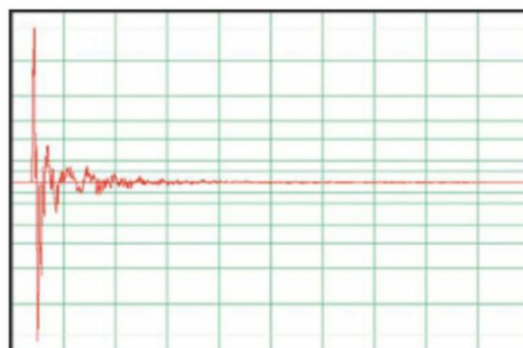


Figure 4: This trace shows the sound produced during the first 50ms of the burst of an ~8" diameter pear-shaped balloon.

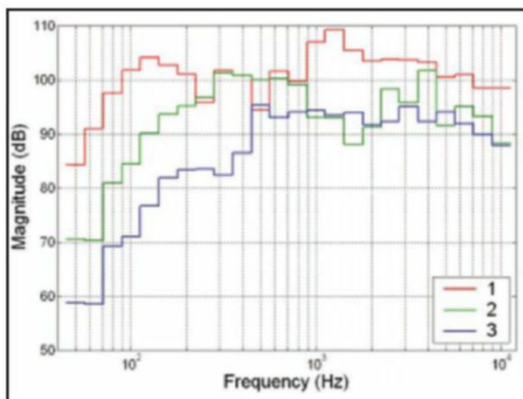


Figure 5: Balloon bursts have less variation in SPL with frequency than do hand claps. The blue trace represents the smallest balloon; the green trace, a larger one, and the red trace, the largest balloon tested.

Balloon bursts provide more controlled and repeatable signals than do hand claps. **Figure 4** [2] shows the well-defined impulse produced by the burst of a small balloon. **Figure 5** [3] shows the spectra of several balloon bursts. Large balloons provide bursts having multiple spikes, and their spectra show more low-frequency energy. Even though balloon bursts are more controlled than hand claps, the SPL and—to some degree—the spectrum are influenced by the degree of inflation prior to the burst.

Although the peak SPL produced by a balloon burst is high (~135dB), a 20dB to 25dB higher peak SPL can be produced by a firecracker. The benefits of greater acoustic energy to overcome the effects of background noise must be balanced against the danger of hearing damage. In addition, most of the acoustic energy of a firecracker explosion is concentrated between 500Hz and 2kHz, with low-frequency energy some 30dB lower in magnitude.

Starter pistols have often been used as impulse sources for RT measurements. As one would expect, the SPL-versus-time plot of a pistol shot is very similar to that of a firecracker, and the spectrum emitted from the front, rear, and top of the pistol is also similar to that of a firecracker (**Figure 6** [4]).

During the late 20th century, Engineer Bill Waslo designed and produced the IMP, a test instrument that produced a series of nicely rectangular electrical impulses which, when amplified and fed to a loudspeaker, created an excellent, repeatable impulse source that could be used for all kinds of testing, including RT measurement.

The article in *Speaker Builder*, "The IMP Analyzer," introduced the IMP and provided detailed DIY instructions and the address from which the accompanying DOS 5.5 software could be purchased. In addition to the "IMPulse" being nearly perfect, its use obviated concerns about hearing damage, since the level did not need to be very high; synchronous averaging helped to overcome the effect of the background noise.

Having oozed into the realm of electrically generated sources, we must now address the most common method of achieving omnidirectionality. A point source transducer would be omnidirectional, but any transducer sufficiently small to act even vaguely like a point source would be incapable of producing adequate low-frequency energy. The most common solution is a dodecahedron loudspeaker such as the one shown in **Photo 3**.

A dodecahedron is a 12-sided solid, and the sides of a regular dodec are identical pentagons. A loudspeaker is mounted in the center of each side, and the acoustical output, while not inherently flat

with frequency, is omnidirectional and predictable. As industry legend Pat Brown says, “The goal here is to excite the room, not to sound good.” While not perfectly omnidirectional, a good dodec is adequate for making repeatable RT measurements.

Two families of signals are useful for RT measurements. Many instruments use random noise such pink or white noise. The noise source is turned on, and the measuring instrument is set to digitally record the decay once the noise source is interrupted. Averaging can be performed by gating the noise source such that the “time on” is longer than the room requires for a constant reverberant level to build up, and the “time off” is longer than the expected RT. Firmware stores a program that calculates the RT in various octave or 1/3-octave bands.

The second family of signals is predictable. One member of this family is the Maximum Length Sequence (MLS), a series of pulses based upon a known, repeatable sequence of numbers. MLS signals provide enhanced noise immunity compared to random noise.

The other family member is a sine sweep, which provides the absolute best noise immunity. The computational load for processing MLS or sine-sweep measurements to extract the RT is greater than that needed for impulse sources, but fewer—perhaps even no—averages are needed to achieve a reliable result.

Procedure

To measure RT, you must first position the sound source. It should be located either far from reflecting surfaces, or in a room corner. The latter location is usually best for small rooms, since it evenly excites all modes. Next, the test mic should be located at the position for which you want the RT. Then the background noise level should be measured using the FAST setting and C- (or flat) weighting.

You want to note the maximum noise level, which may well vary with time. (Some RT measuring instruments automatically measure background noise level.) Then you must set the signal level at least 45dB higher than the background noise. (Some instruments also automate this process.) From here, the process depends upon your RT measuring instrument, but essentially it involves interrupting the signal either once per measurement or periodically, and waiting for the result to appear. 📺

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

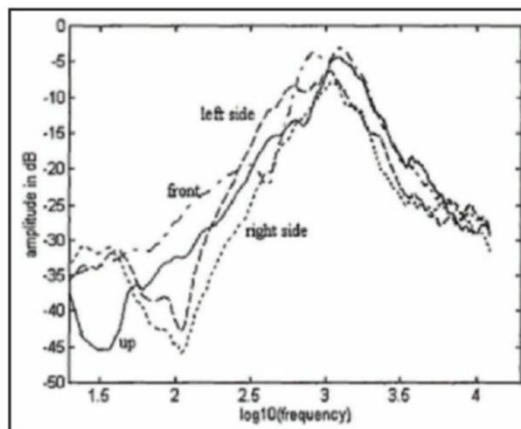


Figure 6: The spectrum produced by a pistol shot is similar to that produced by a firecracker, except that the spectrum emitted from the sides of the pistol is less concentrated in the 50Hz to 1kHz range than that of a firecracker, due to the directionality of the explosively expanding gas.



Photo 3: The NTi Audio DS3 dodecahedron loudspeaker is a calibrated broadband omnidirectional sound source.

About the Author

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



References

- [1] D. Griesinger, “Beyond MLS—Occupied Hall Measurements with MLS Techniques,” Figure 6, Audio Engineering Society (AES) Preprint 4403, presented at the 101st Convention in Los Angeles, CA, November 1996.
- [2] M. Horvat, K. Jambrosic, and H. Domitrovic, “A comparison of impulse-like sources to be used in reverberation time measurements,” Figure 8, *Journal of the Acoustical Society of America*, Volume 123, No. 5, June, 2008.
- [3] Horvat, et al: op cit, Figure 10.
- [4] Griesinger, op cit, Figure 2.

Resources

B. Waslo, “The IMP Analyzer,” *Speaker Builder*, January–April, 1993

Triode Common Cathode Stages, the Oracle Equation, and Theodosian Objections



By
Christopher Paul

For those readers interested in tube amplification, this article proposes a simplified equation of what might appear to be multiple mathematical factoids into a single concept, intended to help characterize and understand triode circuits.

With a title like this, you're probably wondering what you're in for. My motivation for writing this article is to propose a simple, straightforward means of enhancing the understanding of specific triode circuits and to settle certain controversies, some of which are well known and long standing. I hope you'll find this approach to be a mixture of simplicity, elegance, and technical accuracy. For completeness, I'll also cover disagreements with some of the conclusions it reaches.

The article will limit itself to AC small signal (low distortion) behavior. **Figure 1** is a generalized triode common cathode stage. Signal voltages are denoted by the letter "e". Subscripts of g, k, and a denote the grid, cathode, and anode terminals. When a sole subscript is present as is shown in the figure, the parameter is that of the subscript terminal with respect to ground. Cases of a reference terminal other than ground will be indicated with a second subscript. For instance, the differential cathode-to-anode voltage, time constant, and impedance (of the complete stage) are referred to as e_{ka} , TC_{ka} , and z_{ka} . Each of the stage types to be discussed has output(s) at the a, k, or a and k terminals. Only when a terminal is employed as an output, either to drive a subsequent circuit directly or an intervening cable, do one or both outputs encounter parasitic capacitances such as C_a and/or C_k . These are unavoidable and cause undesirable bandwidth limitations. Notably, the -3dB bandwidth of a stage is the reciprocal of $2 \times \pi \times TC$. The TC of a stage is equal to its output resistance times the capacitance that loads it (disregarding the effects of parasitics intrinsic to the triode). C_{large} is a physical capacitor that is optionally employed to increase the stage

gain $-e_a/e_g$. Typically, it's chosen so that its AC impedance is far less than the triode's r_a/μ . When it's used, consider the AC value of R_k in subsequent equations to be zero.

Let me add one more note. To minimize the clutter that is sometimes attendant to algebraic expressions, I'll be using the || operator to indicate "is in parallel with." For example, equivalent expressions for paralleled impedances A and B are:

$$A \parallel B = \frac{A \times B}{A + B} = \frac{1}{\frac{1}{A} + \frac{1}{B}}$$

The Most Popular Stage and the Oracle Equation

If we had to have only one stage, it would probably be the voltage amplifier. In this configuration, R_a is greater than R_k , and the output is taken from the anode. The equation [1,2,3] describing the relationship between e_a and e_g is:

$$e_a = -e_g \frac{\mu \times R_a}{R_a + r_a + (\mu + 1) \times R_k} \quad [0]$$

I call this the Oracle equation because with minor regroupings of terms, it can specify not only three different voltages of interest, but also three different impedances. It does this without recourse to complexities such as Thevenin's theorem and feedback theory. It's named after the Oracle of Delphi, which is claimed to have predicted future events in ancient Greece. Denoted by the letter O, "Oracle" is used to mean an authoritative, wise expression or answer. This equation can be slightly rearranged so that the numerator μ is associated

with the term e_g , and the sum of certain terms in the denominator can be assigned the value R_x :

$$e_a = -\mu e_g \frac{R_a}{R_a + R_x}$$

where:

$$R_x = r_a + (\mu + 1) \times R_k$$

This is an equation of a voltage divider whose source $-\mu e_g$ is driving load R_a through impedance R_x . Accordingly, for such a stage, it's clear that the ground-referenced impedance of the entire circuit at the anode (the junction of R_a and R_x) is given by [1,2,3]:

$$z_a = R_a \parallel R_x$$

In arriving at this result, no assumptions were made about the values of any components, and so it is generally true regardless of the type of stage. As the output is taken from the anode, the circuit's time constant is:

$$TC_a = z_a \times C_a$$

Another Useful Stage

Sometimes it's not voltage gain that's needed, but a low output impedance to drive an interstage feedback network or a parasitic capacitance that would otherwise limit bandwidth. Enter the cathode follower, whose output is taken from, well, the cathode (surprise, surprise.) The Oracle equation gives us the anode AC voltage e_a , which is expressed across resistor R_a . The anode current can be obtained by dividing the equation by R_a . In a well-designed stage, the grid current is negligible, and so the anode current is the cathode current. This current flows through R_k , across which the cathode voltage e_k is developed. The topology of the stage is such that the anode and cathode voltages have opposing phases. Putting all of this together, we have [1,2,4]:

$$e_k = e_g \frac{\mu \times R_k}{R_a + r_a + (\mu + 1) \times R_k}$$

We can pull the μ term from the numerator and a $\mu + 1$ term from the denominator and express this equation as:

$$e_k = \frac{\mu}{\mu + 1} e_g \frac{R_k}{R_k + R_z}$$

where:

$$R_z = \frac{r_a + R_a}{\mu + 1}$$

This is again a voltage divider equation whose source $\mu \times e_g / (\mu + 1)$ is driving load R_k through impedance R_z . For such a stage, the ground-referenced impedance of the entire circuit at the cathode is given by:

$$z_k = R_k \parallel R_z$$

Once again, no assumptions were made about the values of any components, and so this equation is also generally true regardless of the type of stage [1,2,5]. With the cathode as the output, the circuit's time constant is:

$$TC_k = z_k \times C_k$$

For a cathode follower, R_a is typically set to zero. If it's not, a little algebra shows that the only way that z_k can be equal to z_a is if R_a is equal to $R_k \parallel r_a / \mu = R_k \parallel 1/g_m$, where g_m is the triode transconductance. It's difficult to conceive of how such a circuit could be useful.

The Last Stage

Now we come to the Cathodyne aka the Phase Splitter aka the Concertina. The purpose of this circuit is to provide two outputs (anode and cathode) of identical amplitudes and opposing phases to drive a push-pull output stage. The driver output is the

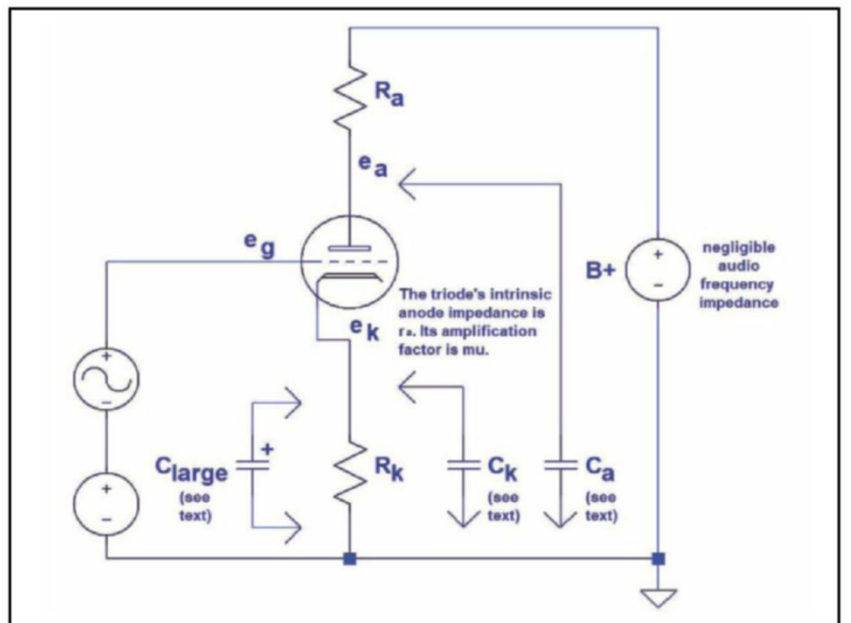


Figure 1: A schematic of a generalized triode common cathode stage

difference of e_k and e_a . We again call upon variations of the Oracle equation:

$$e_{ka} = e_k - e_a = e_g \frac{\mu \times (R_k + R_a)}{R_a + r_a + (\mu + 1) \times R_k}$$

Since the same current flows through R_a and R_k , for the output voltages to have the same amplitudes, these resistances must be equal. Calling them both R and re-expressing:

$$e_{ka} = e_g \frac{2 \times \mu \times R}{r_a + (\mu + 2) \times R}$$

We can pull a μ term from the numerator and a $(\mu + 2)/2$ term from the denominator and associate them with source e_g to obtain:

$$e_{ka} = \frac{2 \times \mu}{\mu + 2} e_g \frac{2 \times R}{\frac{2 \times r_a}{\mu + 2} + 2 \times R}$$

It's déjà vu all over again. We have a voltage source $2 \times \mu \times e_g / (\mu + 2)$ driving a load $2 \times R$ through an impedance $2 \times r_a / (\mu + 2)$. Does this make any sense? Yes, it does. Refer again to Figure 1. We know from experience that the amplitudes of e_k and e_a in a Cathodyne are slightly less than that of e_g . Since they are of opposing phases, their difference would be slightly less than $2 e_g$, so the voltage source portion checks out. One resistance of value R is connected to the anode and one to the cathode. These are interconnected through the comparably negligible AC impedance of the B+ supply and are therefore in series. Hence the triode sees a load of $2 \times R$. Therefore, we can confidently say that for a Cathodyne [6]:

$$z_{ka} = (2 \times R) \parallel [2 \times r_a / (\mu + 2)] = 2 \times [R \times \{r_a / (\mu + 2)\}]$$

Similarly, the cathode and anode are each connected to a parasitic capacitance, the pair of which is also in series (and interconnected by the B+

supply.) Since push-pull output stages are generally symmetric, capacitances C_a and C_k are typically equal, and we can call them both C . C in series with C gives us a capacitance of $C/2$. So for this stage's time constant, we have:

$$TC_{ka} = z_{ka} \times \left(\frac{C}{2} \right)$$

In the literature [10], $z_{ka}/2$ is named as the resistance that determines this circuit's time constant. And it is, when multiplied by C rather than $C/2$. This might tempt thinking of the triode as having two identical $r_a / (\mu + 2)$ resistances, one driving R_a and one driving R_k . But the literature's audio frequency triode models consist of a single resistance, not two [7,8]. And since there was no reason to double the number of triode resistances in our understanding of the previous two stages, surely the triode doesn't suddenly split one resistance into two equal halves simply because its anode and cathode are connected to resistors of identical values. The two-resistor and the Oracle's Cathodyne one-resistor models give us z_{ka} , but not z_a or z_k ; neither model affords a proper triode conceptualization [7,8].

The Benefits of the Oracle Equation

A single equation has been rearranged to derive the voltages and impedances of triode stages without resorting to theorems such as Thevenin's or to feedback theory. A case can be made for this being one of the most cogent and succinct derivations available in the literature. I hope that readers will see some value in it and that it might be useful in settling some of what The Valve Wizard calls "... heated debate in the pages of audio magazines for decades."

Theodosian Objections

The Roman emperor Theodosius was unlikely to have even heard of a triode, let alone to have disputed its characteristics. He had other things on his mind. In 390 C.E., he had the temple dedicated to the Oracle of Delphi destroyed as part of his effort to rid the empire of paganism. So while I admit I'm taking a bit of poetic license here, he is a fitting symbol for objections raised to the conclusions reached with the help of the Oracle Equation. In introducing these objections as the headings of subsequent paragraphs, I considered providing references. Most are from respected works of well-known authors. Doing so would have allowed the reader access to the objectors' reasonings. But I want to avoid being provocative. Authors of these objections can choose to ignore this article

About the Author

Christopher Paul was born in the US the year before the Nobel prize was awarded for the invention of the transistor, the development of which he claims no credit for. His love of music from an early age led to some puzzling experiments with musical instrument amplifiers, and from there to an interest in electronics. Chris finds it interesting to delve into relatively simple circuits and to investigate and write about aspects of their performance that are little known or appreciated. He enjoys deriving equations that can be used to evaluate and design them, although he admits to being in therapy for this. Chris has worked for companies whose products are in the fields of communications, electronic article surveillance, and enterprise hand-held computers. He holds two M.Sc. degrees in electrical engineering from Brooklyn Polytechnic (now part of New York) University.



if they wish. After all, neither they nor references to their works are being cited. Or they can join in the discussion at diyAudio (see the link at the end of this article) with anyone else who wishes to comment. Frankly, I've struggled a bit with whether to provide references, but have decided against it. The following are some claims I've encountered that deserve readers' consideration.

Considered as a stage with a set of differential outputs, the Cathodyne is a balanced driver.

The Institute of Electrical and Electronics Engineers (IEEE) definition of a balanced circuit (signal-transmission system) [9] is: "A circuit in which two branches are electrically alike and symmetrical with respect to a common reference point, usually ground." References [1,2,3,5] and work with the voltage amplifier and cathode follower versions of the Oracle equation establish that for a Cathodyne, $z_a \neq z_k$. Certainly impedances are electrical characteristics and, since in this case they are not alike, the Cathodyne cannot be considered to be a balanced circuit. The identical amplitudes of its desired output signals alone are insufficient to make it so. For comparison, a long-tailed pair terminated

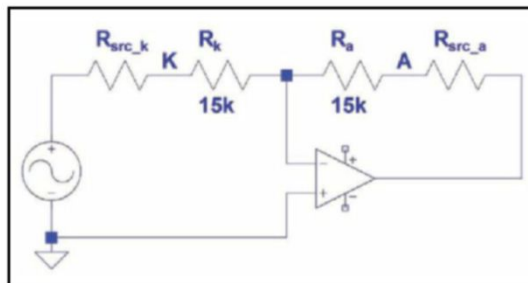


Figure 2: Although the values of loads R_a and R_k and the frequency responses of the voltages across them are equal, the ground-referenced impedances of output terminals A and K are unequal.

with identical loads and driven by signals from the opposite ends of a center tap-grounded transformer winding does qualify as a balanced driver.

A procedure that connects a test load between any two terminals of a balanced circuit to yield an unbalanced aggregate cannot be relied upon to determine the unloaded impedance between those terminals, since a balanced circuit is no longer being tested.

This claim is based on rhetoric, and there are no analytics to back it up. It's akin to claiming that one cannot test the output impedance of an amplifier with a zero-ohm load because then the circuit is no longer an amplifier. Thevenin would

CELEBRATING OVER 40 YEARS!

ACOUSTICS BEGINS WITH ACO™

MEASUREMENT MICROPHONES
Polarized & Electret

Windscreens
3 to 18 inch & Custom
Weather/UV Resistant

Community/Industrial
Noise Monitors/Alarms
Indoors and Out

Microphone Systems
Standalone
IEPE Powered
Phantom Power
Custom

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

DO IT YOUR WAY
Panels/Boxes/Cables
WITH REDCO AUDIO'S
"Build Your Own" web pages

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

REDCO AUDIO
www.redco.com 203-502-7600

certainly disagree with the claim, as he cares not whether a circuit is balanced—Thevenin applies to “any two terminals of a linear circuit.” Thevenin tests with ground-referenced loads connected to a stage’s anode and then to its cathode would yield the same conclusions as the Oracle equation’s voltage amplifier and cathode follower versions. Confirmation of this is left to any doubting reader.

Series feedback through R_k reduces the impedance of the Cathodyne anode.

Impedances are characteristics of a pair of terminals in a circuit, not of a single terminal such as an anode. (Imagine a claim that there is an impedance of a single terminal of a resistor!) By default, when no other terminal of a pair is mentioned explicitly, it is implicitly assumed to be ground. Three separate references herein and a version of the Oracle equation establish a formula in which z_a increases as R_k is increased. These would falsify the claim if the “other” terminal were ground, but not if it were the cathode.

The term “effective impedance”...

...is sometimes encountered in the literature. You might infer something about it from the context in which it is used, but caution is appropriate. Apparently, “effective impedance” is different from impedance, or “effective impedance” wouldn’t have been invented. This term requires its own definition. If that definition is not forthcoming, then the term is useless for explaining anything—it’s simply “hand

waving.” Rely upon what the Oracle equation reveals, and you’ll be on firm footing.

When a Stage has equal anode and cathode loads R_a and R_k , the frequency responses at the anode and cathode terminals are equal, and so the output resistances must be equal.

Actually, the frequency responses are only approximately equal because of certain triode parasitic capacitances, but these effects are minor. As mentioned earlier, resistances are characteristics of pairs of terminals, not of single terminals. It is difficult to give the claimant the benefit of the doubt that the other terminals of the anode and cathode are the cathode and anode, respectively. Surely the trivial fact that $z_{ak} = z_{ka}$ is not what is being asserted. The only other conceivable terminal is ground, and the voltage amplifier and cathode follower versions of the Oracle equation are clearly incompatible with this possibility. So the claim must be rejected. There is yet another reason for rejection—the claim’s logic.

Consider **Figure 2**. Although this is not a model of any triode stage, the circuit satisfies the claim’s predicates. That is, the resistors’ R_a and R_k values are identical, and their voltages’ frequency responses are the same because the same current flows through each resistor (negligible current flows into the op-amp inputs.) The op-amp forces a (virtual) ground at the resistors’ junction. Therefore, the impedance z_k at terminal K is $R_k \parallel R_{src_k}$. If we connect a test load between terminal A and ground, the current through R_a and R_k is negligibly affected (subject to the op-amp’s adequately high open loop gain and its output not clipping). Therefore, the voltage at terminal A is virtually unchanged. This is consistent with a terminal A impedance z_a of nearly zero ohms! So even though $R_k = R_a$ and the frequency responses at terminals A and K are identical, $z_k \neq z_a$. Why should the Cathodyne be any different? The claim remains rejected; there is no requirement that a circuit’s output resistances across identical loads be equal simply because the loads’ voltages’ frequency responses are the same.

Conclusion

Manipulation of the Oracle Equation provides perhaps the most simple and concise derivation of triode common cathode stage voltages and impedances extant. Various objections encountered in the literature to some of its conclusions have been presented, as have reasons to reject them. Hopefully, this provides some benefit to the audio community. Comments are welcome at diyAudio: www.diyaudio.com/community/threads/triode-common-cathode-stages-and-the-oracle-equation.385904. 

References

- [1] A. Preisman, “Notes on the Cathodyne Phase-Splitter,” *Audio* magazine, April 1960.
- [2] G. E. Jones Jr., “An analysis of the Split-Load Phase Inverter,” *Audio Engineering*, December 1951.
- [3] *The Valve Wizard*, equation 12.1 and 12.5, www.valvewizard.co.uk/cathodyne.pdf
- [4] *Ibid.*, equation 12.2
- [5] *Ibid.*, equation 12.6
- [6] *Ibid.*, equation 12.7
- [7] M. S. McCorquodale, Ph.D. “Vacuum Tube Preamplifier Analysis and SPICE Simulation, August 2022, <http://web.eecs.umich.edu/~mmccorq/diversions/simulation/index.php>
- [8] ITT, *Reference Data for Radio Engineers*, p. 17-7
- [9] *IEEE Standard Dictionary of Electrical and Electronic Terms* (see a balanced circuit (1) signal-transmission system)
- [10]: F. Langford Smith, *Radiotron Designer’s Handbook*, 4th edition, p. 330.

BluOS™

Hi-Res Distributed Audio Platform



BluOS® offers a complete whole-home music ecosystem with an intuitive user interface and integrations with popular smart home control systems. Native in the hi-res WiFi-connected streamers, amplifiers, and speakers from Bluesound, PSB Speakers, and NAD Electronics' Custom Install line, BluOS offers distributed audio solutions for every lifestyle, from hi-fi enthusiasts to the digital music lover.

Participate in our CEDIA 2022 Seminars on Thursday Sept. 29th at 10 am or Friday Sept. 30th at 11am in Room C155. Visit lenbrookamericas.com/cedia2022 to learn more about the sessions.

Control System Partners



Audiophile Quality



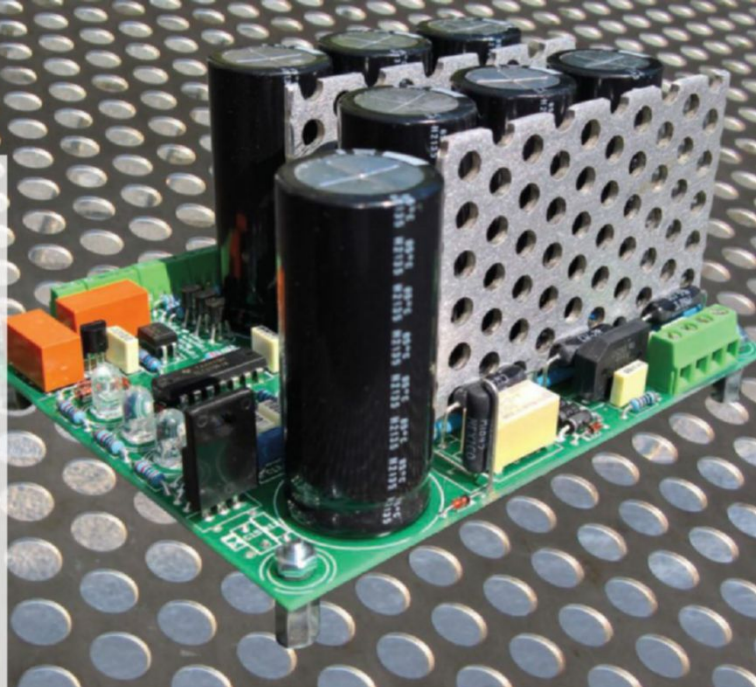
Come visit us at CEDIA 2022 - Booth #19048

September 29 – October 1 Dallas, TX

Making a DC Power Supply with Audible Benefits (Part 1)

Choosing the Design

By
Frans de Wit



Following the two articles exploring multiple power supply circuits with potential for high-end audio applications (*audioXpress* December 2021 and January 2022), Frans de Wit details a high-quality power supply especially targeted toward preamplifiers, record-players, and other low power appliances.

Big goals, such as building a “molten salt nuclear reactor” or a “space craft capable of light speed” are not my ultimate calling, for me it is building the ultimate “high-end audio system.”

In previous articles that I wrote for *audioXpress* (December 2020 and January 2021 publications), I talked about voltage regulation and the accompanied circuitry needed. Now, I would like to take it a step further and discuss building a high-quality power supply especially targeted toward high-quality audio circuits, such as preamplifiers, record-players, and other low-power appliances.

The Design

I considered many factors that would be needed to make this project successful. A few of the more obvious ones include: a good power-factor (e.g., no pulsed currents) and full galvanic isolation from the mains. I also want it to block high-frequency environmental disturbances with no crosstalk (by using “local” power converters and regulation). I also want to eliminate ground loops (by using “local” power converters and regulation).

There are several benefits to including these factors: Missing cycle detector (e.g., watchdog), an

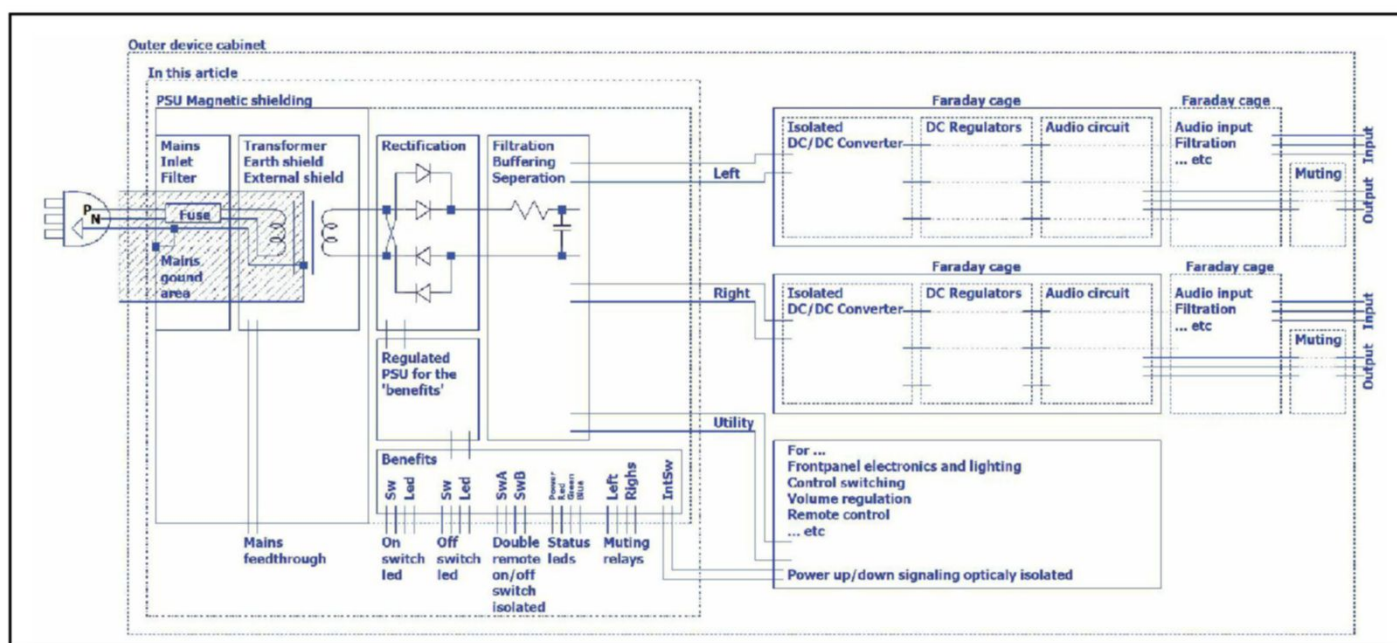


Figure 1: The block diagram shows the design for this power supply unit.

AC output (direct transformer output), a front-panel LED connector (on, muted, playing), and DC output (direct third-channel DC output for control tasks). There would also be separate On/Off push-buttons with status LEDs that have a safety hold/drop. The device would also have an optically isolated output to control an external device (e.g., an amplifier), and a muting output that can drive left and right separated relays.

Using the criteria we have just discussed, I know now what I will be designing and (possibly) building. This article will include all the data needed to create the PCBs and metal work and, as with the previous article, all simulation data will also be available. **Figure 1** shows the block diagram for this project.

One of the main issues this power supply will address is noise induced/injected/leaked by the “outside world.” By this I mean the disturbances that try to enter our high-end devices from the mains and the ether. For this reason, we will create a galvanically isolated mains interface (everything left from the transformer except the secondary side of the transformer). For this reason, the mains inlet filter and the mains transformer are placed in a partially magnetically sealed environment that is isolated from the outer enclosure and all other internal electronics. The mains filter comes in its own metallic box that also includes the fuse(s). The transformer is also packaged in its own metallic enclosure, and it has an earth shield that guarantees full isolation from the mains and many of its disturbances. For this project, I chose the Schurter 3-107-465 for the AC power inlet (**Photo 1**) and the Toroidy TSAS40V 50VAC for the transformer (**Photo 2**).

Let’s talk a bit more about these components. The AC power entry module contains three major components — the on-off switch, the fuse (or fuses), and a mains line filter. The switch is needed to turn the whole thing on and more importantly to turn it off. The fuse is like a switch except that it can only turn off after one single use, then it has to be replaced.

I do like the power inlet with two fuses, because it has a fuse in both current carrying wires. This is important, especially if the power installation is using un-polarized connectors, which means the mains phase can connect to either of the two sides of the power module, and it is better to be safe than sorry. Finally, the filter is there to prevent high-frequency signals from entering high-end applications. We really do not need that high- frequency signal because it will interfere with the electronics and by extension our music or other signals [1].

Secondary Circuit

Next, we will discuss the secondary circuit, which consists of the rectifier and the filtration. The filtration circuit will also be used to improve the power factor and to lower the peak currents generated. In my opinion, it is of the utmost importance to create a magnetically clean environment and low currents are a main concern. For this reason, the DC supply will have a relatively high voltage (a minimum of 50V DC) and there will also be three outputs (separated for left, right, and



Photo 1: For this project, I chose to use the Schurter 3-107-465 power entry module.

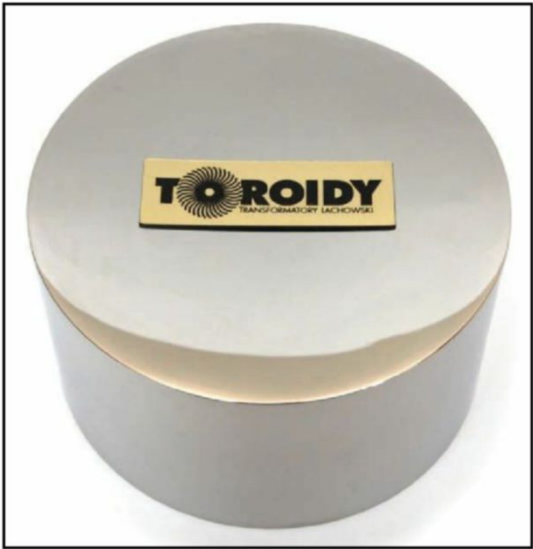


Photo 2: The transformer that I selected was the Toroidy TSAS40V 50V AC.

Specifications	Measurements
Secondary open (e.g., unloaded)	36.7V AC
Primary series resistance	30.7Ω
Secondary series resistance	5.9Ω
After rectification, an a 1mF filter capacitor (unloaded)	49.2V DC

Table 1: Here are some of the measured specifications for the 30V AC and 15V A toroid transformer that I chose from the bin.



Photo 3: The little transformer (shown here) almost behaved as expected, with the output voltage near 30V DC at a full load of near 500mA DC for an output power of near 30W.



Figure 2: The oscilloscope trace, at full load, looks like a square wave with less steep flanks.

About the Author

Frans de Wit was born and raised in the Netherlands. He went to high school to become an electrical engineer, and found his first job in electronics retail, selling components over the counter and advising customers on component selection and application. Here he was spotted by a headhunter and invited to work for the Dutch importer for Motorola, Texas Instruments, Hewlett Packard, and many other great brands of the time. There he started his career as a Desk-Sales Engineer. During this time, he helped customers building trains, organs, milk-floats, and particle accelerators (yes, the one at CERN) among other things. Next, he co-founded an IT company and until 2016 he worked as a programmer and algorithms specialist building operating systems, compilers, and a large DAM system. In 2016 he achieved a long-time ambition and founded an audio research and development company named Signature Origin.



utility). The reasoning behind this is that a higher voltage will need to deliver a lower current, three voltages will also lower these currents and enable separate filtering for the left, right, and utility power sources. The input impedance of the filtering will make ground currents between the multiple power supply impossible (as long as this impedance is effective for the + and – side of the supply).

Before we start looking at simulations (virtual proof of concept), let's have a look at one of the transformers that I picked from the bin. This is a 30V AC and 15V A toroid transformer. Some of its measured specifications are shown in **Table 1**.

Then, after some handy work, I came up with an adjustable current sink made from a handful of resistors, a MOSFET, and a large heatsink. I loaded the output voltage until it was supplying 500mA to the MOSFET. The "expected" output voltage went down to 33V DC and the AC output voltage of the secondary coil went to 29VRMS.

That little transformer (almost) behaved as expected (**Photo 3**), with the output voltage near 30V DC at a full load of near 500mA DC for an output power of near 30W. One thing that must be noted is the unloaded rectified output voltage of 49.2V DC (measured) where a 30V AC transformer is expected to generate $30V\ AC \times \sqrt{2} = 42.4V\ DC$, thus about 7V over the calculated expectation. Concluding that a transformer may create near 15...20% more voltage than one would get from calculating the peak voltage as specified for full load (e.g., 42.4V). Later we will be creating a power supply around the earlier mentioned Toroid transformer of 50V. The maximum voltage that we would design for would then be $50V\ AC \times \sqrt{2} \times 1.2 = 84.8V\ DC$. So we need to select components that withstand 85V or higher.

Final note: The oscilloscope trace (**Figure 2**) looks like a square wave with less steep flanks. The input voltage of the transformer is the sine wave as supplied by the local mains supply. The transformer is not (as often expected) outputting the same waveform measured at the input. The magnetic field in the transformer stores energy and thus will delay and distort as needed, the coupling factor of the transformer does not allow for a large energy residual in the transformer. Any residual is mostly converted to heat (see the Resources at the end of this article).

Square Wave Added

First, I want to discuss the power supply of the power source and some other circuit parts and components. The power supply that powers the virtual power supply in all the following simulations is known as a "Behavioral Voltage Source." It does

not supply much functionality except for one function, it takes a formula and changes it into a (virtual) voltage. Among its benefits are that it is lightweight, simple, fast simulating (as fast as the solver can solve the accompanying formula), and it is able to create headaches (just kidding ... I think).

The next part will contain some formulas, so if you want to skip them then go directly to the simulated circuits.

I'm happy you are still here. Let's see what that voltage source is feeding into the virtual transformer. First, the peak mains voltage ($V_{MainsP} = 340V_{Peak}$) at the mains frequency ($f_{Mains} = 50Hz$) — as specified in the Project Files included in the Supplementary Material on the audioXpress website. In addition to this is a “noise” signal ($V_{Noise} = 1V$) with the specified frequency ($f_{Noise} = 1kHz$) added. To make it even more interesting, the possibility to change the phase

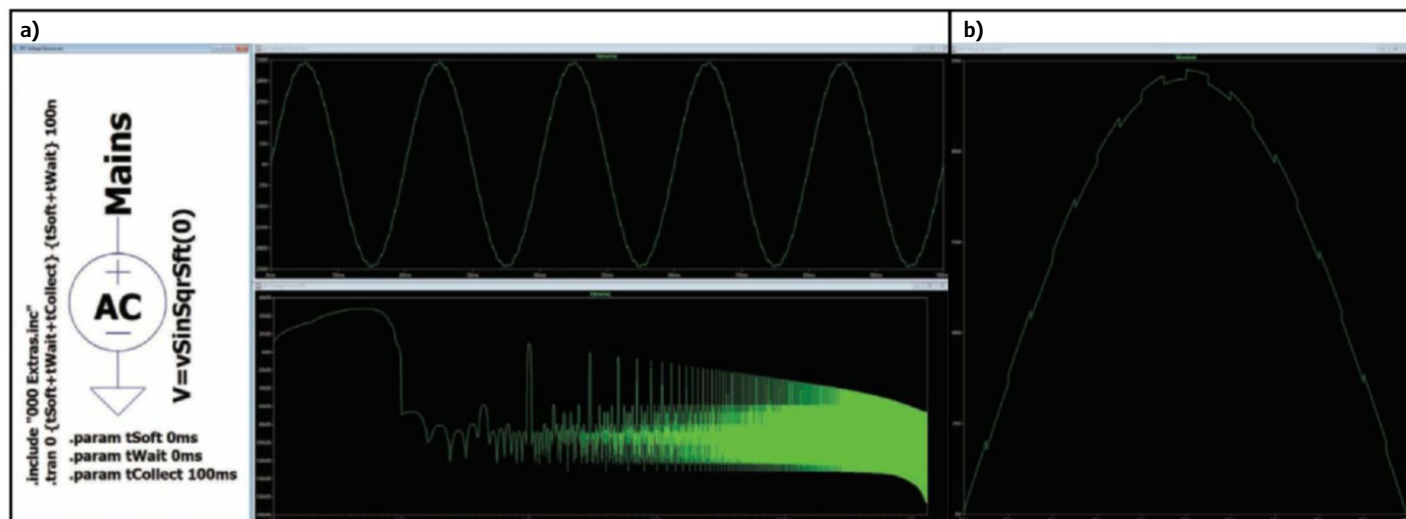


Figure 3: Figure 3a and Figure 3b show the noisy sine wave, the fast Fourier transform (FFT) that belongs to it, and a detail of the sine wave that clearly shows the 10% square wave of 1kHz that was added.

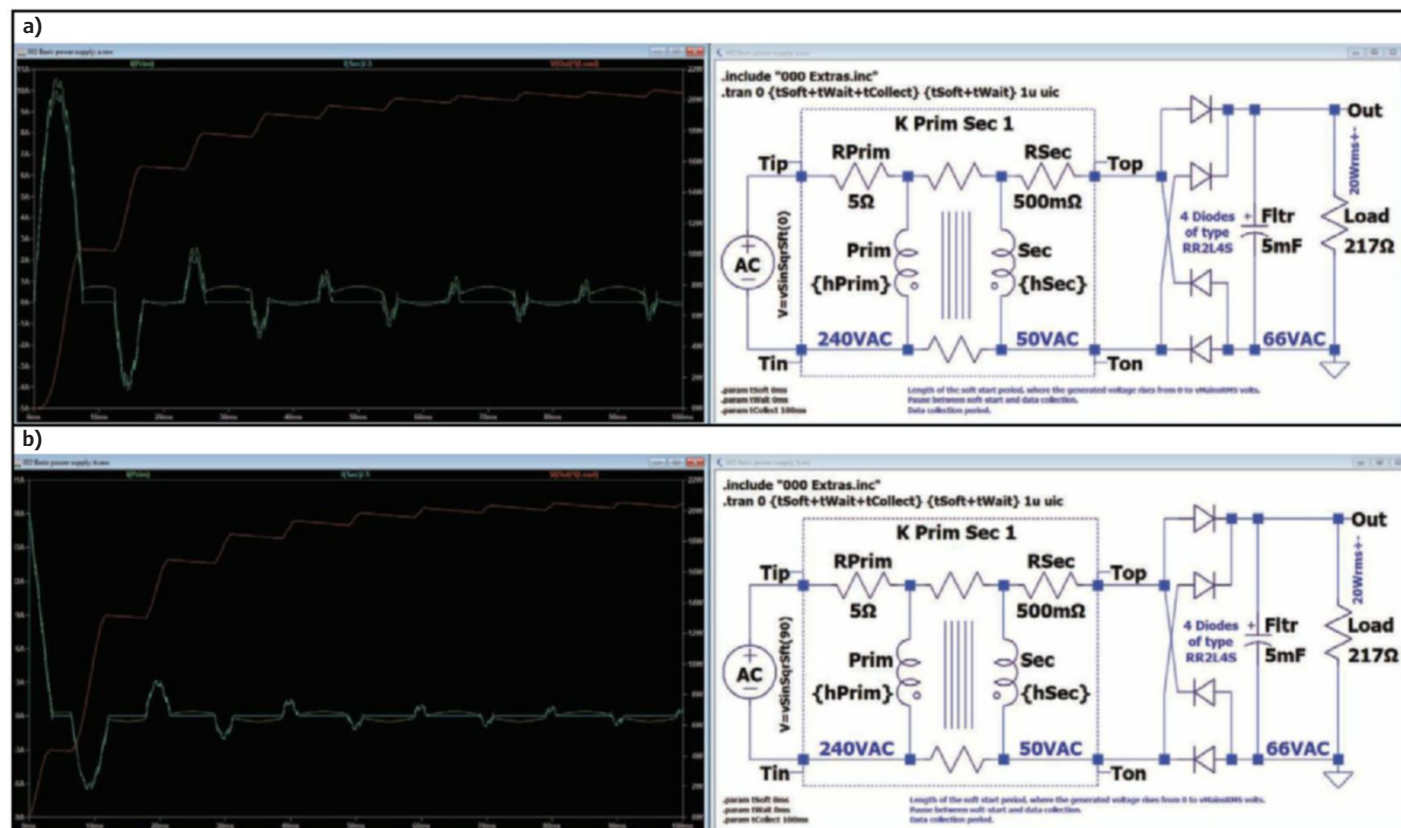


Figure 4a: The “standard” power source and filtration is shown. The green line ($I(Prim)$) is the mains input current, the current measured in the primary coil of the transformer. Clearly visible is the noise that was added. b: This graph shows how wrong it can go. The 2400VA is the “best” case.



is added to all this, which is made possible using a voltage source named “AC” and calculated by the formula $v\text{SinSqrSft}(p\text{Mains})$. The settings shown here can be found in the file named 000 Extras, which is posted in the Supplementary Materials section of the audioXpress website. (This file and its use will be discussed later in more detail.) Figure 3a and Figure 3b show the noisy sine wave, the fast Fourier transform (FFT) that belongs to it, and a detail of the sine wave that clearly shows the 10% square wave of 1kHz that was added. Because a square wave has been used, we see many high-frequency components of that in the FFT. It will be interesting to see what happens to this during the development of the power supply unit (PSU).

The Simulated Circuits

So, without further ado we will start looking at the design process for this power supply, and we will start with simulated circuits — first looking at the conventional approach and then

we will improve on it. Later, we will talk about that high (50V DC) voltage selected and what we can do with it.

Figure 4a shows the “standard” power source and filtration. The green line ($I(\text{Prim})$) is the mains input current, the current measured in the primary coil of the transformer. Clearly visible is the noise that was added. It even looks as though it is being enhanced here when compared with the original wave forms shown in the previous graphs.

The blue line ($I(\text{Sec})/-5$) is the secondary current as measured in the secondary transformer coil, the size, relative to the primary current, has been multiplied by -5 to make it easier to compare to the primary current.

The red line ($V(\text{Out}) \times I(\text{Load})$) is the power delivered to the load in watts. The amount is tuned (by hand, via a trial-and-error methodology) to 20W as will be done in all the following simulations.

The schematic diagram being simulated has only five parts, from left to right: mains supply, transformer, bridge rectifier, bulk

```
.options numdgt = 7
.options method = gear
.options plotwinsize = 0

;Prefixs in use
; c = capacity in farad
; f = frequency in hertz
; h = inductance in henrys
; p = phase in degrees
; r = resistance in ohm Ω
; t = time in seconds
; v = volts in volt
.param hPrim
.param hSec hHPrim2HSec(hPrim,vMainsRMS,vOutRMS)
.param fMains 50
.param vMainsRMS 240
.param vMainsP vMainsRMS*sqrt(2)
.param vMainsPP vMainsP*2
.param vOutRMS 50
.param vOutP vOutRMS*sqrt(2)
.param vOutPP vOutP*2
.param fNoise 1kQ
.param vNoise vMainsP/100
.param tSoft 350ms
.param tWait 50ms
.param tCollect 100ms

2.5H Primary inductance
Secondary inductance
Mains frequency
Mains RMS voltage
Mains peak voltage
Mains peak to peak voltage
Output RMS voltage
Output peak voltage
Output peak to peak voltage
Noise frequency added
Noise voltage added
Soft-start time spam
Waiting time spam
Data collection time
```

Listing 1: Here are the missing settings, the secondary inductance is calculated using the $h\text{HPrim2HSec}()$ formula that has three parameters. The $t\text{Soft}$, $t\text{Wait}$, and $t\text{Collect}$ parameters are preceded by a semicolon that indicates that these need to be provide from the schematic drawing.

```
.func hHPrim2HSec(hPrim,vPrim,vSec) {hPrim/(vPrim/vSec)**2}; Compute hSecondary for hPrimary, vPrimary and vSecondary
.func tPhase2Time(fFrequency,pPhase) {((1/fFrequency)/360)*pPhase}; Change pPhase in degrees to tPhase as time for the
given frequency
.func vSine(pMains) {sin([2*π*fMains]*(time+tPhase2Time(fMains,pMains)))*vMainsP}; Calculate a vSine for fMains, pMains
and vMainsP
.func vSinSqr(pMains) {vSine(pMains)+vSquare()}; Compute vSinSqr = vSine + vSquare where vSine has a phase offset of pMains
.func vSinSqrSft(pMains) {if(time<tSoft,time*(1/tSoft)*vSinSqr(pMains),vSinSqr(pMains))}; Compute vSinSqr = vSine + vSquare
where vSine has a phase offset of pMains using a soft start period of tSoft
.func vSquare() {if(int(time*fNoise)-(time*fNoise)<0.5,-vNoise,+vNoise)}; Calculate a vSquare for fNoise and vNoise
```

Listing 2: These are the formulas used to create our simulations.

```

hHPri2HSec(...) Compute the secondary inductance of a transformer given the primary inductance, the primary voltage
and the secondary voltage.
tPhase2Time(...) Convert a phase in degrees to a time in second given the frequency and the phase.
vSine(...) This will return the data needed for a voltage source to generate a sinusoidal wave form including the
specified phase (time offset in degrees)
vSinSqr(...) The resulting waveform description for adding vSine(...) and vSquare(...)
vSinSqrSft(...) The resulting waveform description for combining vSinSqr(...) with the calculated soft start slope where
the slope has a length of tSoft seconds going from 0 to 100% in that time period.
vSquare() Generates the data needed to create a square wave the frequency is fNoise and the voltage is vNoise, the
date is taken from the data table shown earlier.

```

Listing 3: The descriptions for those formulas are shown here.

capacitor, and a load. In the following circuits, they will be mostly the same, some values may be changed, and some components may be added, when that happens it will be explained.

So, what is wrong here? We used a nice large capacitor (Fltr) and still the output voltage (red line) looks ugly, lots of high residues, in the form of an almost triangular output wattage (on top of an 20W offset). This is surely not what we want to use as a supply for our high-end audio circuit. It is some sort of a “bummer” spending a lot of money for that huge 5mF capacitor, the simulation is even using a theoretical optimal version (no hold-back on quality), and still, it looks like dirt!

Maybe even worse is the input current (green line), a run-in current of almost 10A, and short loading pulses where that “nice” large buffer capacitor is loaded, at near 1.5A as can be seen in the blue line. This is a lot (for a 20W power supply and on the primary (e.g., 240V side), the output current (20W/217Ω) is expected to be near 92mA and thus is the input current (hPrim) expected to be about 19mA (92mA × 50V/240V), there should be no high currents in a power source as shown by the green line.

Note that the 10MΩ resistors (there is no value indicated in the schematic) represents the ground leakage from the mains ground to the internal ground circuit. In the final circuit we will break this connection by using a transformer that contains a ground shield.

You may wonder why, occasionally, the mains fuse blows when I switch my power supply on. Have a look at this simulation, the mains transformer shows a run-in current of near 10A (240VAC × 10A=2400VA), which is quite a bit.

Figure 4b shows how wrong it might go. The 2400VA is the “best” case. Look at the schematic shown in Figure 4b (specifically vSinSqrSft(0)). The only thing that changed was the turn-on phase, it was 0 degrees in the previous simulation, here it was selected to be 90 degrees (vSinSqrSft(90)).

As shown in Figure 4b, the run-in current has gone to 18.6A, that is 4464VA, surely many fuses, especially magnetic fuses, are known to be triggered in one single pulse, will start to protest by executing their “only purpose in life” — they will break the flow of current and you will sit in the dark.

About the Project Files

Now we know how a simple conventional power supply functions and how it fails. There is more to share but we will learn that while examining the why to a solution that works. But before we do that you might have noticed that there is very little data in these

simulation files, data such as mains voltage, mains frequency, transformer inductors, formulas, and more. This is because the information is located in an included file (000 Extras.inc), found in the Supplementary Materials section of the audioXpress website. This file contains all the additional data, which keeps the presented simulation schematic, well, manageable. **Listing 1** shows a sampling of the data. **Listing 2** shows the formulas that were used and **Listing 3** provides descriptions of those formulas.

That’s it. It’s there and that is all we need to know. From here on, we will take this data for granted. Let’s do some real stuff... In Part 2 of this article series, I will show some simulations that provide real improvements. 

Project Files

To download additional material and files, visit:

<http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

References

[1] “Luminiferous aether,” *Wikipedia*,

https://en.wikipedia.org/wiki/Luminiferous_aether#cite_note-1

Resources

“Coil and Transformer Calculator,” www.dicks-website.eu/coilcalculator

“LTspice,” Analog Devices, www.analog.com/en/design-center/design-tools-and-calculators/ltspice-simulator.html

“Mutual Inductance and Basic Operation, Chapter 9 - Transformers,” *Lessons in Electronic Circuits, Volume 2*, All About Circuits, www.allaboutcircuits.com/textbook/alternating-current/chpt-9/mutual-inductance-and-basic-operation

A. Roderick, “Basic Inductance Principles in Transformers,” *EE Power*, June 2021, <https://eepower.com/technical-articles/transformer-operating-principles/#>

“Series: DD12 IEC Appliance Inlet C14 with Filter, Fuseholder 1- or 2-pole, Line Switch 2-pole,” Schurter Electronic Components, datasheet, www.schurter.com/en/datasheet/DD12

“TTSAS0040 - SUPREME AUDIO grade transformer TSAS40VA - voltage to 50V,” Toroidy Transformatory Lachowski, datasheet, https://sklep.toroidy.pl/en_US/p/TTSAS0040-SUPREME-AUDIO-grade-transformer-TSAS40VA-voltage-to-50V/379

All American Tower Speakers

By
Ken Bird

In this article, our experienced speaker builder takes us along with him on his quest to build a loudspeaker made solely from drivers manufactured in the United States.



Photo 1: This image shows the completed All American Tower system.

With the majority of raw speaker drivers being imported today I wanted to build a system using only drivers made in the US. That does not mean that there is a quality issue with imported drivers, it's just the opposite. The quality and workmanship of drivers made overseas is usually superb with state-of-the art engineering and materials, but I wanted to see what was available in the US.

I began researching "American-made loudspeakers" on the Internet and found many listed but most of them made complete systems, and many high-end models that while assembled in

the US, used imported drivers. I finally settled on a brand that I remembered from my early days in engineering with a major manufacturer of consumer electronics now long defunct. The manufacturer bought carloads of speakers from US manufacturers and one was a Wisconsin-based firm called Oaktron that had been established in 1957.

Oaktron had remained in business for decades addressing the US original equipment and replacement market. In 2016, Oaktron was acquired by another longtime speaker manufacturer, Minneapolis Speaker Company, now known as MISCO. MISCO was established in 1946 by World War II Air Force veteran Cliff Digre, who after taking a National Radio School course, started a radio repair business and eventually a speaker reconing business that morphed into a speaker manufacturing business in 1956. The company, now under the direction of Cliff's son, Dan Digre, is one of the most innovative in the industry, serving the original equipment market. For years, if you wanted to buy a MISCO speaker you had to go through a local distributor but the distribution market changed with the advent of the Internet, which means MISCO will now sell you a speaker direct via www.miscospeakers.com.

This system is a simple two-way tower mass loaded design incorporating a 6" woofer and dome tweeter selected from the Oaktron line of MISCO speakers and made in the US. The crossover is a two-way design, incorporating an impedance compensator and crossing at 3.5kHz. The completed system is shown in **Photo 1**.



Photo 2: I used the Oaktron woofer and tweeter in the speaker system.

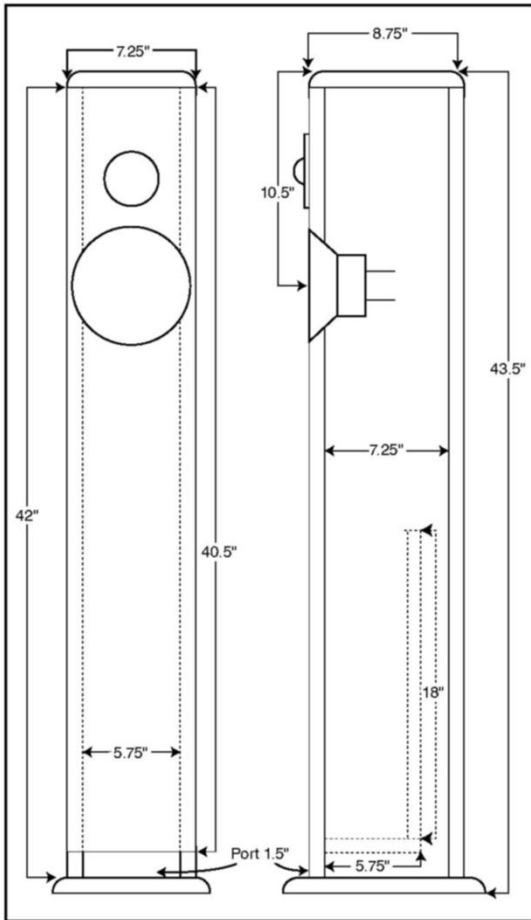


Figure 1: For this design, I used some pieces of pre-sawed oak plywood 7.25" wide and 42" long that worked out fine for a tower design.



Photo 3: The wood parts for one enclosure are shown.

The Drivers

The drivers selected are the Oaktron 165-WF08-01 6" woofer and the 25-TD08-03 dome tweeter. The woofer has a 41Hz resonant frequency and a usable midrange response that dovetails well with the tweeter (**Photo 2**).

The Enclosure

Thanks to a friend who builds custom kitchen cabinets I was offered some pieces of pre-sawed oak plywood 7.25" wide and 42" long that worked out fine for a tower design (**Figure 1**). The builder can use any type of wood or medium density lumber substitute. The dimensional lumber size would be 1' x 8", which is also 7.25" wide. The wood parts for a single enclosure are shown in **Photo 3**. Note that the internal tunnel section is 5.75" wide. The cut out for the tweeter is 3.58" (91mm) and the woofer cut out is 5.787" (147mm).

Assembly of the Enclosure

After cutting the sides to the dimensions shown, cut out the speaker holes using a router with a Jasper circle jig or a saber saw. Note the "mouse ear" cut outs for the tweeter. Before assembly, staple a 4" wide piece of outdoor carpet to the blind side of the tunnel (**Photo 4**). Do not place carpet on the outside wall of the tunnel. Begin the assembly by first attaching one side to the front panel and attach the tunnel partition. Use of a nail gun is



Photo 4: This image shows the application of the internal carpet in the tunnel area.

Photo 5: The carpet material is applied to the interior of the sides and top.



Figure 2: The crossover frequency is 3500Hz. The builder can use any other assembly method desired, including a custom PCB to mount the parts. Just wire as shown in the schematic.

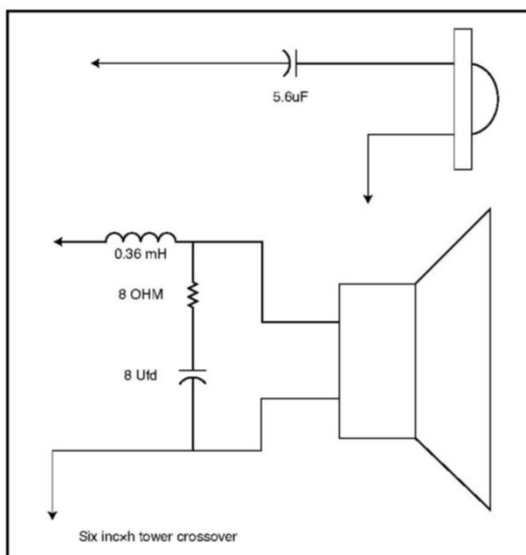


Photo 7: The crossover is mounted directly behind the woofer for ease of installation and maintenance.



Photo 6: The assembled enclosures are ready for stain.

handy but 2" finishing nails will work also. Use a nail set to drive the nails in 1/8 inch and stainable wood filler to seal the holes.

Spread wood glue on the face edges of the boards using a putty knife and keep a damp rag handy to wipe off any leakage from the joints. I also used corner clamps to keep the walls aligned when nailing. Caulk all the inside joints to assure a good air seal using a caulk gun or do it by hand for the more difficult areas to reach.

Cut the outdoor carpet material in 5" widths and staple as shown to the enclosure inside walls (**Photo 5**). Add the other side, stapling the carpet the full length of the side, and then stuff 3 ounces of polyester pillow stuffing (see the side bar on stuffing) into the blind end of the tunnel and try not to compress the material. Attach the back of the enclosure after stapling a 45" length of speaker cable to the inside wall then add the top and bottom pieces.

Finishing the Enclosure

The ends of the plywood will show unless they are either covered with a 3/4-inch veneer strip available from a home supply store, or as was done with this project, using 3/4-inch screen door trim nailed to the edges. This type of trim is available plain or with a decorative pattern embossed in the wood. For this project, I used the plain material. Next, I lightly sanded the enclosure surface and stained with a dark type that I had on hand. After staining the enclosure, I applied two coats of semi-gloss polyester varnish (**Photo 6**).

About the Author

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



Installing the Crossover and the Drivers

The crossover is a first-order Butterworth-type with an impedance compensator hand wired and mounted to a block of 3/4" pine scrap. The crossover frequency is 3500Hz. The builder can use any other assembly method desired, including a custom PCB to mount the parts. Just wire as shown in the schematic (**Figure 2**). The parts were chosen from a supply that I had on hand, but new substitutes are available as shown in the Parts List. In the interest of disclosure, I can't be guaranteed that all the parts listed for the crossover were made in the US.

I attached the crossover network in the back inside wall, centered just under the woofer and attached with two 1-and-5/8-inch drywall screws (**Photo 7**). I pre-wired the woofer and the tweeter leads and I attached the feed wire from the speaker terminal prior to mounting the crossover.

After installing the crossover make sure the speaker leads are outside the cut-outs and then add 6 ounces of polyester stuffing to the speaker's upper chamber. (See the side bar on measuring the stuffing.)

Before installing the drivers place a strip of door gasket foam on the inside of the driver mounting

Parts List

Enclosure parts are based on dimensional lumber sizes for two enclosures.

Number Item

Lumber

- | | |
|---|--|
| 6 | Side and rear panels, 42" x 7.25" |
| 2 | Front panels, 7.25" x 40.5" |
| 2 | Top pieces, 7.25" x 8.75" |
| 2 | Bottom pieces, 7.25" x 10.75" |
| 2 | Tunnel partitions comprised of one piece 0.75" x 5.75" x 7.75" and one piece 0.75" x 5.75" x 18" |

Drivers

- | | |
|---|---|
| 2 | Oaktron dome tweeters, Model 25-TD08-3 |
| 2 | Oaktron 6.5" woofers, Model 165-WF08-01 |
| 2 | 60" lengths of 18-gauge speaker wire |

Crossover Parts for Two Circuits

- | | |
|---|--|
| 2 | 5.6µF 100V non polarized capacitors, Part Express #027-334 |
| 2 | 8µF 100V non polarized capacitors, Part Express #027-338 |
| 2 | 0.35mH 20-AWG air coils, Part Express #255-030 |
| 2 | Resistors—8Ω 10 precision audio grade, Part Express #DNR-8 |

Speaker Terminal

Multiple binding post type, Part Express #260-292

Stuffing and Wall Lining

- | | |
|--------|---|
| 1 bag | Polyester pillow stuffing, Hobby Lobby or WalMart |
| 1 yard | Indoor-outdoor carpet, Menards SKU: 7674633 or equivalent |

Miscellaneous

- | | |
|----------|---|
| Nails | Wood glue |
| Caulking | Optional Woofer grilles, Parts Express #260-422 |

GETTING THE MOST OUT OF
VACUUM TUBES

By Robert B. Torner

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

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

In his refreshing, down-to-earth style, Mr. Torner sheds light on the "Why so many tube types?" "What About Tube Testers?", etc.

Other Interesting and Informative Topics:

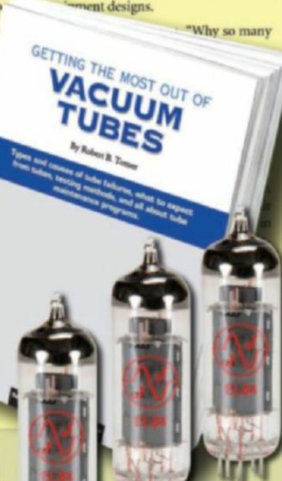
- 10 reasons why tubes fail
- Predicting tube performance
- Selected and preferred tube types
- Lengthening tube life

Anyone who has anything at all to do with tube equipment design, or proper use—should read this book. It is a must for anyone who produces or maintains consumer goods such as radio, TV, and production and manufacturing processes—or even computers and radar gear.

You'll also learn about:

- Catastrophic failures—what they are, when and how they occur
- Degenerative failures—what happens to tubes over time
- Subjective failures—why a seemingly normal tube may not perform as expected
- Characteristic variables—differences that make one tube type better than another
- Special-purpose tubes—considerations for filament and thyratron types

Order your copy at cc-webshop.com



ALL ELECTRONICS CORPORATION

Thousands of Electronic Parts Available Today!

LEDS · CONNECTORS · RELAYS
SOLENOIDS · FANS · ENCLOSURES
MOTORS · WHEELS · MAGNETS
PC BOARDS · POWER SUPPLIES
SWITCHES · LIGHTS · BATTERIES
and many more items...

We have what you need for your next project.

www.allelectronics.com
Discount Prices · Fast Shipping



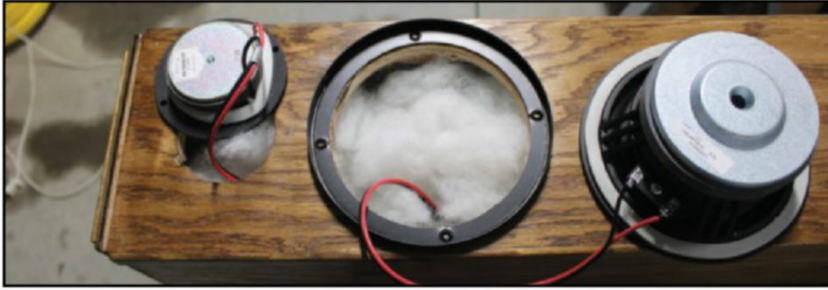


Photo 8: When installing the drivers, note the use of foam door insulation tape to get a good air seal.

lip (**Photo 8**). This will ensure a good air seal for the speakers. Once the drivers are installed, add the speaker terminal cup to the lower rear of the enclosure. Note that the speaker terminals noted in the Parts List are not the same as the ones I used in this project as they are no longer available from Parts Express. The only difference in the listed terminals is that they are round instead of the shape shown here. Mounting the handle above the speaker terminal is an option that makes carrying the speaker easier. With one hand on the handle and the other at the top front of the enclosure you can pick the tower up with no damage to it or yourself. I think it's a great addition to any tower speaker you might already own (**Photo 9**).

The speaker grilles that I used were left over from a previous project and are an optional addition to the project. You can opt for no grilles or use



Photo 9: The speaker terminal and optional lift handle are shown. The terminal shown here is not the one described in the Parts List, as it is no longer available. The listed part is the same but is round in shape.

those shown and detailed in the Parts List. There was no discernible difference in the sound with the grilles on or off.

Test the speakers at low volume to assure the wiring is correct, and then put on some John Phillip Sousa band music, relax, and enjoy the sound from your All American Towers.

Measuring Polyester Stuffing

Trying to determine the weight of the polyester stuffing for enclosures can be a daunting task, but by using a small postage scale and a cardboard tray it's possible to get a good estimate of the weight. The postage scale can be found at most office supply stores and the tray can be easily made from a cardboard box lid. Just center the empty tray on the

box, measure its weight, and subtract this number from the amount of polyester placed into the tray. Always tease out the polyester a bit before weighing it (**Photo A** and **Photo B**).



Photo A: A cardboard tray centered on a postage scale can be used to measure polyester stuffing in ounces.



Photo B: This angry looking storm cloud is actually 6 ounces of polyester stuffing that was used in the upper portion of the system.

MENLO SCIENTIFIC

Engineering
Success
since **1983**

Engineering Consulting for the Audio Industry

We Know

- Product Development

We Provide

- Design and Technical Support

We Deliver

- Liaison, Sourcing, and Guidance

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

MENLO
SCIENTIFIC

Richmond, California, USA

- tel 510.758.9014
- mike@menloscientific.com
- www.menloscientific.com



Old-Timer, Williamson, and Ultralinear Amplifiers

Families of Hollow-State Power Amplifiers

By
Richard Honeycutt



Photo 1: This Western Electric model 91A is a classic single-ended vacuum triode power amplifier.

The July and September Hollow-State Electronics articles examined the distortion performance of the three families of amps. This article recaps those findings and compares them to the last two families, the Williamson and ultralinear amps.

Hollow-state power amplifiers have gone through a series of stages since they first escaped the confines of home radio receivers. The first state stage focused on the early triode-based single-ended (SE) power amplifiers for cinema use (**Photo 1**). Then came triode-based push-pull (PP) power amplifiers using input transformer and transformer phase splitters. (The Western Electric Model 92 is such an amplifier, and visually resembles the Model 91A.) Next, we have push-pull power amplifiers using RC-coupled inputs and transformerless phase splitters. These mainly used pentodes or beam power tetrodes. And then, Williamson-type power amps with overall feedback including the output transformer in the feedback loop followed by ultralinear amplifiers. Now let's recap those findings from my previous two articles and compare them to the last two families of amplifiers.

Early Triode Single-Ended Amps

Probably the most highly regarded classic triode-based SE power amplifier is the original Western Electric Model 91. **Figure 1** shows its schematic diagram. The permanent-magnet

loudspeaker was not yet in existence when the Model 91 was designed (early 1940s) so the amp designers followed the usual practice of using the electrodynamic speaker's field coil as a power supply filter choke; hence, the connections labeled "monitor field" in the output connector. Other than that, the 91's circuit differs from simple modern SE vacuum triode power amps mainly in its use of an input transformer.

The Model 91 has overall negative feedback from the primary of the output transformer to the screen grid of V1. The September 2022 Hollow-State Electronics article reported results of modeling an approximate WE 91 circuit: the total harmonic distortion (THD) was a very low 3% at an output of 14.97W, and because of the negative feedback, the second harmonic distortion was slightly lower than the third.

Triode-Based PP Amps

When a cinema required a more powerful amplifier, it could upgrade to the Western Electric Model 92, whose schematic is shown in **Figure 2**.

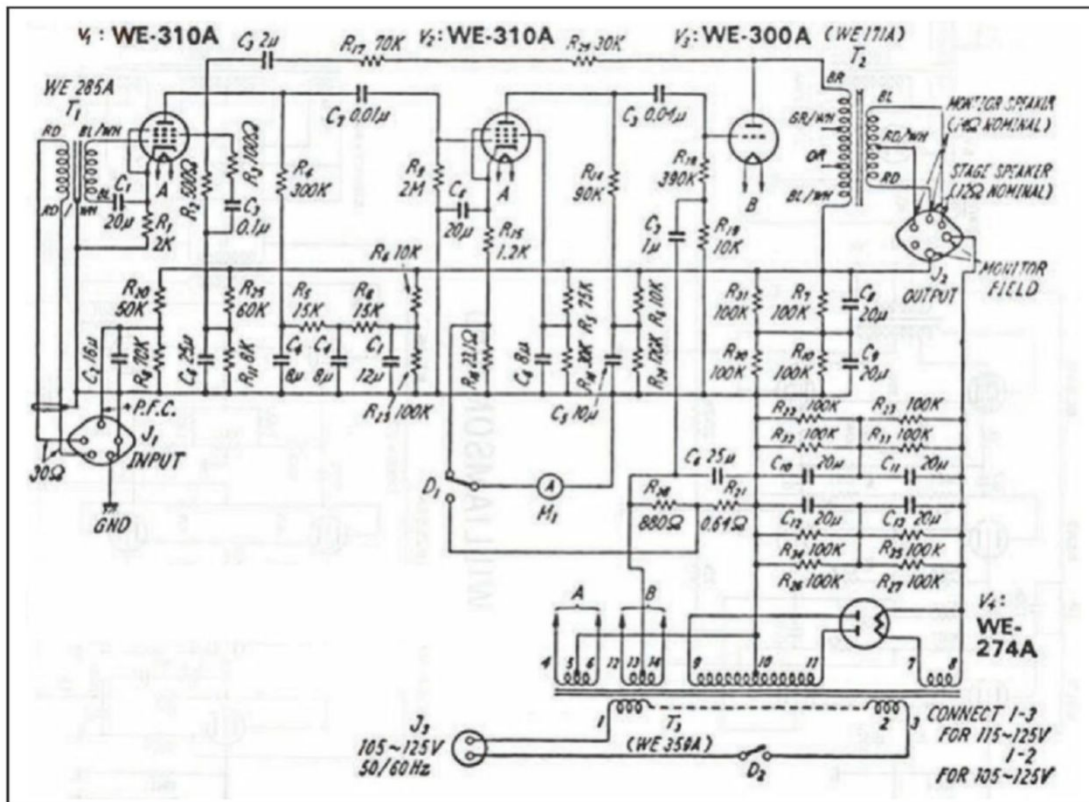


Figure 1: The schematic of the Western Electric Model 91B is straightforward.

This amp used only triodes, unlike the Model 91B, in which the first two tubes were Western Electric 310A pentodes. (The Model 91A used 6C6 triodes for V1 and V2.)

As detailed in my September 2022 article, creating an accurate computer model of the Western Electric 92 was fraught with difficulties due to unknown component characteristics. Thus we

can examine the performance of the push-pull (PP) triode amplifier family by looking at Pete Millett's "Fat Boy" design ("Fatboy" 300B Push-Pull Amp (www.pmillett.com)). **Figure 3** shows its schematic.

Millett's design demonstrates that a modern PP amp using 300B triodes can exhibit very low THD: only 0.2% at 10W output—2W below the onset of clipping. Since no negative feedback is used, THD

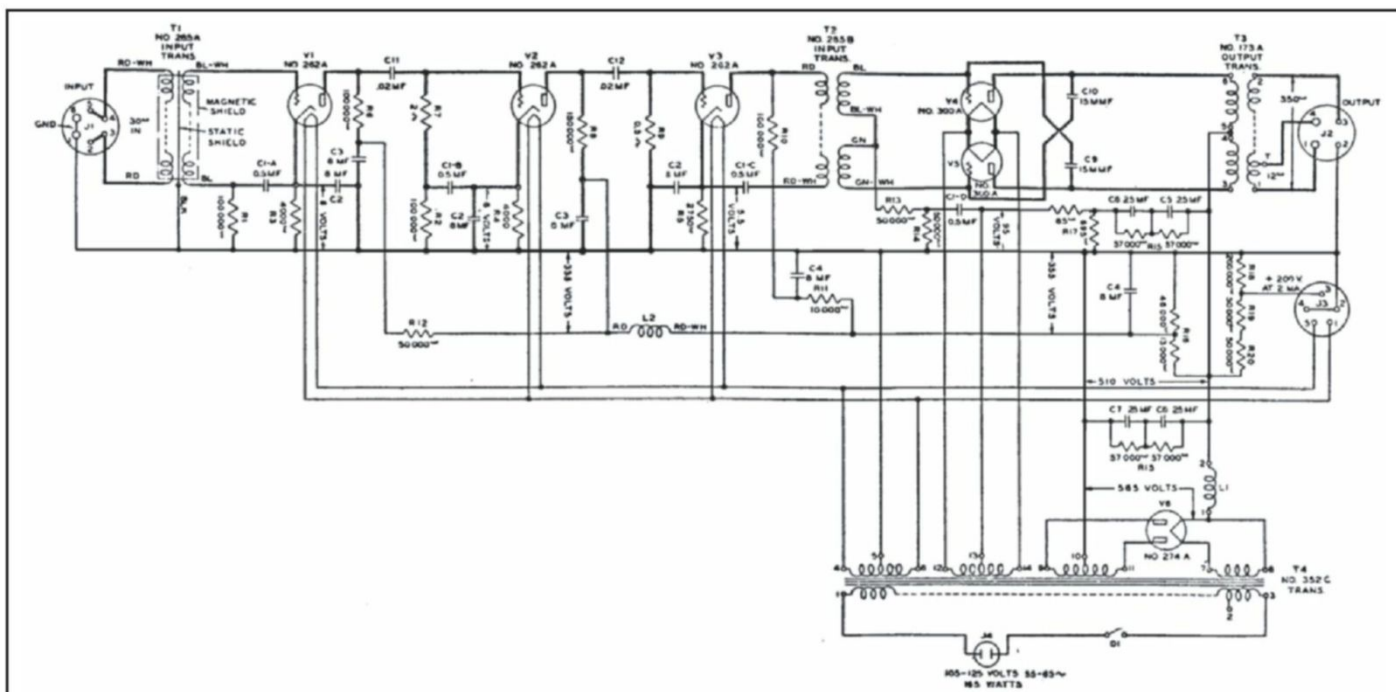
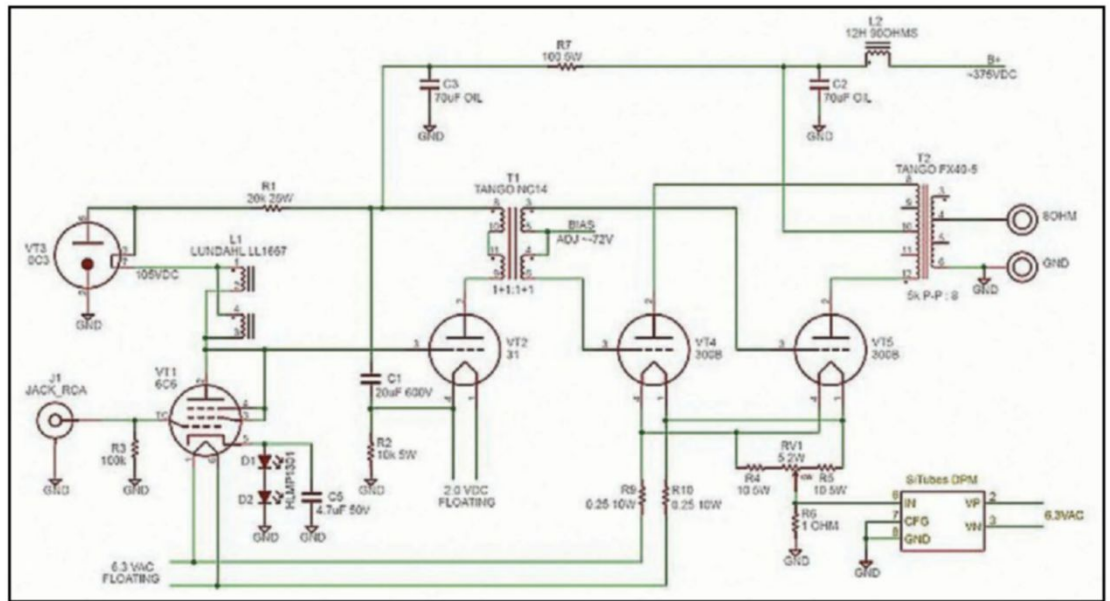


Figure 2: The Western Electric Model 92 used triodes throughout, as well as push-pull circuitry to reduce even-order harmonic distortion.

Figure 3: Pete Millett's "Fat Boy" is fine example of a modern push-pull triode amp using Western Electric 300B triodes.



only increases to 5% at 19W output. Like the Western Electric 91, the Fat Boy produces a bit more third harmonic than second.

There were significant harmonics up to the seventh, dominated by the third.

Pentode and Beam Power PP Amps Without Negative Feedback

In many applications, a demand for greater power output led to the use of power pentodes or beam power tetrodes in audio power amps. The early models often used little or no negative feedback (NFB). **Figure 4** shows an LTSpice model of a Tung Sol designed PP amplifier using 6CA7/EL34 output tubes.

The results of analyzing this model were presented in my September 2022 Hollow State column. The only NFB is the partially unbypassed cathode resistor of the 6267 pentode preamp tube.

The Tung Sol datasheet gives a maximum output power of 70W at a THD of 5%, but the computer model only showed 4.73% THD at 75W output.

Williamson and Derivatives

The PP configuration eliminated much even-order distortion, which (especially second-order) was the primary distortion component produced by triode power amplifiers. However, the overall THD of a pentode or beam power PP amp was higher than that of the best triode-based (lower-power) amplifiers.

David Theodore Nelson (DTN) Williamson introduced his power amplifier design in *Wireless World* in April and May of 1947 [1]. Williamson correctly concluded that the output transformer was a significant source of distortion, so his design incorporated negative feedback that included the output transformer within the feedback loop.

The Williamson design replaced the transformer phase splitter with cathodyne or "split-load"

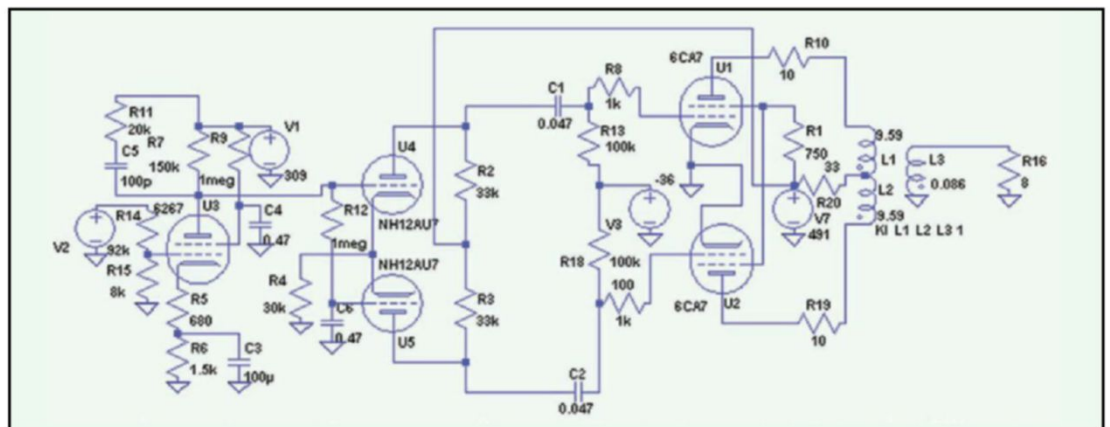


Figure 4: This push-pull pentode power amp has no negative feedback other than that provided by unbypassed cathode resistor R5.

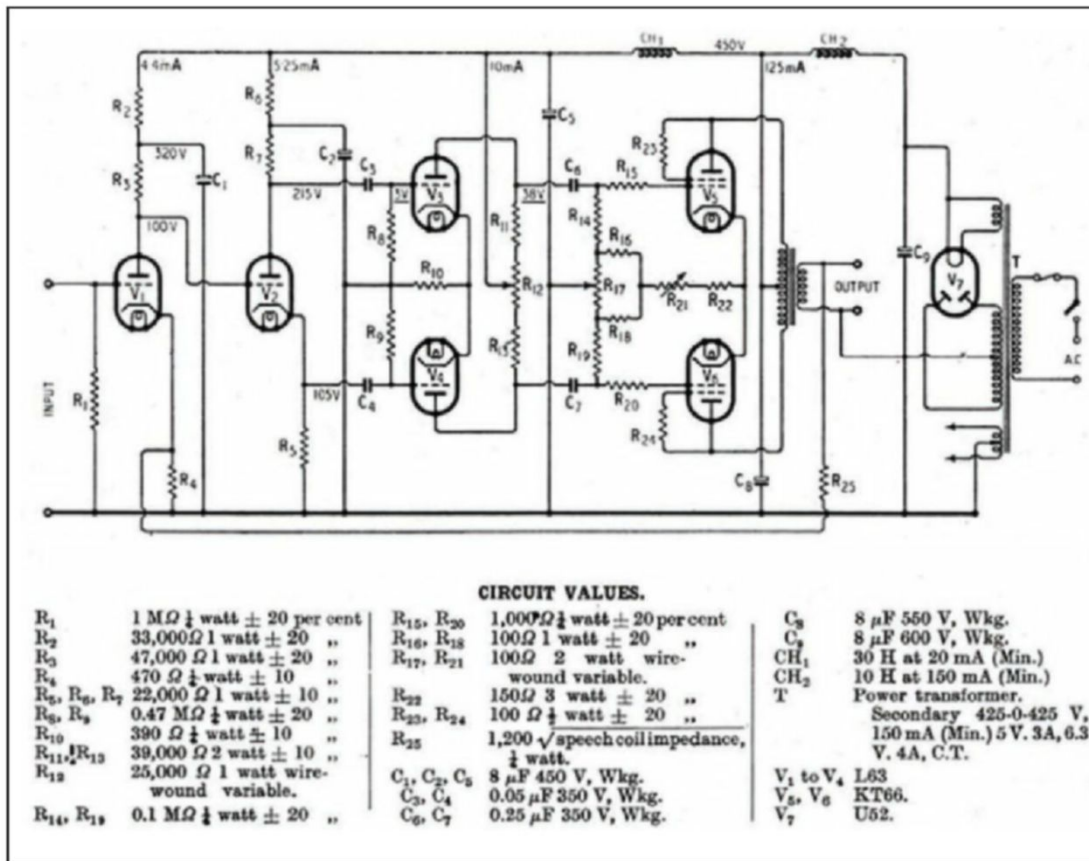


Figure 5: This is the original Williamson amplifier design.

ACOUSTICS
building your sound

4" SB12NRXF25-4/8

4 inch midwoofer with foam surround for maximum transparency and midrange resolution.



Find us on:

[f SB Acoustics](#)
[sbacoustics](#)
[sbacoustics_](#)
[SB Acoustics](#)
[sbacoustics.com](#)

type, providing a balanced (essentially Class-A transformerless push-pull) driver stage. Maintaining stability against regenerative feedback due to phase shifts in the output transformer made design of

the compensation network in Williamson's amplifier critical. Williamson's initial design was a 15W amplifier circuit and is shown in **Figure 5**. He used Osram KT66s for the output tubes, but stated that a

Figure 6: The Svetlana KT88 datasheet compares plate curves for a pentode (left-hand curves), triode (upper right-hand curve), and pentodes in an ultralinear connection (lower right-hand curve).

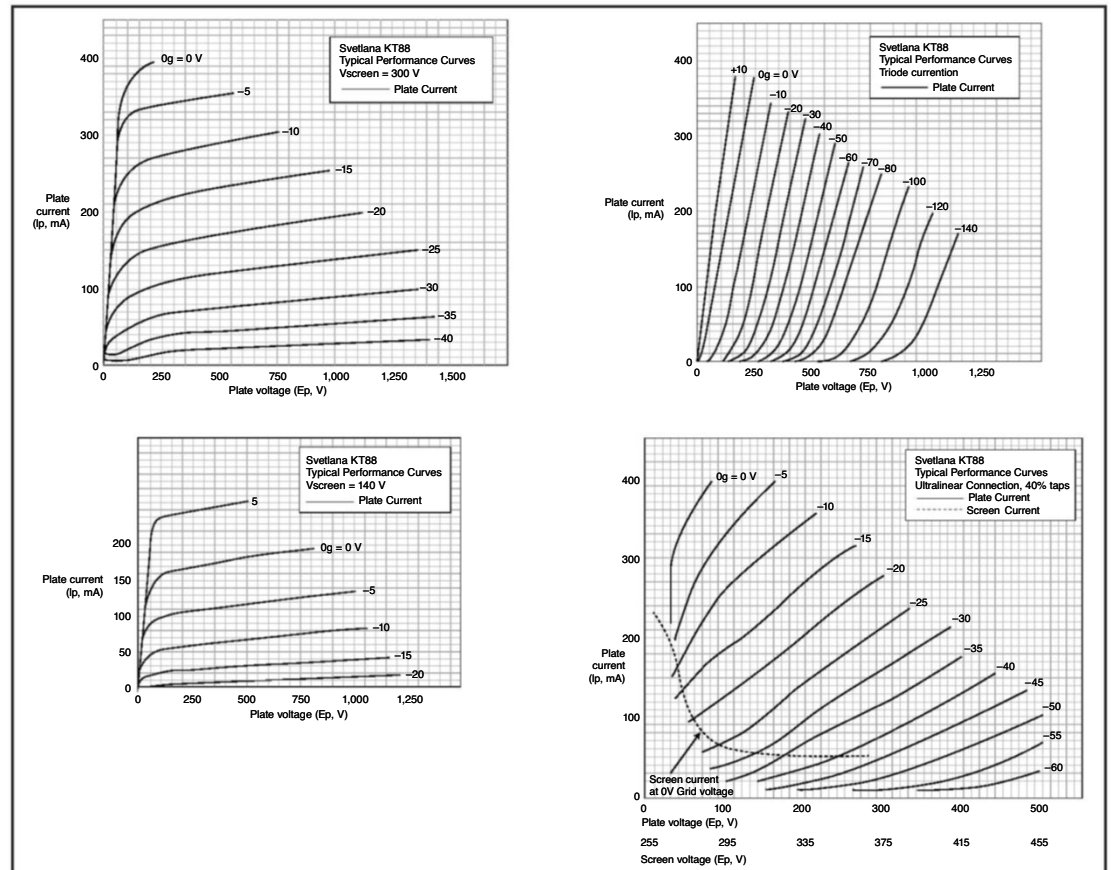
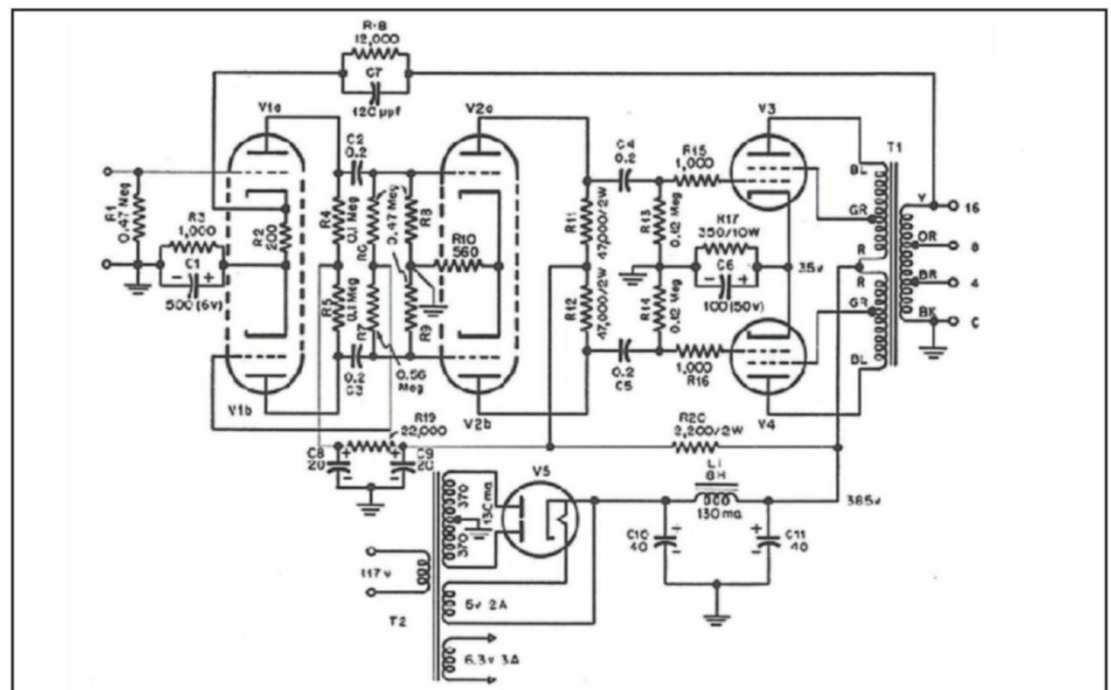


Figure 7: Note the screen-grid taps on the output transformer primary in this Hafler-Keroes ultralinear circuit.



6L6 or 6L6GT could be substituted if the total anode and current dissipation were reduced from 125mA to 110mA by adjusting potentiometer R21 appropriately. The designer measured THD at under 0.1% at full output, and reported that intermodulation distortion (IMD) was inaudible.

In 1949, Williamson published an updated design [2] including several small changes to the circuit, but these seem to have been made with an eye to stability against oscillation.

The Ultralinear Design

In 1937, Alan Blumlein patented a circuit technique known as “distributed loading.” In 1938, he was granted a US patent. However, it was not until 1951 that the name “Ultralinear” was applied to the use of distributed loading in an audio power amplifier.

In that same year, David Hafler and Herbert I. Keroes published an article [3] in which they claimed that improved linearity of audio power amplifiers could only be achieved by the use of a new type of tube. In their article, they listed the requirements for such a tube:

About the Author

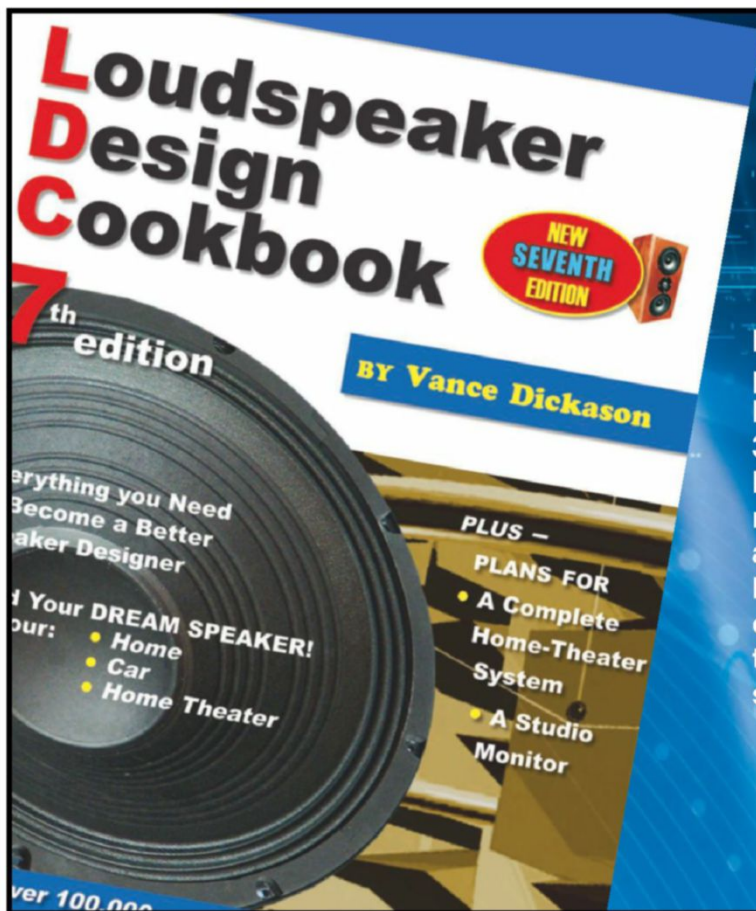
Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built 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.



1. Low internal impedance, such as is offered by the triode.
2. High power sensitivity of the tetrode so as to minimize drive problems.
3. Lower harmonic and intermodulation distortion than either triode or tetrode at both high and low levels of operation.
4. Sufficiently high efficiency to permit adequate output without undue bulk or cost.

Rather than actually introducing a new tube design, they proceeded to provide a circuit design that made a PP pair of pentodes or beam power tetrodes act like a different type of tube.

Figure 6 shows the plate characteristics for a



The “Must Have” reference for loudspeaker engineering professionals.

Back and better than ever, this 7th edition provides everything you need to become a better speaker designer. If you still have a 3rd, 4th, 5th or even the 6th edition of the Loudspeaker Design Cookbook, you are missing out on a tremendous amount of new and important information! **Now including: Klippel analysis of drivers, a chapter on loudspeaker voicing, advice on testing and crossover changes, and so much more!**

A \$99 value!
Yours today for just \$39.95.

www.cc-webshop.com

KT88 configured as a pentode, a triode, and an “ultralinear device.”

The ultralinear connection involves feeding the screen grids of the output PP pair from taps on the output transformer primary, as shown in the Hafler-Keroes circuit described in **Figure 7**. The location of the tap (% of the secondary winding between the plate and the screen) controls the ratio of screen load impedance to plate load impedance, which affects maximum power output, distortion at low

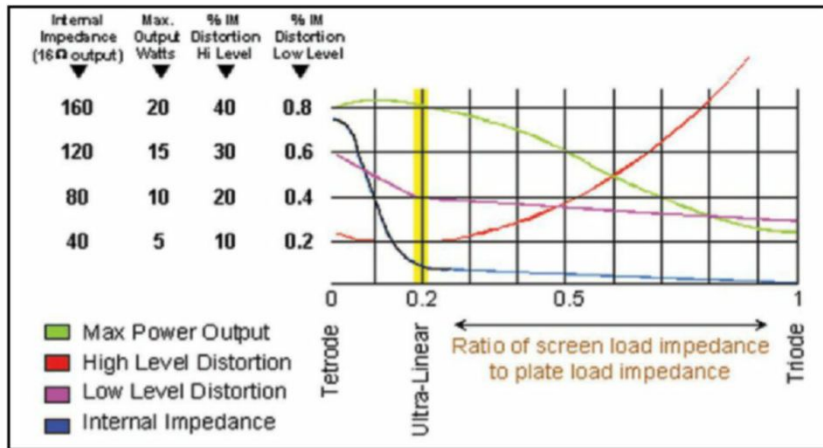


Figure 8: The location of the screen taps in an ultralinear amplifier affect four performance characteristics.

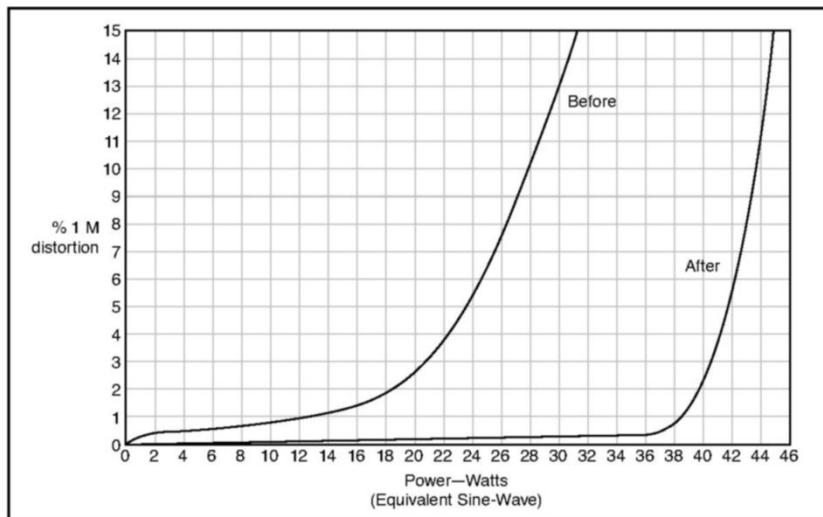


Figure 9: Wright published this comparison of IMD in an unmodified “Williamson Ultralinear” amp and the amp with his modifications incorporated.

signal levels, distortion at high levels, and internal (Thevenin) impedance, as graphed in **Figure 8**.

In 1961 *Electronics World* published an article by Talbot M. Wright [4] suggesting improvements to the Williamson amplifier to decrease IMD. Wright suggested changing the values of three resistors to optimize the Q point of the first voltage amplifier, the cathode follower, and the phase splitter. He stated that these changes were necessary because of differences between the operating characteristics of European versus American 6SN7s.

It is important to note that Wright was discussing Williamson amplifiers with an ultralinear circuit, having a rated power output of about 50W: about 3 1/3 times that of Williamson’s original. Recalling that Williamson stated that IMD in his original 15W amplifier was inaudible (no measurements were given), it is surprising that Wright states that in the 50W version IMD was about 2% at 8W. While not giving actual measurements, Wright stated that with his modifications, no “Williamson/Ultralinear power amplifier should produce over 1% distortion (**Figure 9**). Further discussion of the “Wright-Williamson” amplifier appeared in my August 2017 Hollow-State Electronics article.

The Heathkit W5M, Fisher Model 70, and Dynaco MKII hollow-state power amps are classic examples of highly regarded power amps that used the “Williamson/Ultralinear” configuration. The Dynaco MKII specified IMD < 0.5% for an output of 35W, and < 1% at the full rated 50W continuous power.

In a Family by Itself

The other superb classic hollow-state amplifier is the McIntosh MC75. Using a unique patented circuit design (described in my April 2015 Hollow-State Electronics article), the MC75 had THD < 0.5% for the frequency range of 20Hz to 20,000 Hz at the full rated power of 75W, and IMD < 0.5% at any instantaneous power below 150W. Because of the patented design features and the special output transformer required, there have been few attempts by DYIers to build their own MC75s. Although the mono MC75 is no longer made, the stereo MC275 model is still being manufactured.

References

- [1] D.T.N. Williamson, “Design of a High-Quality Amplifier,” *Wireless World* 53.4 (April 1947), p. 118; and 53.5 (May 1947), p. 161.
- [2] D.T.N. Williamson, “High-Quality Amplifier—New Version,” *Wireless World* 55.8 (August 1949), p. 282; 55.10 (October 1949), p. 365; and 55.11 (November 1949), p. 423.
- [3] D. Hafler and H. I. Keroes, “An Ultra-Linear Amplifier,” *Audio Engineering*, (November 1951), pp. 15–17.
- [4] T. M. Wright, “Improving the Williamson Amplifier,” *Electronics World* (June 1961) pp. 104-106.



ELECTRONIC TRANSFORMERS

- Original Equipment
- Replacement
- Amateur

TRIAD MAGNETICS, INCORPORATED

460 HARLEY KNOX BLVD., PERRIS, CA 92571

www.TriadMagnetics.com



Made For Any Venue

DAYTON AUDIO ODEUM Series

Professional High-End Drivers from Dayton Audio

ode· um | \ ō-'dē-əm:

An ancient Greek and Roman concert hall used for music and poetry.

With decades of driver design and engineering, placing emphasis on technical excellence, product reliability, innovation, and value the Dayton Audio ODEUM series takes the stage with a new line of professional drivers made for any venue. Engineered for the ultimate balance in power, control, and affordability, the versatility of the ODEUM series makes these drivers a great solution to upgrade, replace, or develop a new pro sound system.

Featuring

- Two 18" 1200 watt tour grade subwoofers with enough power handling to provide thundering low end that fills your event space
- Five midbass drivers with powerful neo motors deliver low distortion and high SPL perfect for monitors and line arrays
- Two 80 watt 1" Ferrite exit compression drivers produce a smooth response with bolt on and screw on configurations and your choice of titanium and polyimide diaphragms



LEARN MORE

Scan here for more information about the Dayton Audio ODEUM Series or visit daytonaudio.com