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Distortion Below -140 dBc

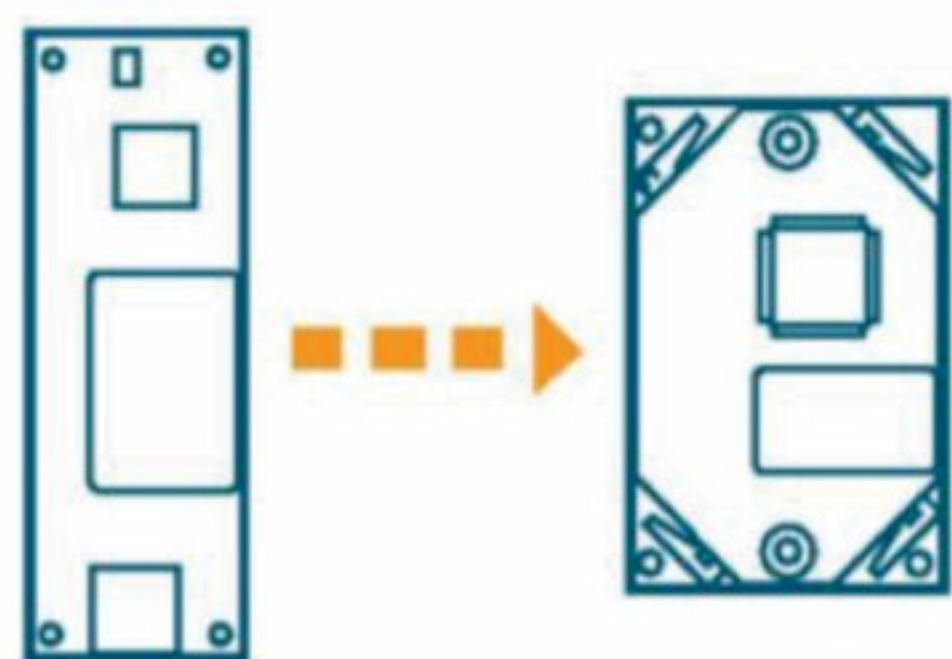
By Michael Steffes



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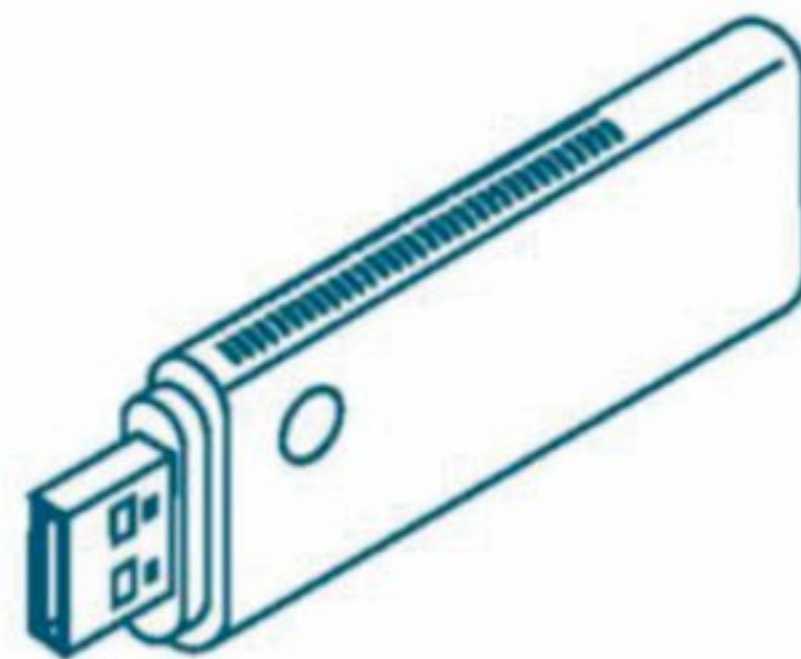


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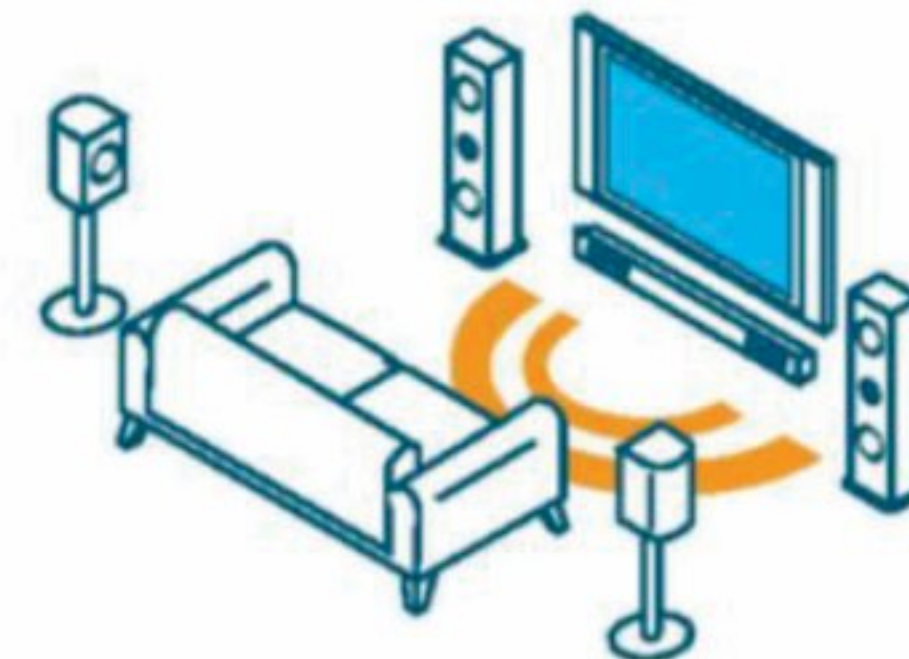
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The Importance of Speakers and Sound



In this edition of *audioXpress* we have decided to focus on speakers and transducers. This might seem odd to some readers who also follow our sister publication *Voice Coil*, which is often referred to as the voice of the loudspeaker industry. But the reality is, as much as we follow loudspeakers, there’s always more to say, so many references that should be highlighted and product stories to share—and this is particularly important for those who don’t necessarily work on speaker design but have drivers as part of the range of components needed for audio design. We would like this Market Update on Speakers and Transducers to be the first of many.

And talking about speakers, at the ALMA International Symposium & Expo (AISE) 2019, held in June, the Association of Loudspeaker Manufacturing & Acoustics (ALMA) International announced its official name has been changed to Audio & Loudspeaker Technologies International (ALTI): “Built for and by professionals in the Audio and Loudspeaker Technologies Industry to learn, to teach, to grow and to get business done!” The change in name and the enhancement of ALTI’s mission to embrace the entire audio signal path has been met with universal approval. This acknowledgement of the interdependence of every segment of the audio signal path is vital to the continued growth and advancement of the electro-acoustics industry. ALTI plans to offer more in-depth coverage and exploration of DSP, signal processing, and new technologies that will impact the loudspeaker industry.

With the change, came a transition of the Symposium & Expo activities—now rebranded to ALTI-EXPO—taking place every year before the InfoComm show and promoted in partnership with the Audiovisual and Integrated Experience Association (AVIXA), producer of InfoComm trade shows around the world, co-owner of Integrated Systems Europe (ISE), and the international trade association representing the audiovisual industry.

It is now more important than ever that all loudspeaker manufacturers and professionals join ALTI and support the association’s efforts. In 2019, the association’s activities took place June 9–10 in Orlando, FL, just prior to InfoComm 2019. Once again, it was the strongest event yet for anyone in the loudspeaker and electroacoustic development in general. Nowhere else were there so many outstanding presentations of interest for the loudspeaker community, and no other event allows such close and direct contact with the key vendors and suppliers for the industry.

Among the new initiatives in 2019, and those now being planned for 2020, is the Speaker Builders Workshop, which was certainly a highlight. All participants enjoyed the full-day, hands-on format that immersed them in the entire process of concept through build and testing of a loudspeaker product. Now, ALTI plans for these workshops to become a multi-part series, and ALTI-EXPO 2020 is going all in on developing the concept begun in 2019. The workshop will expand, with participants learning how to build a driver for a specific application as well as design and test an enclosure for the driver. On the second day, participants will have a half-day session that adds an amplifier with Bluetooth to the loudspeaker and enclosure built on Day 1. Once again, the process will include time to design, build, and test the finished product. And, as always, qualified full-time students never pay for membership in ALTI, and students can attend ALTI-EXPO at no charge.

ALTI-EXPO 2020 will take place on June 14–15, 2020, at the South Point Hotel and Casino in Las Vegas, NV. InfoComm will take place on June 17–19, 2020 at the Las Vegas Convention Center. ALTI has also issued its “Call for Content” for ALTI-EXPO 2020. More information available here: <https://almaint.org/altiexpo-2020-content>.

Also, don’t forget, 2020 will be the International Year of Sound (#IYS2020). This is an initiative of the International Commission for Acoustics (ICA) and is an outcome of UNESCO Resolution 39 C/59 2017 “The Importance of Sound in Today’s World: Promoting Best Practices.” We will discuss more about this important initiative next month. For more information, visit: www.sound2020.org

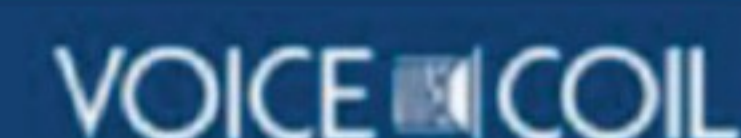
J. Martins
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COLUMNISTS

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

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

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

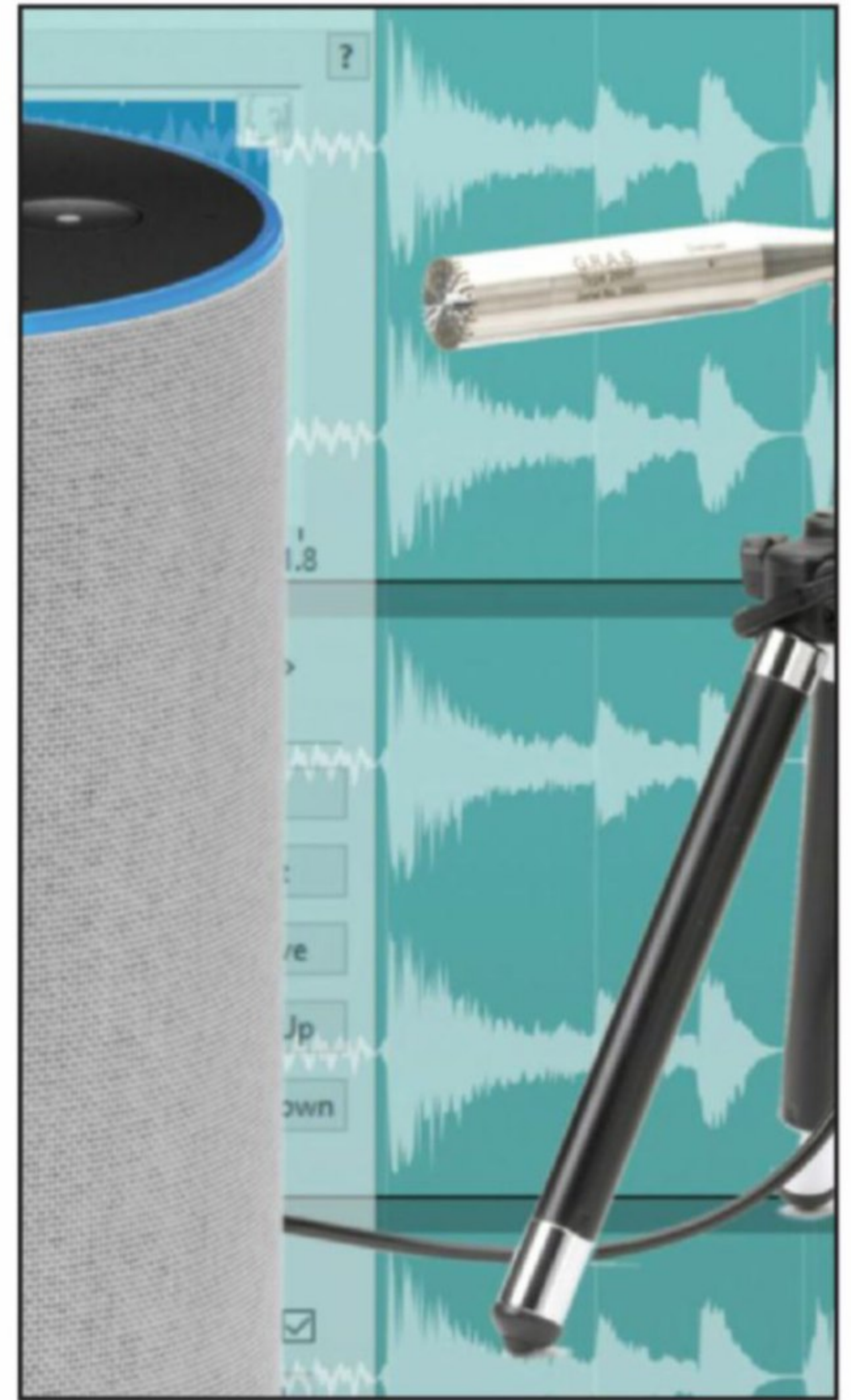
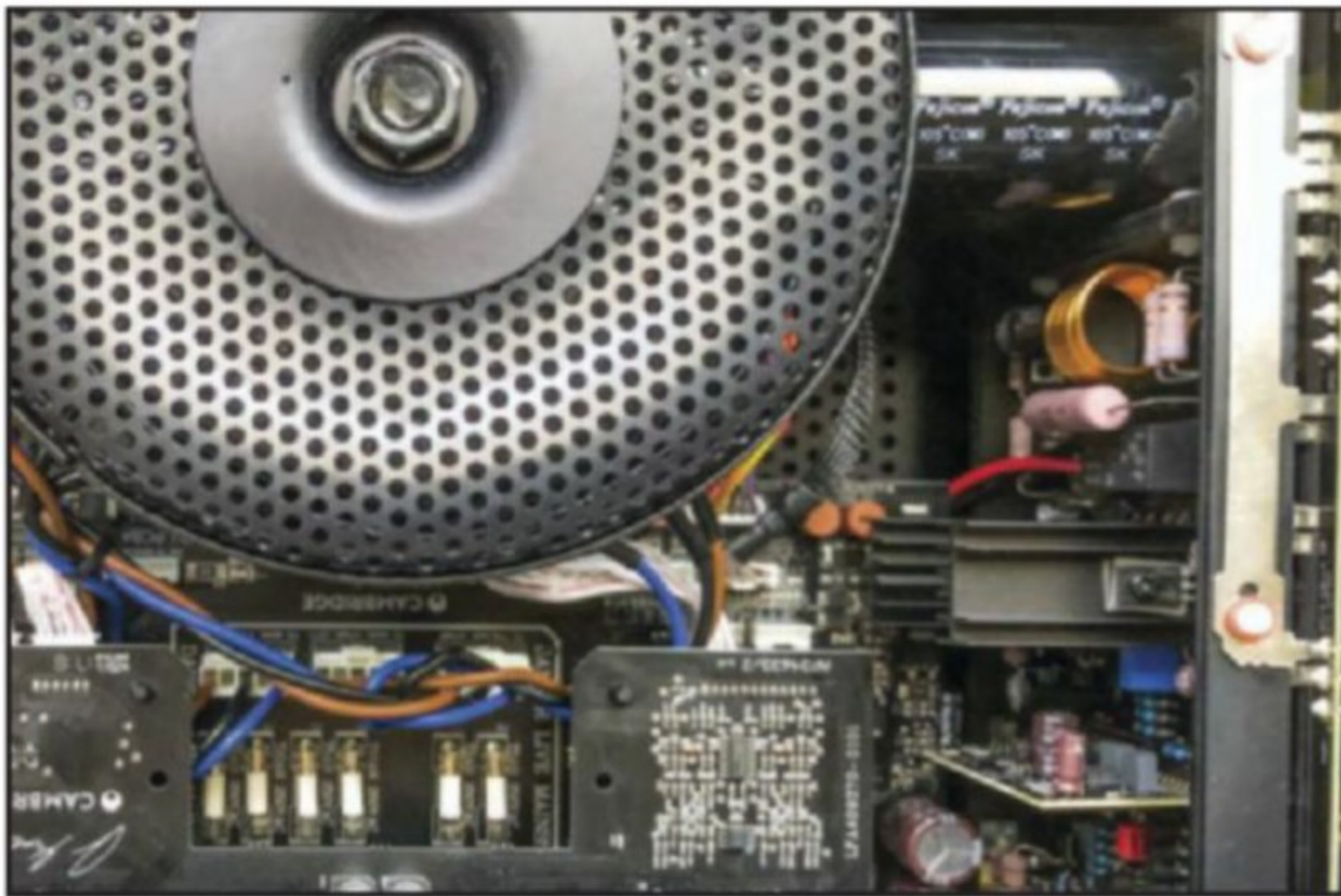
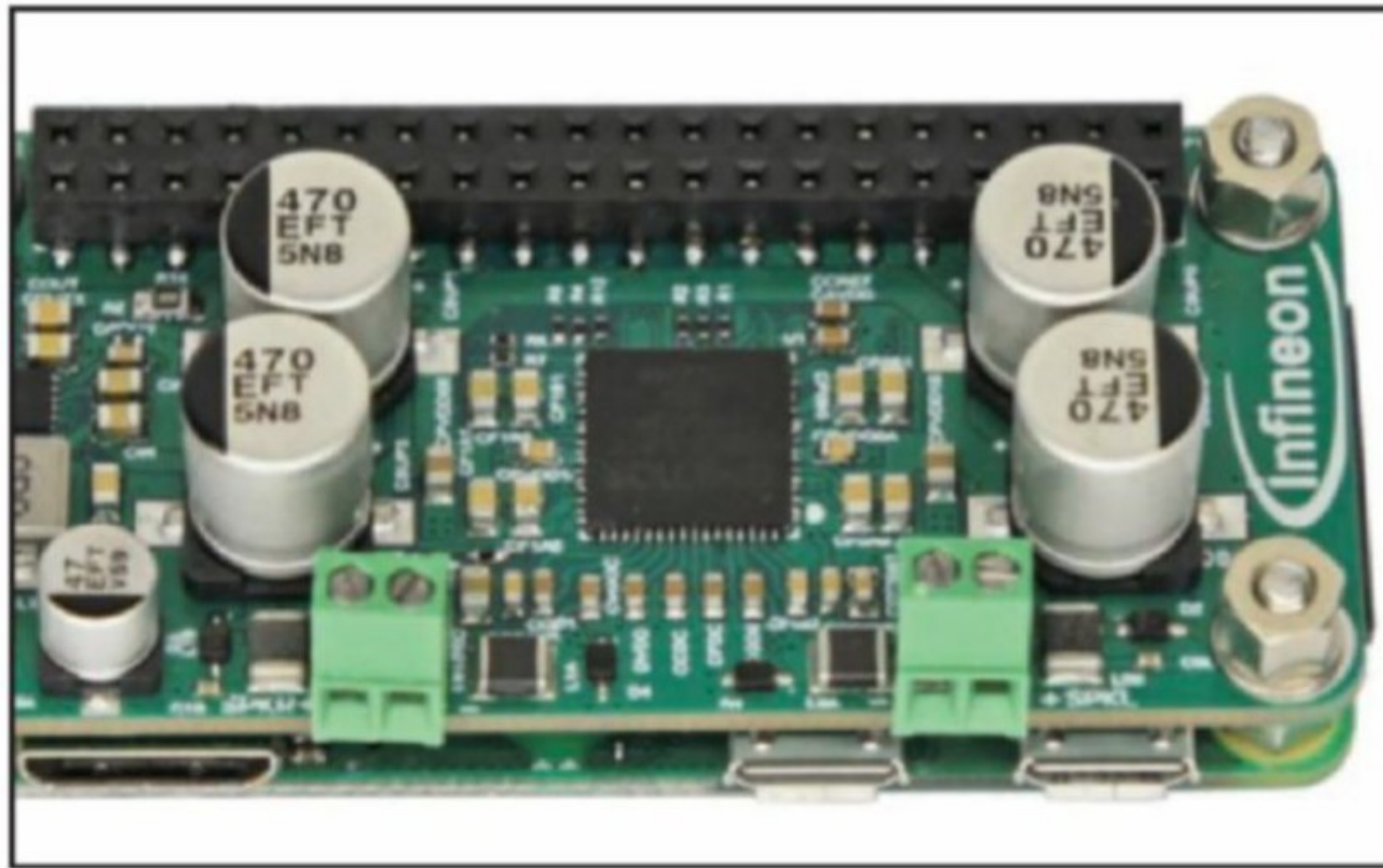
Michael Steffes has worked for more than three decades in high-speed signal path IC development, spanning IC design roles for the BurrBrown high-speed amplifier products, and product launch/application manager roles through Texas Instruments/Intersil. He continues to focus on high-performance ADC interface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

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By Oliver A. Masciarotte

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By Stuart Yaniger

For a few weeks, two *audioXpress* reviewers, Oliver Masciarotte and Stuart Yaniger, tested Cambridge Audio's Edge series power amplifier, the Edge W. *audioXpress* submitted this design, which claimed to be the culmination of the brand's audio engineering expertise, to extensive listening and measurement sessions.

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By Steve F. Temme

This article discusses tools and techniques that are available to accurately measure the audio performance of voice-controlled and connected devices under the many various real-world conditions they may be used.

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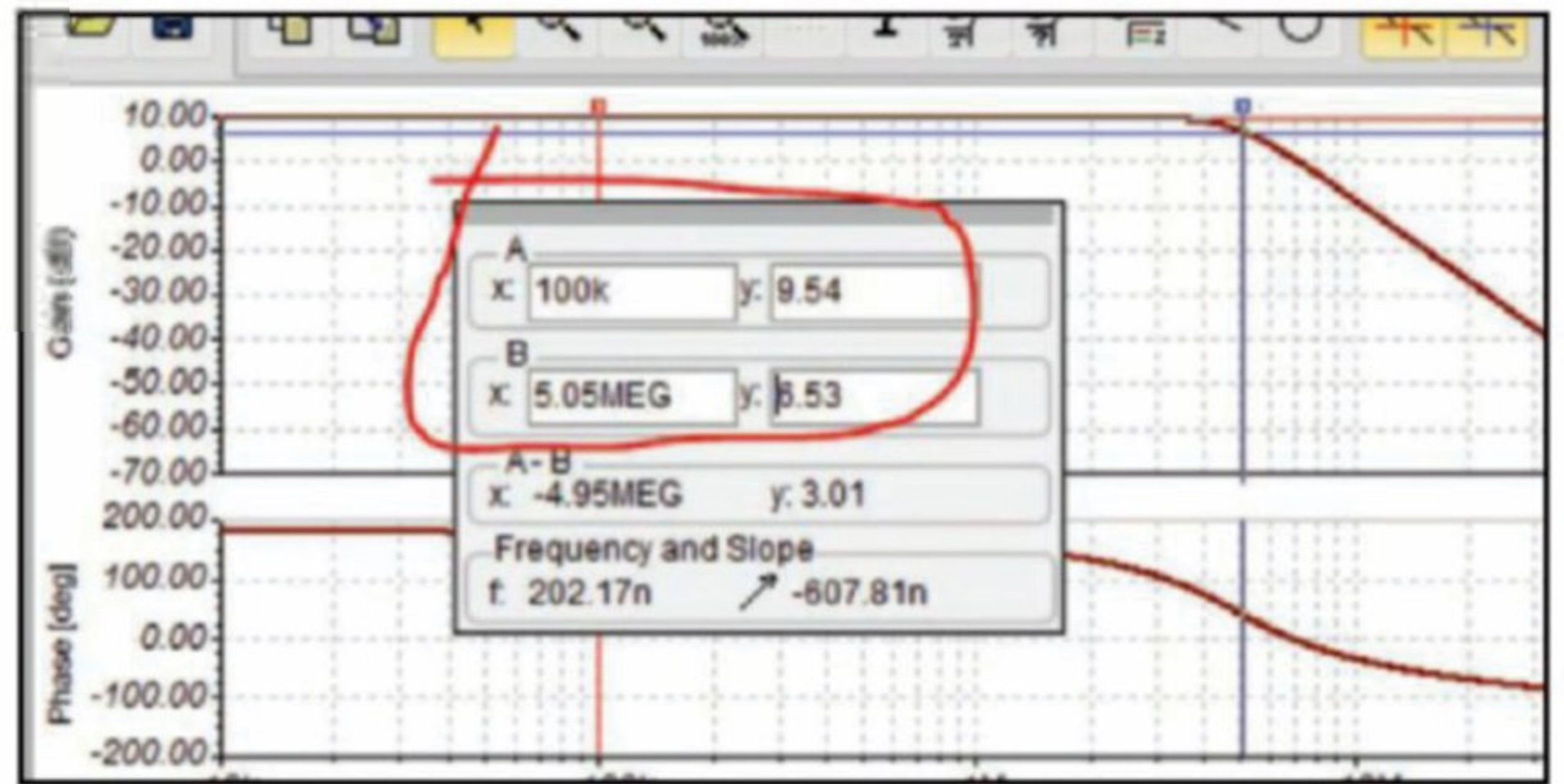
By Jens Tybo Jensen

Learn more about this new affordable high-end 2 x 40 W Class D audio amplifier in a Hardware Attached on Top (HAT) board to connect onto a Raspberry Pi Zero platform, creating a solution ideally suited for development and DIY audio projects.

59 An Essential Piece of Equipment for the Audio Enthusiast Heavy-Duty Dummy Load

By Bruce Brown

If you are a regular reader of *audioXpress*, you might know that Bruce Brown restores vintage tube equipment, as well as select transistor stuff. There are a number of essential tools needed for evaluating and testing audio equipment. These tools include an oscilloscope, a good voltmeter, an audio generator (sine, square, and triangle waveforms) and last but not far from least a serious dummy load. This article will explain how to build such an essential piece of equipment.



● Market Update—Speakers and Transducers

18 Big Promises in Large and Small Parameters

By J. Martins and Mike Klasco

In this Market Update, we address new trends in drivers and transducers, from new diaphragm materials for loudspeakers to woofers, tweeters, and compression drivers to microspeakers and drivers for headphones. And we share some vital market data that shows how speaker technology and components continue to be at the core of a thriving industry.

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2019 COMSOL Conference Boston

One Workflow to Rule Them All

In October 2019, David Logvin attended the 2019 COMSOL Conference in Boston, MA, and explored the latest tools in acoustics simulation at this three-day event. He says the adoption of COMSOL Multiphysics simulation software throughout the audio world should come as no surprise.

By

David Logvin

(United States)

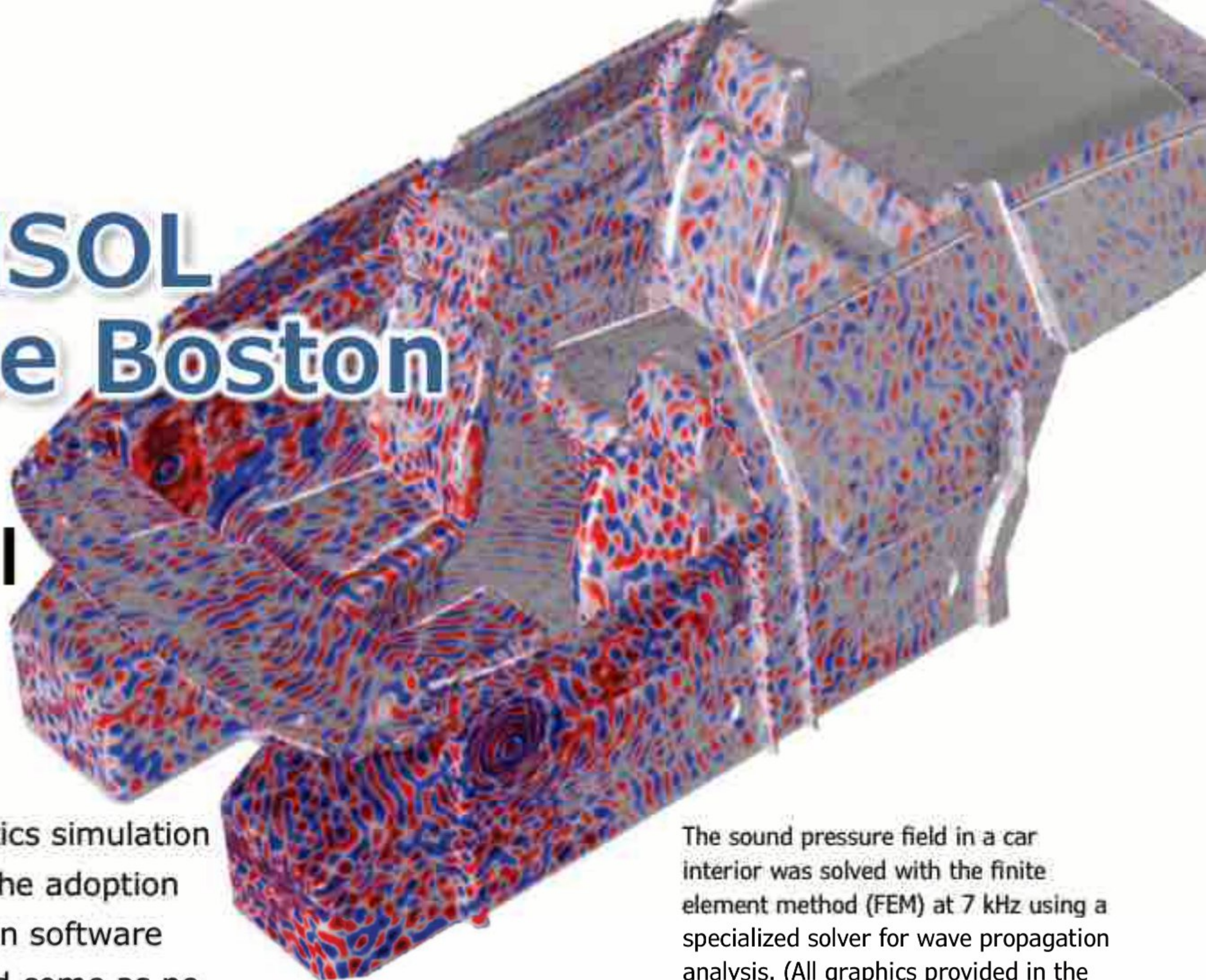
In 1998, the first version of COMSOL Multiphysics simulation software was released. Fast forward 20 years and this powerful tool has grown exponentially, thanks to a worldwide community of users and formal conference gatherings in Japan, China, Taiwan, South Korea, India, the United Kingdom, and the United States. The COMSOL US conference in Boston, MA, has the added advantage

of being within driving distance to the COMSOL corporate headquarters in Burlington, MA, so it is particularly well-attended by staff (see **Photo 1**).

Structurally, this meeting included user paper presentations from industry and academia, poster sessions, panel discussions, keynote sessions, and an awards ceremony for papers and posters, as well as technical mini-courses in subjects as varied as computational fluid dynamics, heat transfer, electromagnetics, structural mechanics, and acoustics (see **Photo 2 and Photo 3**). These tools have become particularly well-suited for those researchers working on loudspeakers and microelectromechanical systems (MEMS) microphone technology, and this is what motivated me to attend the event and learn more about COMSOL with a deep dive into multi-dimensional physics modeling.

Presentations

On the user presentation side, Samsung America shared an interesting study on “Optimal Design of Bass Reflex Loudspeaker Ports.” This rigorous examination investigated the classic question of how much flare is optimal both by simulation and measurement validation. It is of interest to anyone who has ever worked on vented loudspeaker design and wants to understand more about flow separation



The sound pressure field in a car interior was solved with the finite element method (FEM) at 7 kHz using a specialized solver for wave propagation analysis. (All graphics provided in the article are used with permission from COMSOL, Inc.)



Photo 1: On Day 1, the 2019 COMSOL Conference in Boston kicked off with a sneak peek at the latest version of the COMSOL software during the keynote. Left to right: Bjorn Sjodin, Daniel Smith, Svante Littmarck, Kristian E. Jensen, and Mats Nigam of COMSOL, discuss upcoming software features in the 5.5 Release.

and vortex shedding at high air velocities. Other notable acoustics presentations included “Estimating Parasitic Capacitances in MEMS Microphones Using Finite Element Modeling” from Knowles Electronics, as well as “Thermo-Acoustic Simulation of a Piezoelectric Transducer for Interstitial Thermal Ablation with MRTI-Based Validation” from Katie Gandomi, Paulo Carvalho, Zhanyue Zhao, Christopher Nycz, and Gregory Fischer, (Worcester Polytechnic Institute), with Everette Burdette (Acoustic Med Systems).

The last paper was a particularly interesting study using a lead zirconate titanate (PZT) ceramic transducer and was distinguished with a COMSOL Conference award for both best paper as well as best poster (see **Photo 4**). **Photo 5** shows COMSOL’s founder and CEO Svante Littmarck (left) with all of the 2019 Best Paper and Best Poster award winners.

At its core, COMSOL software provides a universal solution for any sound or vibrations study using a Multiphysics approach. Using the Model

Builder, audio transducer simulation can be built with electrical, structural, and acoustics modules.

Methodology is likewise flexible and scalable and can use Finite Element Methods (FEM), Boundary Element Methods (BEM), Classical Ray tracing methods, or discontinuity FEM for large and transient studies.



Photo 2: On Day 1 there was a discussion on best practices during the “Advances of Acoustics Simulations for Product Development” panel. From left to right: Nagi Elabbasi, (Veryst Engineering); Jin Liu (Carrier); Andri Bezzola (Samsung Research America); Sarah Heile (Bose); and Mads Jensen (COMSOL).



Photo 3a and Photo 3b: More than 60 users in industry and academia participated in the Poster Session on Day 2 and shared their research.



Photo 4: On Day 3, COMSOL’s founder and CEO Svante Littmarck awards Katie Gandomi of Worcester Polytechnic Institute with both a Best Paper award and Best Poster by Popular Vote award.



Photo 5: COMSOL’s Svante Littmarck (left) is pictured with all of the 2019 Best Paper and Best Poster award winners.

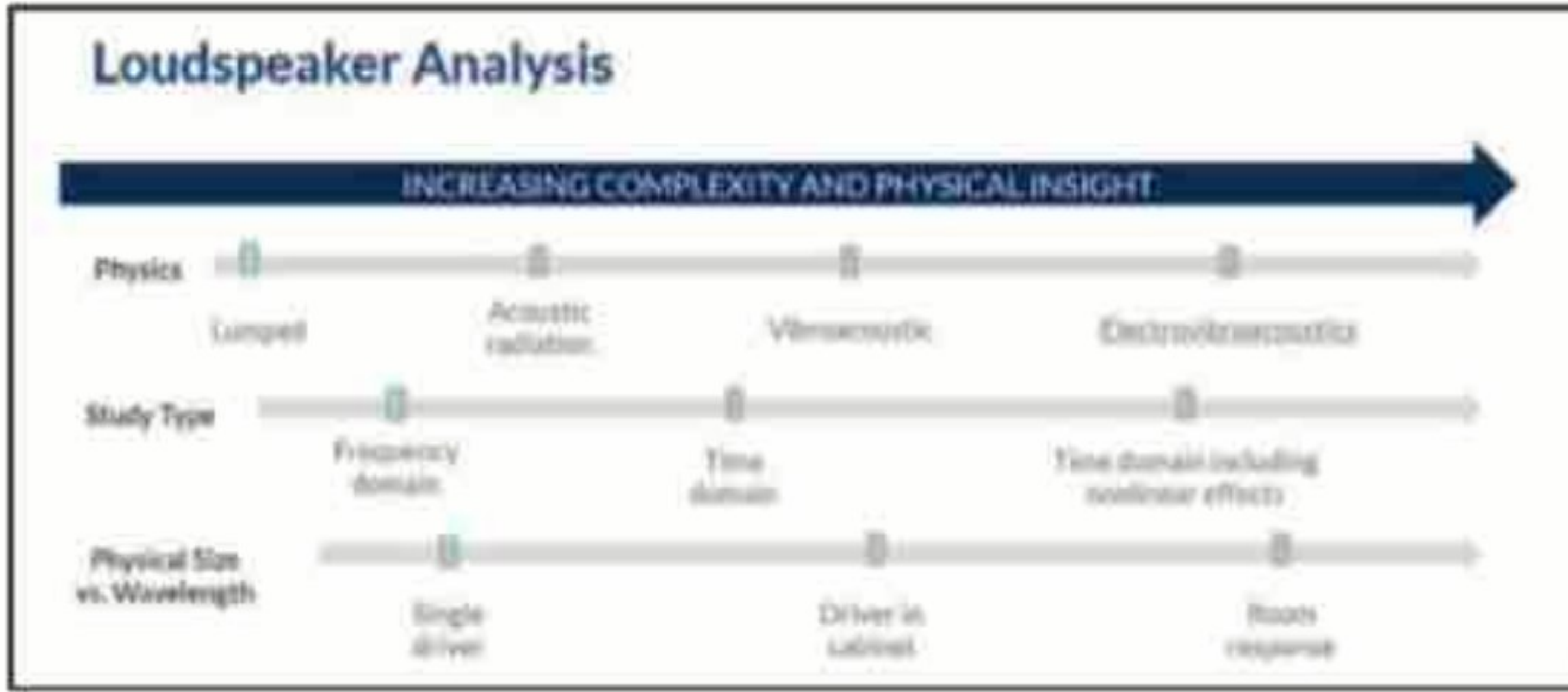


Figure 1: The Loudspeaker Analysis information highlights the power of COMSOL in analyzing complex loudspeaker problems.

Supporting the Acoustics World

I attended several intriguing acoustics mini courses during my conference attendance: "Modeling Speakers, Microphones, and Other Transducers," "Modeling Acoustic Propagation in Fluids and Solids," and "Acoustics & Vibrations Modeled with the Discontinuous Galerkin Time Explicit Interfaces." All three sessions were presented by Mads Per Jensen, the product manager for the Acoustics division. He is a highly experienced expert in the field, and he is based out of the company's Denmark office.

As part of conference registration, all attendees received a 30-day trial of the COMSOL Multiphysics 5.5 Release. In addition to changes to the GUI to streamline physics selection, there are also enhancements to existing solvers. Other notable improvements include a new Elastic Waves, Time Explicit physics interface, advanced modeling techniques for the analysis of ports in microacoustic applications using both transfer and scattering

matrices, as well as pipe acoustics multiphysics coupling.

From the acoustician's point of view, what is particularly notable in this release is the breadth of scale in the new 5.5 features. This provides improved capability when modeling the smallest microspeakers and MEMS microphones (port boundary conditions), while the Elastic Waves updates provide advantages when modeling the acoustically large domains for concert venues or car cabins.

This is quite analogous to the power that COMSOL unlocks for loudspeaker analysis. Available tools have a holistic approach, allowing a complete design to be modeled, from transducer to full audio system to large acoustical space for far field evaluation. Electroacoustic designs can be analyzed using lumped element, fully-coupled, or lumped element and fully-coupled methods (see **Figure 1**).

This release also includes the capability for using anisotropic material properties in the Pressure Acoustics module, which is particularly useful when working with the latest in metamaterials. A demo copy of the 5.5 Release additionally provides trials for all COMSOL modules (Electromagnetics, Fluid Flow & Heat Transfer, Structural Mechanics & Acoustics, Chemical Engineering, Multipurpose, and LiveLink products for integration with other software such as MATLAB, Excel, SolidWorks, etc.), which are detailed in **Figure 2** and **Figure 3**.

Development Tools

In the world of loudspeaker development,

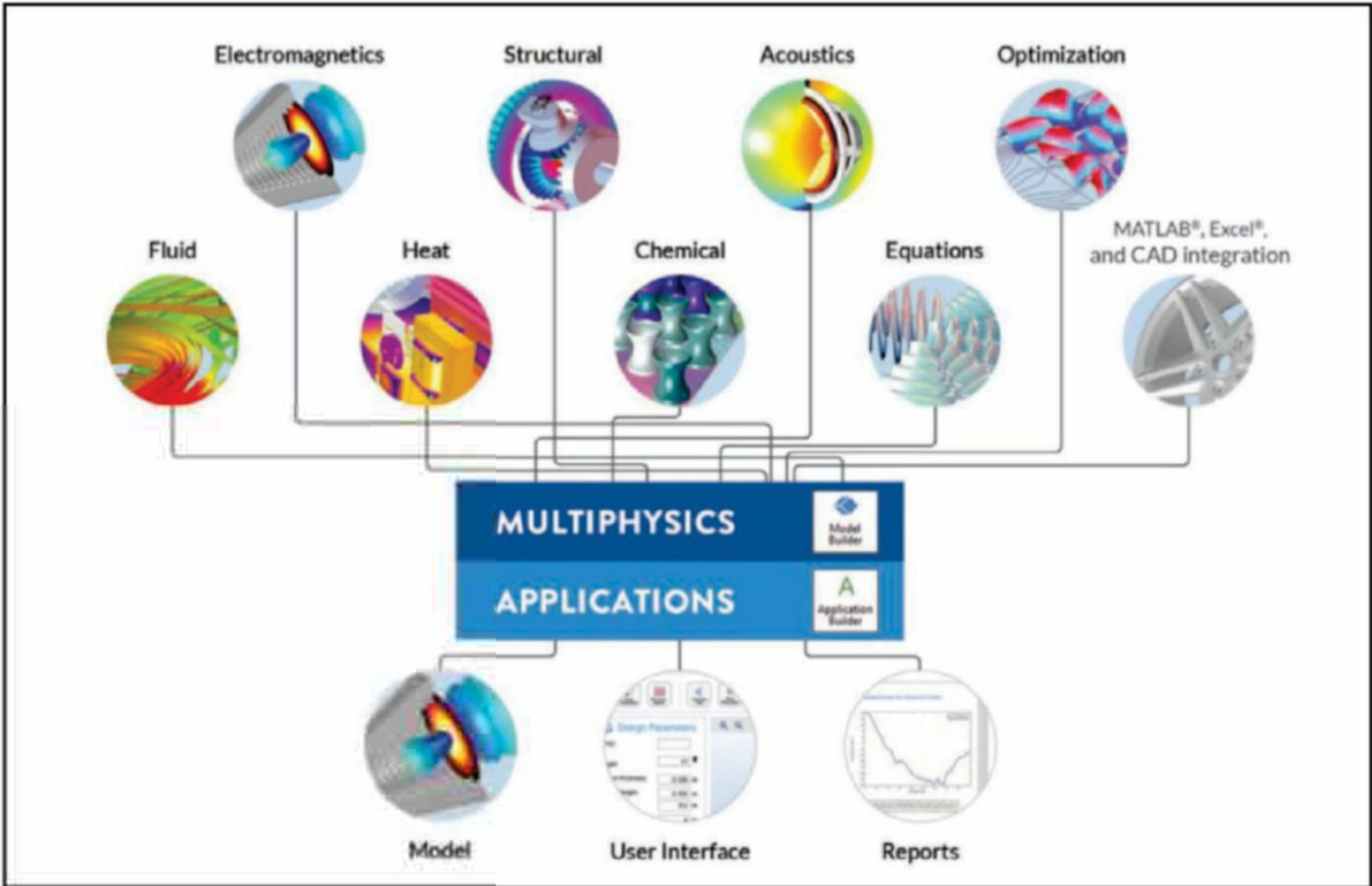


Figure 2: This chart provides a summary list of the available modules.



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there have been many significant events that truly revolutionized the engineer's toolbox and design process, such as Leo Beranek's eponymous reference *Acoustics* (1954) which clearly outlined lumped element electroacoustic models, or Neville Thiele and Richard Small's Audio Engineering Society (AES)

papers in 1971 on loudspeaker box alignments. When COMSOL was first released a little more than 20 years ago, it wasn't targeted toward researchers working specifically in the areas of audio and acoustics (interestingly, the first industry to widely adopt and leverage this software was chemical

Figure 3: This chart provides a complete listing of the available modules in the 5.5 Release.

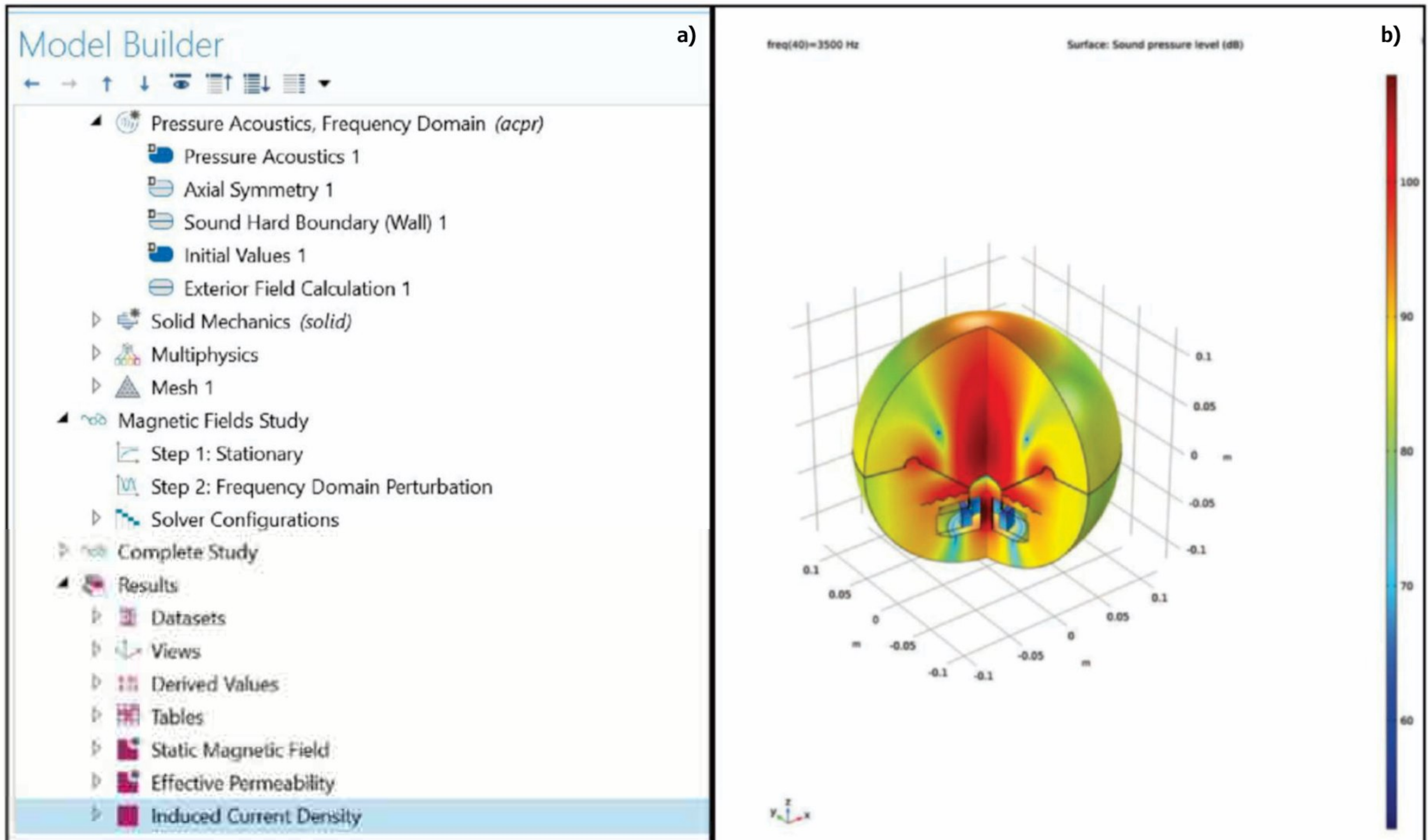


Figure 4: The list of materials and magnetic fields used are displayed in the feature tree of the Model Builder (a), as well as the 3D surface sound pressure levels in decibels (b).

engineering). And that is clearly its greatest strength and why it is finding so much success in audio applications, as it applies the power of multi-physics coupling and multi-method approach.

Multi-method provides four approaches: FEM, BEM, ray tracing, and dg FEM (unique to COMSOL numerical solvers using the discontinuous Galerkin method). The field of acoustics is by its very nature interdisciplinary and multi-dimensional. The wide adoption of this tool throughout the audio world should come as no surprise.

Loudspeaker engineers have long relied on customized personal tools for modeling and simulation, or commercial simulation tools

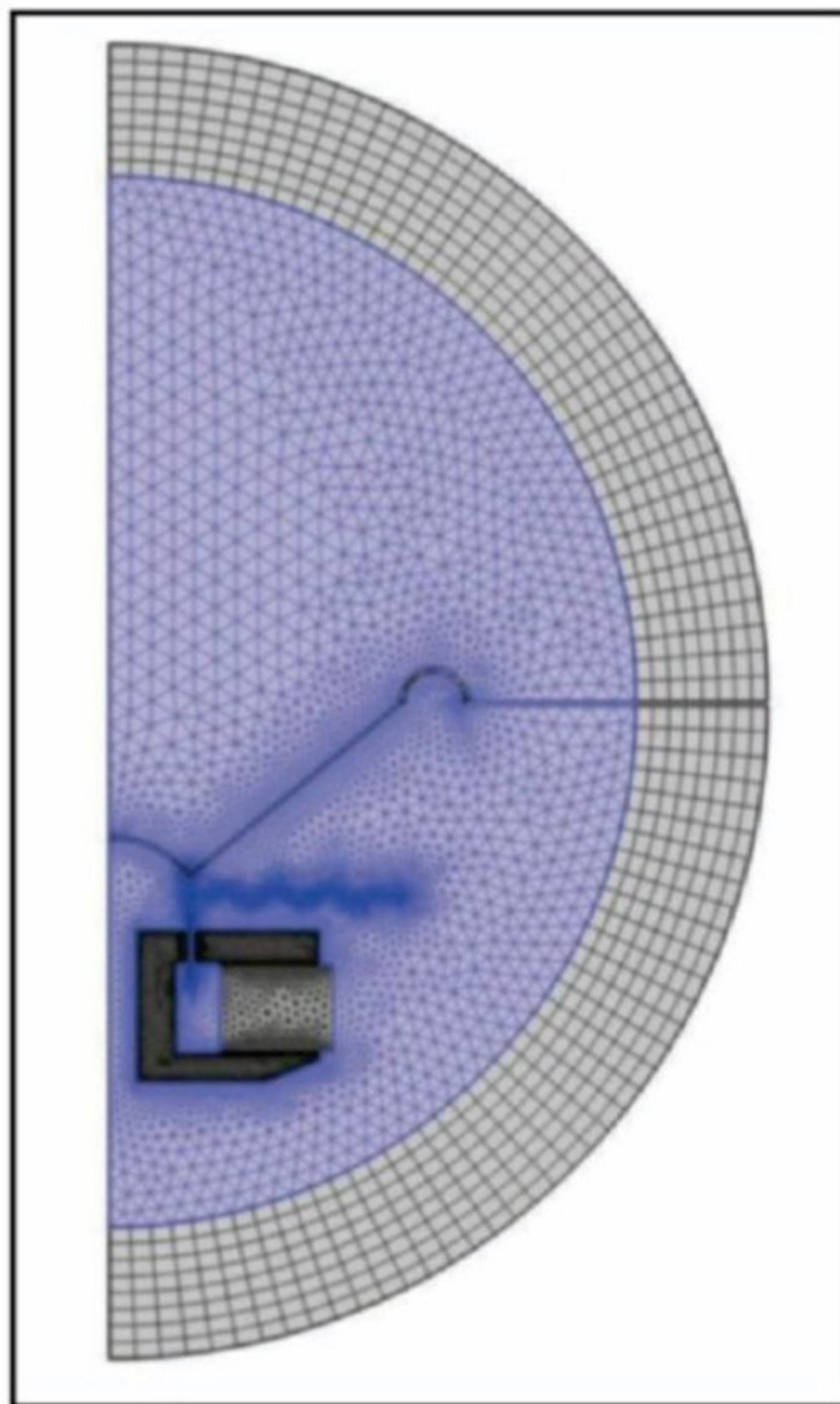


Figure 5: The triangular mesh is displayed in dark blue for the loudspeaker driver components, light blue for air around the driver. A rectangular mesh is used for the outer perimeter of the PML to eliminate boundary reflections.

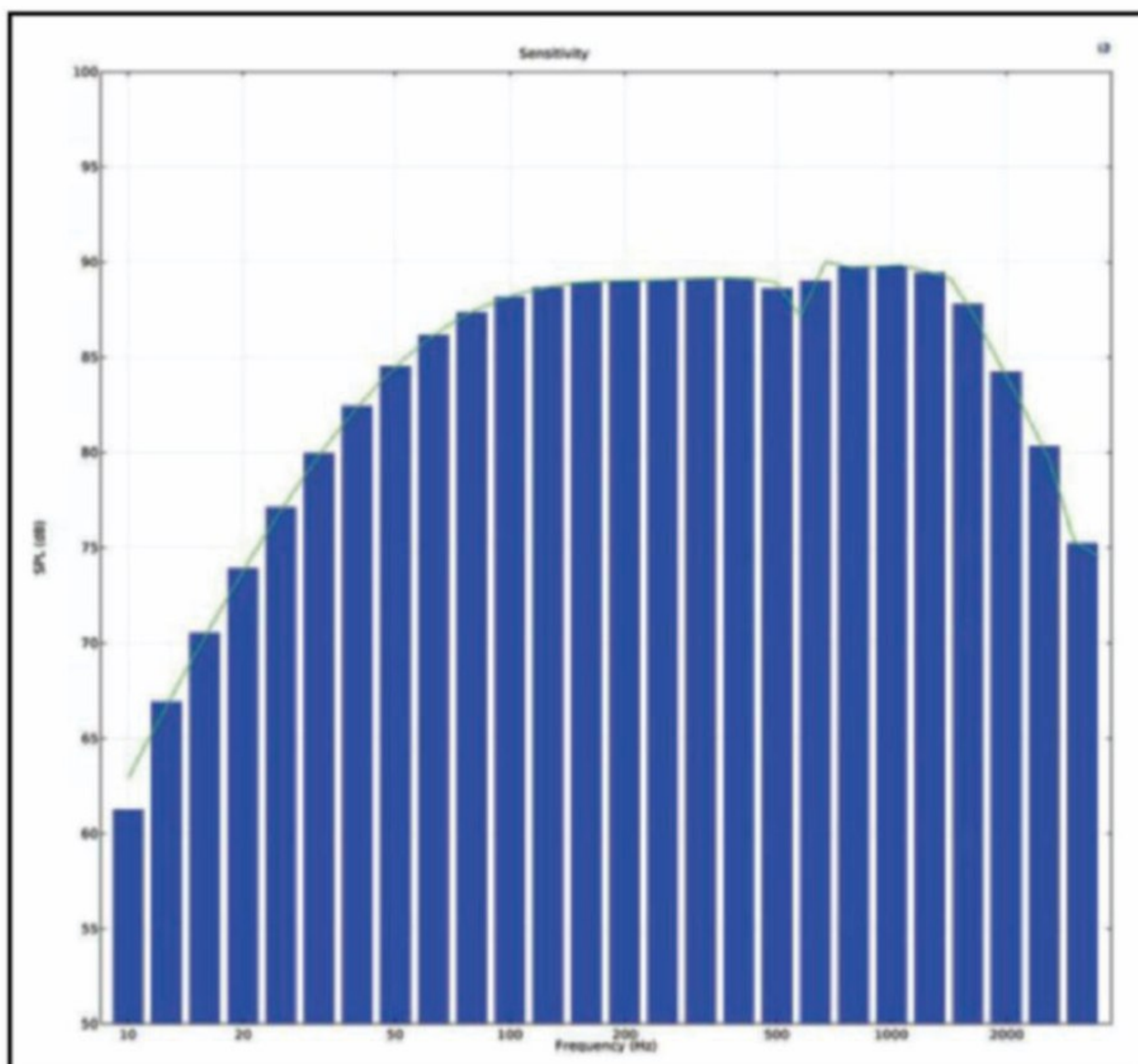


Figure 6: This graph shows the computed sensitivity plot for the driver.



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Figure 7: Here are computed real and imaginary impedance curves for the simulated loudspeaker driver.

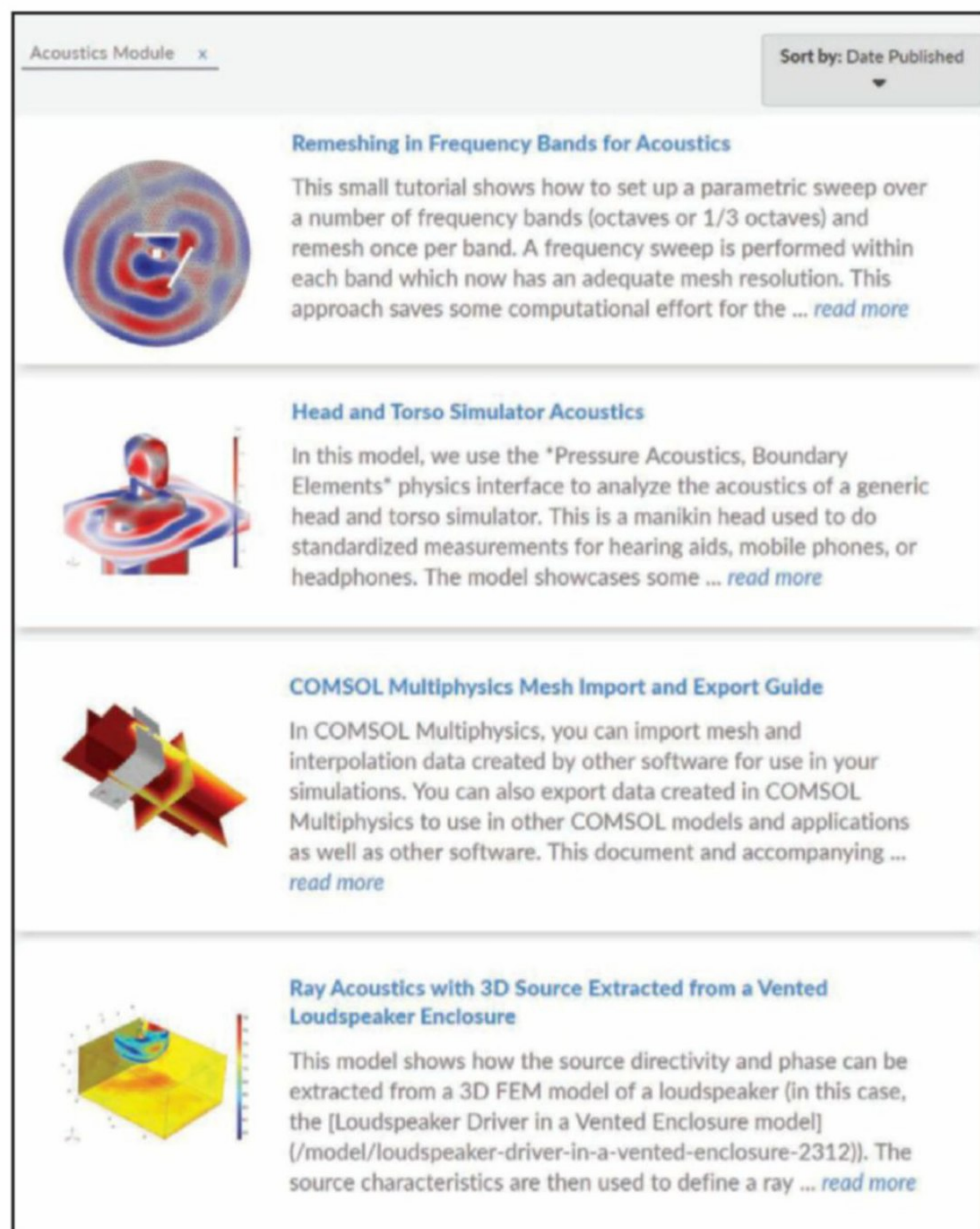
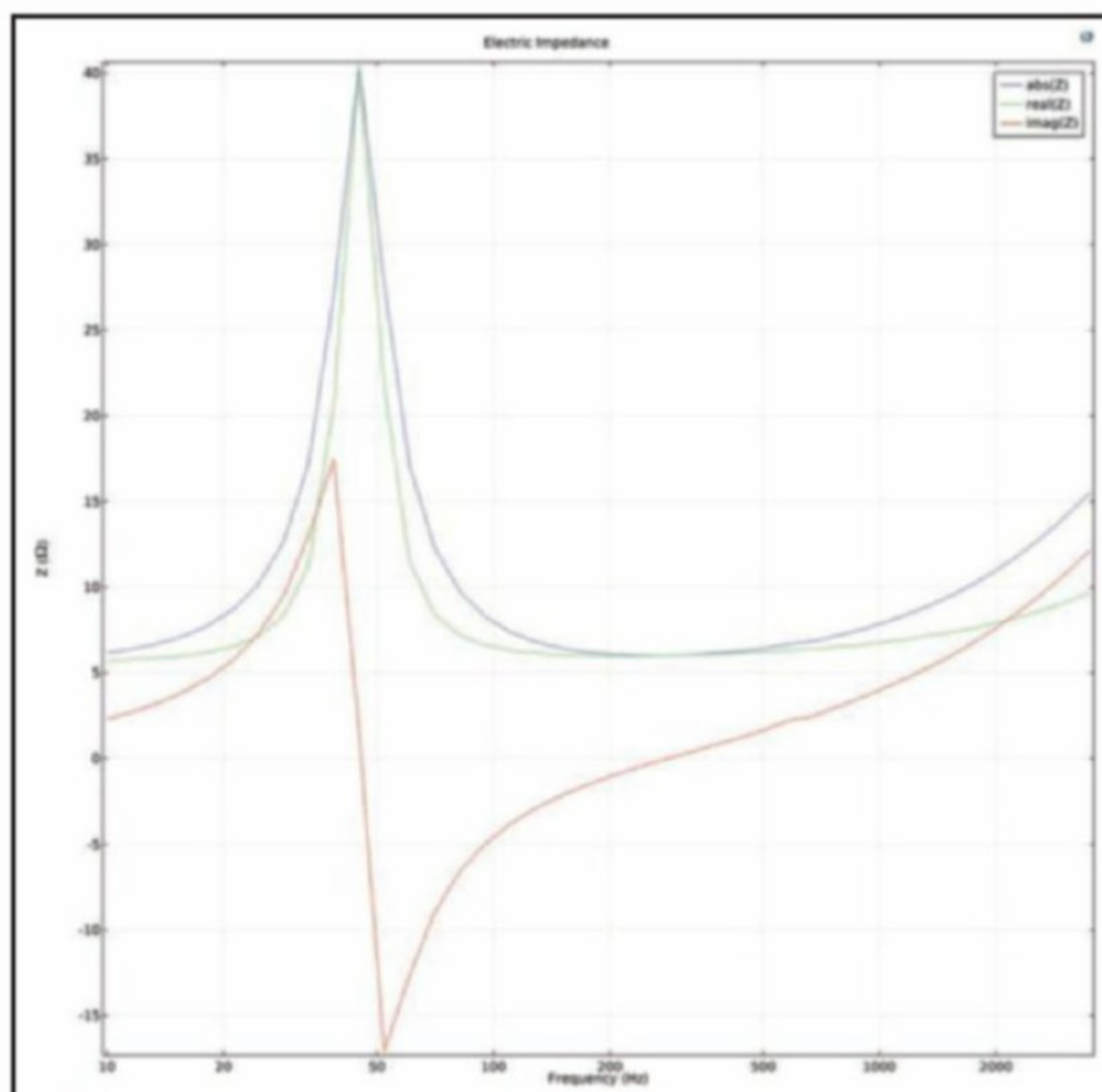


Figure 8: This is a sample of recently updated tutorials and examples from the Application Gallery.

that are often very narrowly focused (historians and veteran engineers have certainly explored other software packages such as XOPT (network simulation); LEAP/LspCAD (box, driver, and network simulation); FINEMotor and Finite Element Method Magnetics (FEMM), for magnetic circuit design; Enhanced Acoustic Simulator for Engineers (EASE), ray tracing for room and architectural acoustics; and AkAbak (lumped element and waveguide components).

One Workflow to Rule Them All

The first step in working in Model Builder is to define your geometry. You can either choose to use the built-in sketching tools, or import a CAD model. The Application Library provides a detailed example, "Loudspeaker Driver—Frequency Domain Analysis." This is an excellent introduction to linear analysis tools and theory. You will need to enter (or load from an external database) the critical parameters. In the case of an electromagnetic cone driver, only four electroacoustic parameters are required for linear behavior simulation: resonant frequency, flux density, number of turns on the voice coil, and the maximum drive voltage.

The next steps are to apply material properties to the components of the driver subassembly. Then you can enter appropriate values for the damping of the cloth spider and the resistivity coefficient of the voice coil (see **Figure 4**).

With the materials selected, the physics need to be defined, and conditions need to be set for Multiphysics coupling to the Acoustics-Structure boundary. The best practice is to use the Perfectly Matched Layer (PML) feature to avoid reflections and allow anechoic simulations of the loudspeakers frequency response at any arbitrary distance (see **Figure 5**). Then the model can be built and the relevant studies for sound pressure level and driver impedance can be solved (see **Figure 6** and **Figure 7**).

Other relevant loudspeaker models in the Application Library include several lumped element, transient, and balanced armature types. Using these as tutorials will pay great dividends in advancing your knowledge and COMSOL proficiency.

Thanks to frequent updates, the Application Gallery is an invaluable tool, providing tutorials, examples, demos, and documentation for learning more about modeling, or as a starting point for a new simulation (see **Figure 8**).

Ray Tracing Capabilities

The foundation of all EASE-based ray tracing and modeling is the application and usage of EASE .GLL files (a proprietary format for defining 3D acoustic data).

These models are typically available from loudspeaker driver manufacturers for existing products. For new designs, prototypes need to be built and then carefully measured through a time consuming process in order to proceed with a simulation.

Alternatively, if you have built models in COMSOL for new driver designs, then the prototype and measure step is not needed in order to begin simulations of a new loudspeaker in a large acoustic space. COMSOL's Mads Per Jensen has dubbed this process "sub-modeling," and this enables you to use the directivity of the loudspeaker source created in Model Builder in order to perform a ray tracing simulation without the need to use a stand-alone ray tracing software tool such as EASE. **Figures 9–11** show an example of a dome tweeter mounted on an automobile dashboard from a model first presented at the 2017 Audio Engineering Society (AES) Automotive Audio conference.

COMSOL has an elegant method of entering absorption coefficients for room materials by simply using an interpolation function on a text or Excel file. Furthermore, absorption, speed of sound, and density also can support graded materials, enabling users to model the effects of atmospheric temperature

gradients in the atmosphere or depth variations in underwater acoustics.

All of this flexibility is quite powerful and allows users the opportunity to model sophisticated designs

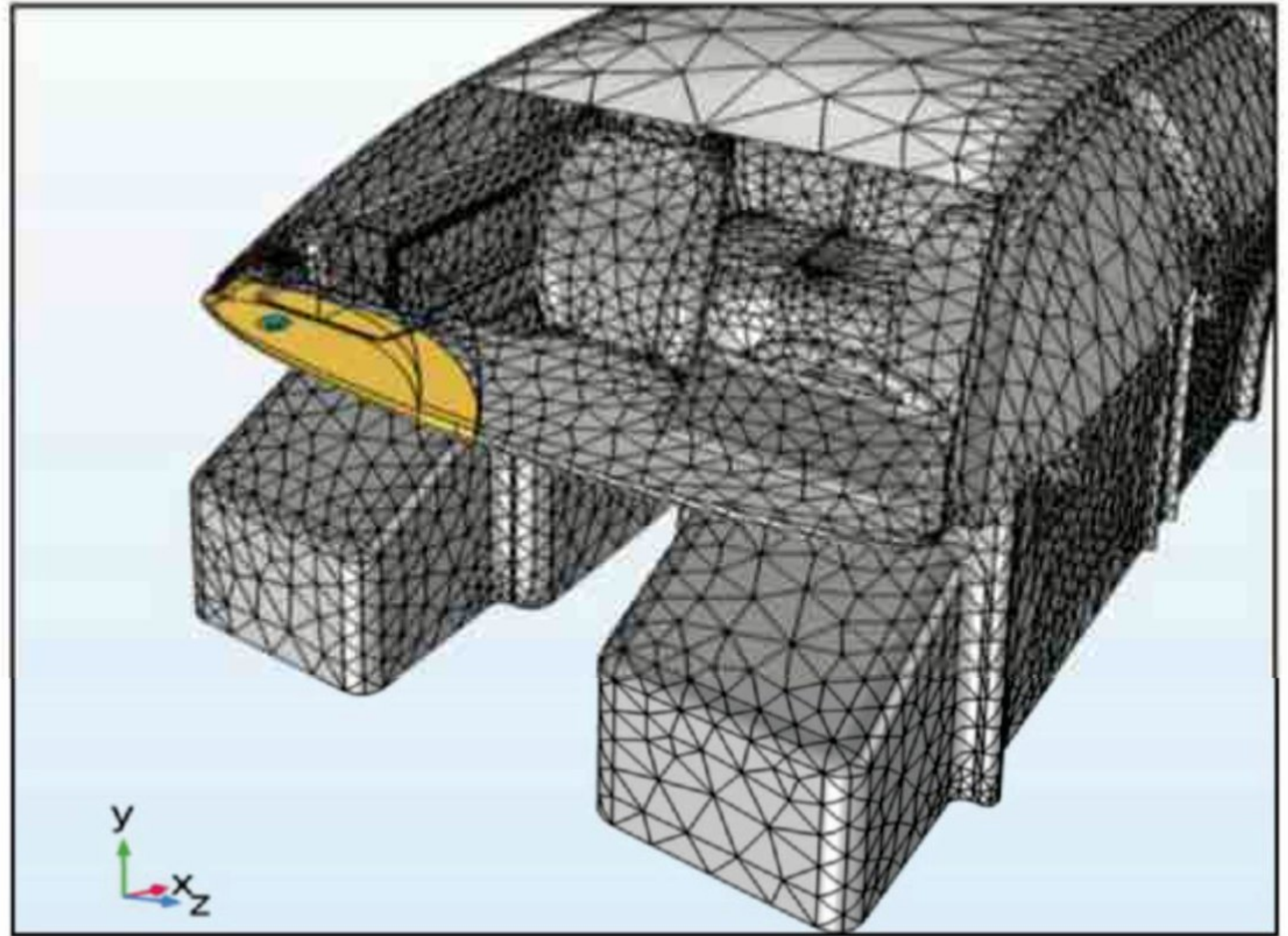


Figure 9: A dome tweeter in a dashboard is modeled and the environment is meshed.

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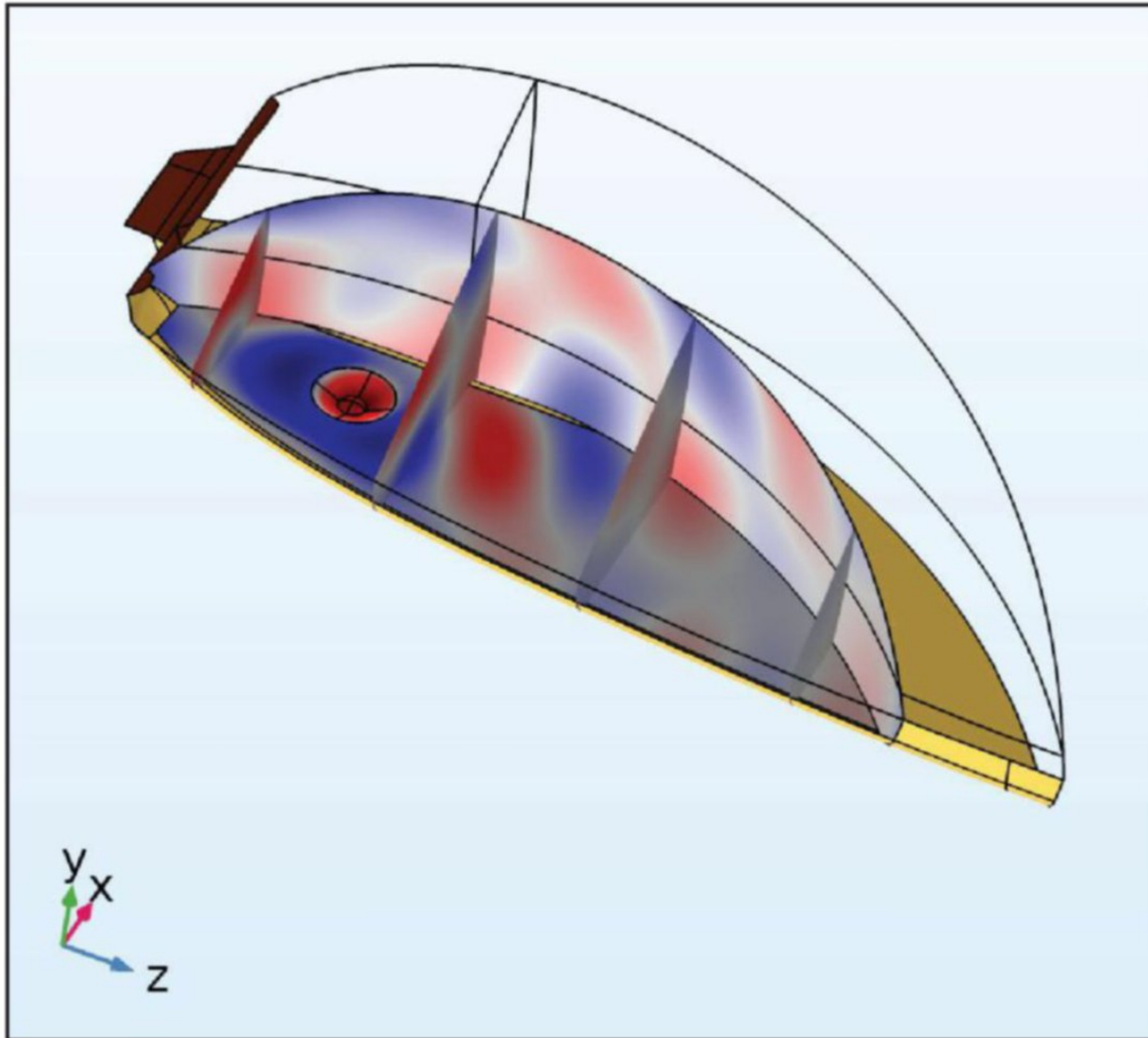


Figure 10: The directivity of the tweeter is modified due to its boundary loading.

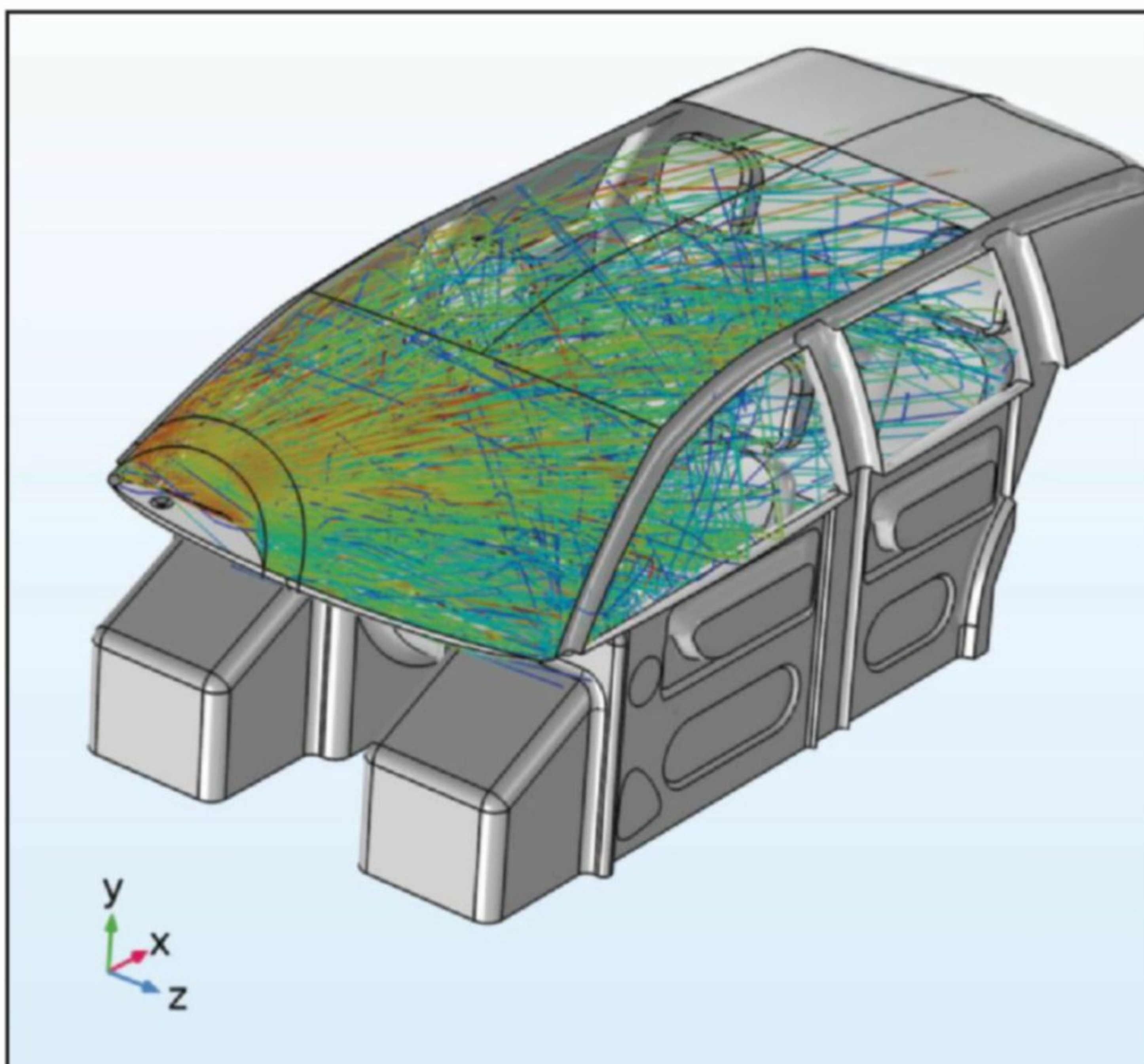


Figure 11: The FEM loudspeaker driver model is used to generate the direction and the magnitude of the rays used in the tracing propagation in this automotive audio system simulation. You can then calculate surface SPL, transfer function, or impulse response for the receiver.

that can accommodate real-world variations in the shells of the acoustic space, providing direct advantages over the built-in assumptions of a traditional ray tracing simulation tool such as EASE. This is very useful when working with ray tracing techniques beyond traditional architectural acoustics, (e.g., automotive environments with windows partially open or environmental noise studies involving highway sound barriers).

Application Builder

In the realm of numerical solver software, COMSOL is quite approachable. Still, there are many situations where an organization wants to share a portion of design “know-how” with team members and applications staff who might not have the time, knowledge, or mathematical training to build a COMSOL model.

Clearly, this democratizing benefit pays powerful dividends by sharing the structured toolboxes constructed by subject matter experts to those personnel or customers who are focused on practical results. Developing a COMSOL-derived app is a streamlined process and it can be built quickly, without the need for any code to be written.

The Application Builder is tightly integrated and linked to the Model Builder, allowing the creator to use simple forms that can be easily edited in a simple GUI. Ribbons can be used to improve the user experience, while error checking can be provided to prevent faulty results if the end user provides an input that is out-of-scope for the model. An executable file can then be shared with colleagues and customers, even those who don’t have a COMSOL license. This is a particularly good option when a designer wants to share results without sharing IP or the underlying models.

When a user installs the first standalone COMSOL built app, an additional and one-time download of the COMSOL Runtime environment is required. As the user installs more apps, no additional files are needed. Having the Runtime environment as a separate install, as opposed to building it in to the installer of every app, helps minimize file size and aids in distribution. On the negative side, there is no version control for these standalone apps. In addition to the COMSOL Compiler that is used to build the executable file, there is also a COMSOL Server application that can assist in deploying and monitoring app output, as well as manage computational loads on cloud cluster arrays.

Final Thoughts

The COMSOL 2019 Boston Conference provided both insights into the utilities of the software (Application Builder, LiveLink, working with CAD, and deploying applications), as well as technical tracks targeting specific subjects (acoustics, structural mechanics,

electromagnetics, ray optics, multiphase flow, composite materials, heat transfer, and so many more topics).

A trained engineer or researcher can manipulate COMSOL to provide the simulation power and leverage a vast array of numerical solvers to analyze any sound and vibration problem, all within the design language of a single software package. This will also allow scaling both computing and human resources as needed to tackle the most interesting problems.

Application Builder allows the power of simulation to be leveraged throughout an organization. This extends the understanding of the mathematics and physics of the SME to individuals attuned to applications and specific customer requests. Given the memory and storage requirements for large models, the ability to use Cloud Cluster Arrays (CCA) to accelerate time for solution generation is a strong advantage for any organization that needs to reduce their processing time. As mentioned, the ability to tightly integrate and leverage existing code built in other software packages (e.g., MATLAB, Excel, and SolidWorks) provides the bridge from current in-house tools to newer next-generation tools, applications, and models that are built from COMSOL. 

Resources

Dr. A. Bezzola (Samsung America), "Optimal Design of Bass Reflex Loudspeaker Ports," COMSOL Conference 2019.

<https://comsolconference2019boston.exordo.com/programme/presentation/159>

K. Gandomi, P. Carvalho, Z Zhao, C. Nycz, and G. Fischer (Worcester Polytechnic Institute), with Everette Burdette (Acoustic Med Systems), "Thermo-Acoustic Simulation of a Piezoelectric Transducer for Interstitial Thermal Ablation with MRTI Based Validation," COMSOL Conference 2019.

<https://comsolconference2019boston.exordo.com/programme/presentation/168>

S. Shubham and M. Nawaz (Knowles Electronics), "Estimating Parasitic Capacitances in MEMS Microphones using Finite Element Modeling," COMSOL Conference 2019.

<https://comsolconference2019boston.exordo.com/programme/presentation/120>

About the Author

David Logvin is an acoustics engineer at TDK InvenSense, working in the MEMS microphone division. With more than 20 years of experience, he has designed loudspeakers and other products for Snell Acoustics, Outlaw Audio, ClearView Audio, Cambridge Sound Management, and others. During his 14 years at Snell Acoustics, he developed and designed dozens of high-end loudspeakers and amplifiers, as well as helped to create loudspeaker standards in close collaboration with THX. David then focused his efforts on R&D, creating innovative, award-winning piezo-driven membrane loudspeakers with ClearView Audio (formerly known as Emo Labs). At Cambridge Sound Management, he put his skills in room acoustics to use in the field of sound masking for corporate applications. Most recently, he rejoined his fellow Snell Acoustics alumni, Dr. Joseph D'Appolito, to design several new loudspeakers for Outlaw Audio. In addition to his lifelong love of music and acoustics, David is an active participant in the Boston tabletop game design community. He is a member of the Audio Engineering Society (AES) and the Boston AES Section and lives in Lowell, MA, with his family.

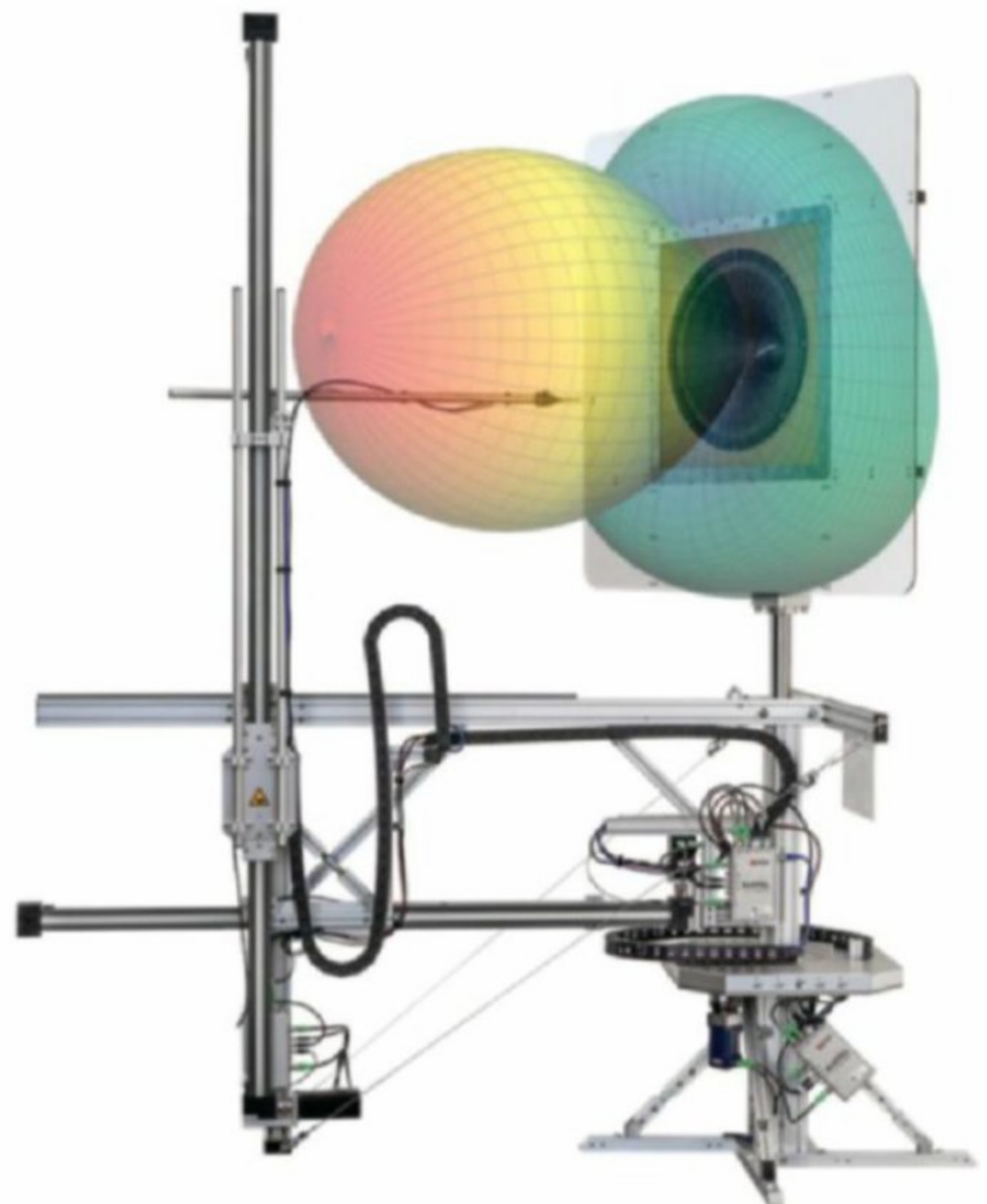
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Big Promises in Large and Small Parameters

In this Market Update, we address new trends in drivers and transducers, from new diaphragm materials for loudspeakers to woofers, tweeters, and compression drivers, to microspeakers and drivers for headphones. And we share some vital market data that shows how speaker technology and components continue to be at the core of a thriving industry.



By
J. Martins (Editor-in-Chief)
and **Mike Klasco**

KV2 Audio's ESD Cube is the most compact loudspeaker yet in the company's history, showcasing one of the trends in professional audio. The ultra-compact passive 5" loudspeaker is ideal for applications requiring discreet, high-quality peripheral and/or distributed sound reinforcement including bars, hotels, houses of worship, museums, tourist attractions, and corporate venues.

Even though the story of the invention of the loudspeaker is now more than 100 years old and many industry veterans still like to say that very little has changed in decades, speakers continue to be a thriving business with no signs of slowing down. As British market research firm Futuresource Consulting recently stated, the world of audio is in ascendance, "with renewed consumer interest and a groundswell of technological innovation from couch to car." And it all indicates that the growth curves will continue to track upward in 2020.

Because of this growth, some of the market research organizations that traditionally follow technology are starting to look more carefully at key product segments within the consumer electronics categories, including a first serious look at the growing professional and commercial audio application categories. And this renewed interest in the audio industry allows us to obtain a valuable overview of the loudspeaker economy itself.

In 2019, the global professional speaker market size is expected to reach more than \$2 billion USD, growing at a compound annual growth rate (CAGR) of approximately 3% during 2018–2024. Line array speakers are expected to experience the highest CAGR at about 3%. This, according to the latest Arizton's

"Pro Speakers Market—Global Outlook and Forecast 2019–2024" research report.

Being one of the very few dedicated studies about the professional speaker market, it is difficult to say how accurate those valuations and forecasts from Arizton actually are, especially given the fact that there's still a lot of production from China that is challenging to correctly take into perspective, both in terms of revenues and volumes. Arizton says the Asia-Pacific (APAC) and US markets are the fastest growing and that the demand for pro speakers in the large venues and events segment are also growing significantly.

Arizton also predicts a high-impact from the introduction of connected devices and voice recognition technologies in the global professional speakers market and in particular for corporate and commercial applications, as the report details.

According to this research, point source systems are the most popular product class, capturing more than one-third of the pro speaker market. Line array speakers are the second most demanding variant, followed by subwoofer and low-frequency devices. The corporate segment represents nearly one-third of the market due to the growing demand for sound reinforcement and conferencing solutions. With the rise

in live performances, music concerts, festivals, and music tours, the entertainment segment represents the second largest application market.

As Arizton’s research confirms, North America captured around one-third of the professional speakers market. “The region is the highest contributor to the market in terms of revenue as well as unit shipment.”

According to another recent forecast from Arizton— focusing on the wider sound reinforcement market, which also encompasses microphones, speakers, amplifiers, mixers, and processors—the industry will see revenues reaching \$9 billion USD by 2024, and growth at a CAGR of 3.33% during the period 2018-2024.

Our only way to gain some perspective over the Arizton report is to compare it with the Audiovisual and Integrated Experience Association (AVIXA) research about the global professional audiovisual industry. According to the latest *Industry Outlook and Trends Analysis (IOTA) Global Summary* produced by AVIXA in collaboration with market research firm IHS Market, the AV sector will grow to more than \$325 billion USD in 2024, from \$247 billion in 2019. According to AVIXA, the overall demand for pro-AV products and services—which includes and clearly identifies product segments such as loudspeakers and microphones—is rising across much of the globe. The AVIXA research forecasts audio equipment revenues to reach \$6.7 billion in 2024, in the US alone. This would confirm an even rosier picture for the audio industry globally.

Home Studio and Pro Studio Markets

Futuresource Consulting has also studied the pro audio segment—looking more specifically at the studio and recording categories, which encompass a large range of application segments. According to Futuresource, the market for pro audio products in professional and home studio applications posted global revenues of more than \$1.2 billion USD for 2018, an increase of more than 6%, according to the industry report. Covering mixers, microphones, audio interfaces, studio monitors, and professional headphones, the report is the first of its kind, and describes continued growth in volume and value out to 2023, at a CAGR of 9.7% and 6.5%, respectively.

“Pro audio products operate in a dynamic market, with increasing competition between professional and home use,” according to James Kirby, Pro Audio Analyst at Futuresource Consulting. “The home content creation space is flourishing. An individual in their bedroom can create for a few thousand dollars what would have cost tens of thousands of dollars in the past.”

“The home studio segment is now responsible for more than half of total studio vertical revenue and the



In the search for increasing performance in high-power, low-frequency applications for sound reinforcement, Eighteen Sound—now part of the B&C Speakers SpA group of companies—launched the new 18NTLW500, 21NTLW5000, and 12NTLW3500 subwoofers using the impressive Tetracoil Double Voice Coil (TTC) technology that allows reaching very high sound pressure levels. In this approach, two different axially separated magnetic gaps and two inside-outside voice coils are wound on the same former and suspended evenly in the two magnetic gaps.



Driver design for smart and portable speakers faces increased competition to create enhanced functionality and ease of use. To succeed, designs must distinguish themselves in terms of overall system cost, battery runtime, thermal considerations, voice recognition error rate, and premium audio quality. And reducing system cost while increasing efficiency is a priority in the segment. Pictured is the Bose Portable Home Speaker.



In high-end audio—where cost is not so much a consideration—the race for higher fidelity and improved specifications continues to be the hallmark. Pictured are the PMC two-way fact.8 signature and three-way fact.12 signature high-performance loudspeakers.

numbers are climbing. One result of this revolution has been the disappearance of many small and medium-sized recording studios. However, these have been replaced by millions of amateur musicians, podcasters, streamers, and content creators, working from home studios and fueling a high-growth market for pro-audio hardware and software. Despite the challenges faced by pro studios, a new ecosystem is now emerging, where professional and home studios are complementing and supporting each other,” says Kirby.

Another important trend impacting the sales of studio monitors and recording equipment is the rising

popularity of Podcasts, which is causing an explosion in related content audio production.

And although home studios are competing in the music studio, podcast and vlogger space, the creation of broadcast and game content, as well as audiobooks—another growing format—remains largely the reserve of professional studios. Demand for both types of content continues to grow significantly, particularly games, driving the value for studio equipment vendors in the professional segment.

“Games typically require 3D audio, a higher quality, which is required if a game hopes to become popular and achieve a foothold in the lucrative exports market,” says Kirby, “and 3D sound is also vital for VR games. Our research in content markets indicates that worldwide consumer spend on games exceeded \$130 billion in 2018, an increase of more than 10% compared with 2017. Against the backdrop of all this consumer activity, game developers are putting more money and resources into providing high-quality audio, including in-game music, voiceover and sound effects. It’s all great news for pro audio vendors.”

Home Audio Market

The worldwide home audio hardware market grew 20% in 2019, also according to Futuresource. The sector, which includes wireless speakers, soundbars, A/V receivers, Hi-Fi systems and dedicated speaker docks, also witnessed higher revenue growth last year, with trade values reaching \$15 billion.

The wireless speaker category continues to be the largest growing product category, fueled by smart speakers. Futuresource forecasted 53% shipment growth in 2019, also fueled by new product categories, which combine voice-controlled speakers with screens and the introduction of voice in many new types of devices and appliances, all requiring complete audio systems, including speakers. Smart speakers with screens took off in 2018 and continued to accelerate in 2019.

The US remains the world’s largest smart speaker market, with Amazon and Google leading the way. China takes the second position, despite traditionally being a country where home audio products perform poorly. Across Western Europe, smart speakers are now widely available, although the UK remains by far the leading country, given the general availability of more advanced English language voice assistants.

Bluetooth speakers, in general, still maintain a solid market share where smart speakers have not yet taken hold, and the more high-end, hi-fi speakers also continue to represent a growing category. While the more traditional home audio equipment categories (e.g., integrated receivers, portable digital audio players, PC/multimedia speakers, and traditional



“With Echo Studio, which supports Sony’s 360 Reality Audio, we have engineered a completely new listening experience for customers. Through one, simple device they will be able to immerse themselves in music like never before, hearing each song the way its creators intended it to be heard,” says Miriam Daniel, VP of Alexa and Echo Devices, Amazon.



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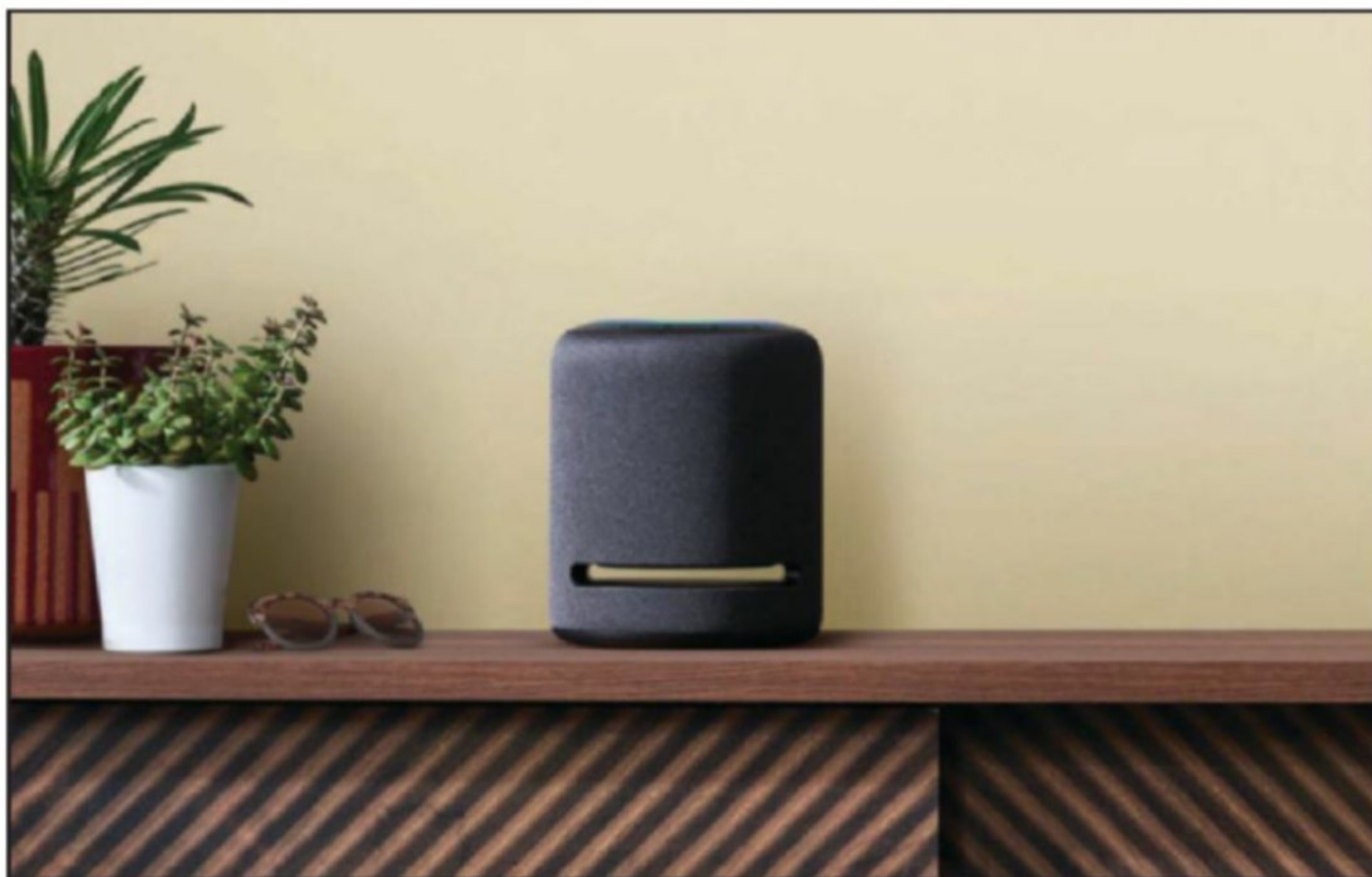
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The new Amazon Echo Studio will be able to decode and playback immersive audio content, including Dolby Atmos and Sony 360 Reality Audio.

speakers) are all trending negatively, the good news is that wireless audio technologies will soon enable a strong return in the home theater category, including in the more affordable segments. Thanks to the efforts of the Wireless Speaker and Audio (WiSA) Association—founded by Summit Wireless Technologies and now comprised of more than 60 leading consumer electronics brands—the industry can get behind a single standard solution for wireless home cinema and surround gaming entertainment, something from



Targeting high-end gaming, the Audeze LCD-GX headphones use Audeze's gigantic Fluxor magnets and Uniforce diaphragms and a detachable microphone.

which the consumer and hi-fi speaker manufacturers could benefit extensively.

Soundbars are another highlight of the home audio segment, growing more in value than volume as consumers continue to explore the benefits of premium options. Looking again at the Futuresource research, shipments increased 6% in 2018, while value growth was double this, at 12%. New smart soundbars, now equipped with more sophisticated DSP and voice recognition have been released in 2019, and more are planned for 2020, as Futuresource confirms.

While the soundbar is now moving beyond its role as an accessory to improve TV sound, 3D audio, such as Dolby Atmos and new immersive formats like Sony 360 Reality Audio are also expected to cause renewed interest in wireless speakers and soundbars with sophisticated digital processing. Sony's 360 Reality Audio is an immersive audio experience utilizing MPEG-H object-based spatial audio technology from Fraunhofer IIS.

The promotion and availability of more content in these immersive formats, from broadcast to streaming services, including for music, is also inspiring a new generation of more sophisticated and better speakers, as already seen with the new Amazon Echo Studio.

Sony Corp. announced its 360 Reality Audio content initiative at CES 2019, and the new immersive music experience is now available on the new Amazon Echo Studio speaker with content available on Amazon Music HD, Qobuz, and Tidal to listeners in the US, the UK, Germany, and Japan later this year. Amazon also partnered with Dolby to promote music remixed in Dolby Atmos.

All this activity around immersive music formats distributed by streaming services, added to the increasing availability of movies and TV series with Dolby Atmos and DTS:X soundtracks, is certain to cause an impact in the market with new speaker form factors needed to serve the home audio market. This, in turn, will certainly inspire the professional monitor and residential home theater segments to create new solutions better adapted to the needs of object-based spatial audio.

As *audioXpress* already noted in its Market Update dedicated to the installation market (October 2019), this renewed interest for immersive audio production and content distribution will only benefit the already ongoing trend of immersive audio configurations used in professional sound reinforcement for live shows, touring and even fixed installations.

Global Headphone Market

And while the market is agitated by these announcements from Amazon, Dolby, or Sony and the promise of new immersive audio content now

being easily accessible through streaming services, the global headphone market continues to expand, benefiting directly and indirectly from all this activity in the technology and content space.

According to the latest “2019 Headphones Market Report” from Futuresource, 2019 was another year of growth, with 5% year-on-year in volume and 14% in value. True wireless continues to be the fastest growing category, with average sales prices increasing as consumers demand new premium features such as active noise canceling (ANC) and voice recognition.

But wireless headphones continue to sell very well, including portable models with noise-cancelling, and growth in the premium high-end segment, which continues to capture the imagination of consumers, with global revenues in 2019 growing nearly four times faster than shipments.

“True wireless now accounts for almost one in every five shipments and has established itself as the driving force behind the unshakeable growth in headphones,” says Adriana Blanco, Senior Market Analyst at Futuresource Consulting. This is certainly a nice indicator for all manufacturers of balanced armatures and the more efficient dynamic drivers that are used in TWS earbuds, a segment where there is still some marking for differentiation in higher quality and more power-efficient designs.

Other important trends in the global headphone market, particularly when looking at the trends affecting the choice of drivers, are the growing demand for sports headphones, with users expecting water, moisture, and sweat resistance; and also a significant expansion of gaming headsets, where users are demanding not only the best quality but are also looking at designs that allow less listening fatigue and can be used for longer periods of time. In response to that rising category, California high-end headphone manufacturer Audeze launched its first high-end purist gaming headphones—the Audeze LCD-GX—using the brand’s large Fluxor magnets and Uniforce diaphragms, combined with a detachable microphone and flexible gooseneck arm. With the gaming segment inundated with cheap, plastic headsets, maybe this will signal a trend for a new product category all together: high-end audio gamers.

Louder Microspeakers

But when it comes to cutting-edge technology in drivers and transducers, we need to mention the smallest (in size) classes, where a lot is happening and the volumes are by far the largest.

With more than 1.6 billion mobile phones built annually, each with a microspeaker for the speakerphone and another for the receiver, this side of the speaker industry attracts the giants. Top players



include AAC, Goertek, Gettop, ForGrand, Bujon, and Merry Electronics along with others that share the sizable market—which includes laptops and tablets.

Now with the leading smartphones priced at \$1,000 and more, the industry in general is also looking at better quality sound. The speakerphone function on most smartphones goes south at around 750 Hz to 800 Hz, not so impressive when 300 Hz to 3,000 Hz has been considered a marginal standard for telephony speech for about 100 years. It appears that while we expect reasonable sound from our soundbars, auto-sound systems, and headphones, when we pick up our beloved smartphones, we leave our expectations behind.

When the latest iPhone model costs \$1,400, which is about the cost of the Devialet Phantom reactor 900—the best-sounding speaker available in Apple stores (with real bass to 18 Hz)—it is hard to accept how bad the iPhone microspeaker buzzes when pushed hard and Apple did improve the sound quality a lot in the past three generations.

That works until LG, TCL, Huawei, or OPPO Digital comes along with something closer to “hi-fi” quality and an extra octave of bass response without the rattles. And they are also working hard in that area.

We are in for a step-change in sound quality from our smartphones. The smartphone platform is comprised of a system-on-chip or chipset (SoC), which includes audio codecs, DACs, and power amplifiers—and actually these are all quite decent.



A display of Microspeakers for multiple applications, from in-earphones, headphones and portable speakers at the Onkyo booth during Mobile World Congress (MWC) 2019.

AAC Technologies is a leading supplier of microspeakers, including ultra-thin speakers for automotive applications, capable of embedment in headrests, headliners, and other areas.

ESS Technology, Asahi Kasei Microdevices (AKM), Cirrus Logic (previously Wolfson), Texas Instruments, Infineon, NXP Semiconductors, Qualcomm, and even commodity vendor Realtek more than deliver the goods on the circuit performance.



There's not much to see in this photo of the AAC Technologies/Dirac Research demonstration of sound in a smartphone, but this was an impressive demo of the Dirac Distortion Control, a patent-pending audio solution to reduce audible distortion in smartphones and any device that uses microspeakers to play music.



The 2019 LG G8 ThinQ smartphone stands out for its screen speaker and improved audio features. The new LG G8 ThinQ features a Crystal Sound OLED (CSO) panel, an innovative technology that utilizes the phone's OLED display as a speaker, reinforced with a bottom speaker used as a resonance chamber to generate more bass and volume. To generate even more volume and leverage this dual speaker combination in the new phone, at MWC 2019 LG was showing a speaker dock where users can place the LG G8 ThinQ and get room-filling sound. The company calls this a "Boombox Speaker."

Achieving wideband clean sound from the microelectromechanical systems (MEMS) mics is mostly a design effort of the mic port stack-up configuration and acoustic wind noise suppression techniques so as not to degrade the mic's performance. The microspeakers in the smartphone are going to be the limiting consideration in speakerphone and stereo audio playback (gaming, movies, music, etc.).

The push to reduce the footprint and depth continues, but the microspeaker depth (Z-axis) has ratcheted down to 2.5 mm from 3 mm for smartphones—driven by the evaporation of the 3.5 mm headphone jack, which had been the bottleneck. The same product development teams that are demanding increased efficiency, power handling, cleaner sound, and extending the bass and top-end responses are also demanding the lowest possible price.

Several approaches are being used to overcome the limitations of size, acoustic output, and power handling in smartphone microspeakers. One path is to overdesign the speaker to absorb more power while maintaining practical mechanical excursion limits. Another more productive path is using DSP to maximally drive the speakers close to the edge of their safe operating area (SOA) without overdriving or damaging the speakers. The SOA can be crudely defined as the power limits, but it is more accurately defined as the thermal and mechanical limits.

Sensitivity and the power to drive the speaker to desired levels are not just limited by the amplifier's available power. The speaker must also have the capacity to handle this power. Two parameters must be respected. We must limit over excursion at the lower frequency range to prevent the moving parts from going beyond where they should. Higher excursion can result in bottoming where the voice coil hits the magnetic structure. We must also be careful not to exceed thermal limits, which are the limiting factor in the midrange and top-end. Clipped amplifier output and the ring tones can build up heat in the speaker's voice coil faster than it can be dissipated, resulting in thermal failure.

What if we could move more air (pumping power or volume velocity) without more piston area or increased excursion? A recent innovation (US Patent 9.294.841 B2, "Acoustic Radiator Including a Combination of a Co-Axial Audio Speaker and Passive Radiator") by speaker guru Joseph Sahyoun, which is close to productization, compliantly suspends the microspeaker with a flexible membrane that encircles the speaker. The kinetic energy that is generated by the moving mass is tuned to be in phase to extend the bottom end almost an octave, essentially using the entire microspeaker concurrently as a passive radiator.

Magnetic structures are also heading for a shake-up. The Z-axis stack-up is a fight for space and with only 2.5 mm to 3 mm there is barely room for any diaphragm excursion. Help is on the way as higher mGO (e.g., N40 to even N52 neodymium) can be thinner than the garden variety neodymium, yet pack more punch. While not breaking news, the issue has been that the strong flux is lost in the magnetic return structure.

Recently, Carpenter Technology has introduced soft metals for the return structures enabling less flux loss with 40% thinner magnetic structures. Some experts vote for more excursion, but most industrial design teams will push for a slimmer microspeaker.

Microspeaker diaphragms range from paper, thermoplastic films (e.g., PET, PEN, PEI, PEEK, etc.) to laminated composites. Some developments include liquid injection-molded (LIM) surrounds, which enable more consistent bass roll-off point and slightly reduced enclosure volume.

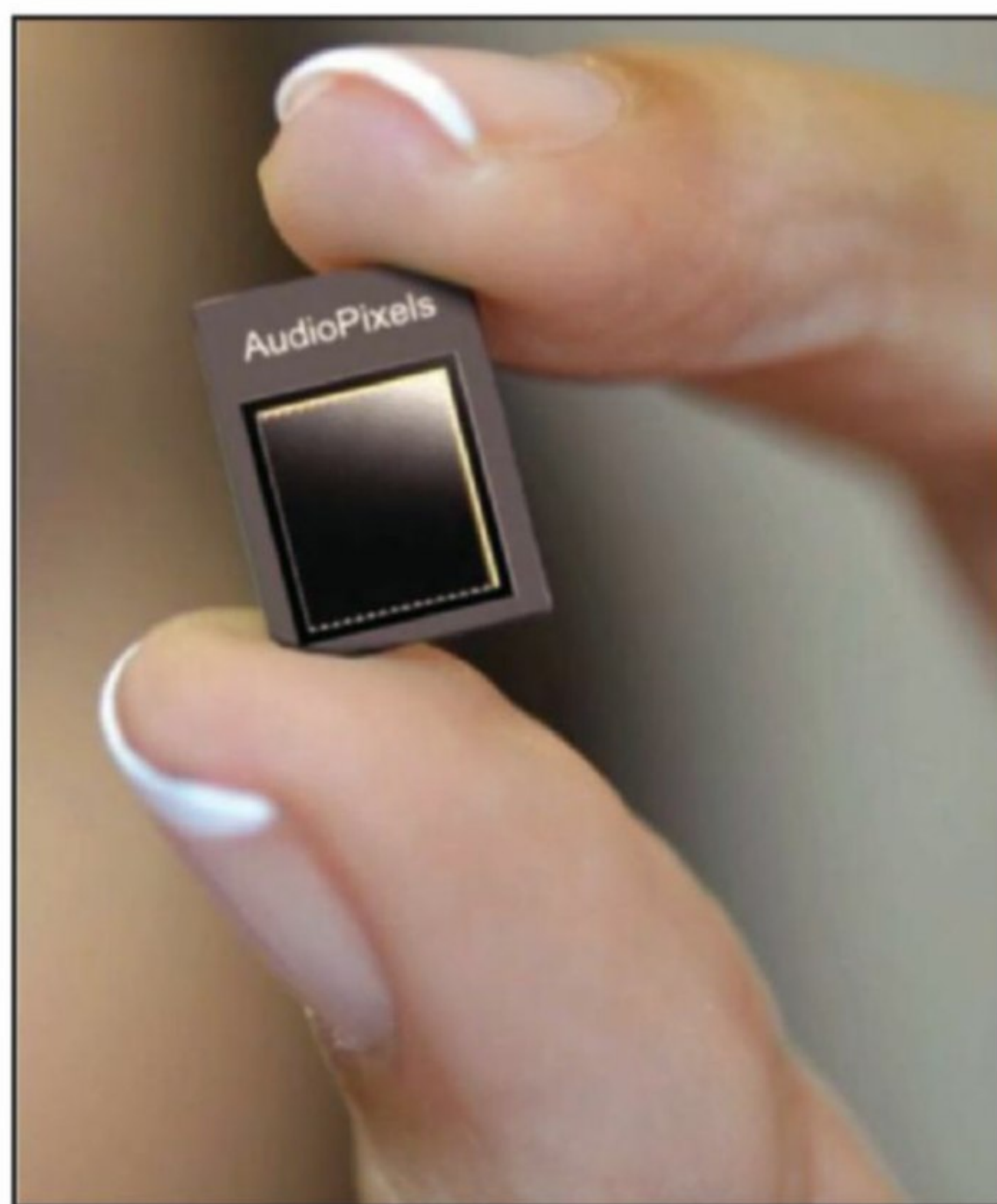
AAC Technologies has demonstrated its enhanced SLS microspeaker technology, allowing a noticeable improvement over the current crop currently used in smartphones. And AAC has also teamed up with DSP-specialist Dirac Research to deliver a complete solution. After two years of collaboration, the two companies finally unveiled a series of breakthrough product prototypes that demonstrate the now-possible audio experience available across all consumer technology product categories.

Other prototypes already demonstrated include a smart speaker, smartphones, an augmented reality headset, and an ultra-thin automotive speaker—all of which are custom-configured with sophisticated AAC-Dirac audio subsystems.

The automotive speaker demo explores how ultra-thin speakers, capable of embedment in headrests, headliners, and other areas, can be used to create individual soundstages within the vehicle. All demos feature AAC's new high-performance Super Linear Speaker.

MEMS Microspeakers

While MEMS mics dominate the market, having progressively taken over from electret condenser microphone (ECMs) in high-volume consumer electronics and smartphones, MEMS speakers are part of a relatively new research field. MEMS are the technology of very small devices, usually consisting of a micro-transducer and an application-specific integrated circuit (ASIC). Expanding MEMS development from microphones to microspeaker is an epic challenge as while the diaphragm excursion on a mic is minuscule, a MEMS microspeaker is going to have to move some air. While the industry



Audio Pixels developed a MEMS-based digital speaker chip that uses entirely new techniques to generate sound waves directly from a digital audio stream. This allows companies to mass produce a single type of a MEMS-based silicon chip that can be used either as a stand-alone speaker or cascaded in any multiples to achieve the desired performance specifications.

has been able to digitize and shrink all other device electronics, the last remaining barrier is the speaker, which remains bulky and very analog.

There are half-dozen MEMS microspeaker development initiatives, including USound's piezo-actuator, GraphAudio's graphene electrostatic, and Audio Pixels' ultrasonic modulator.

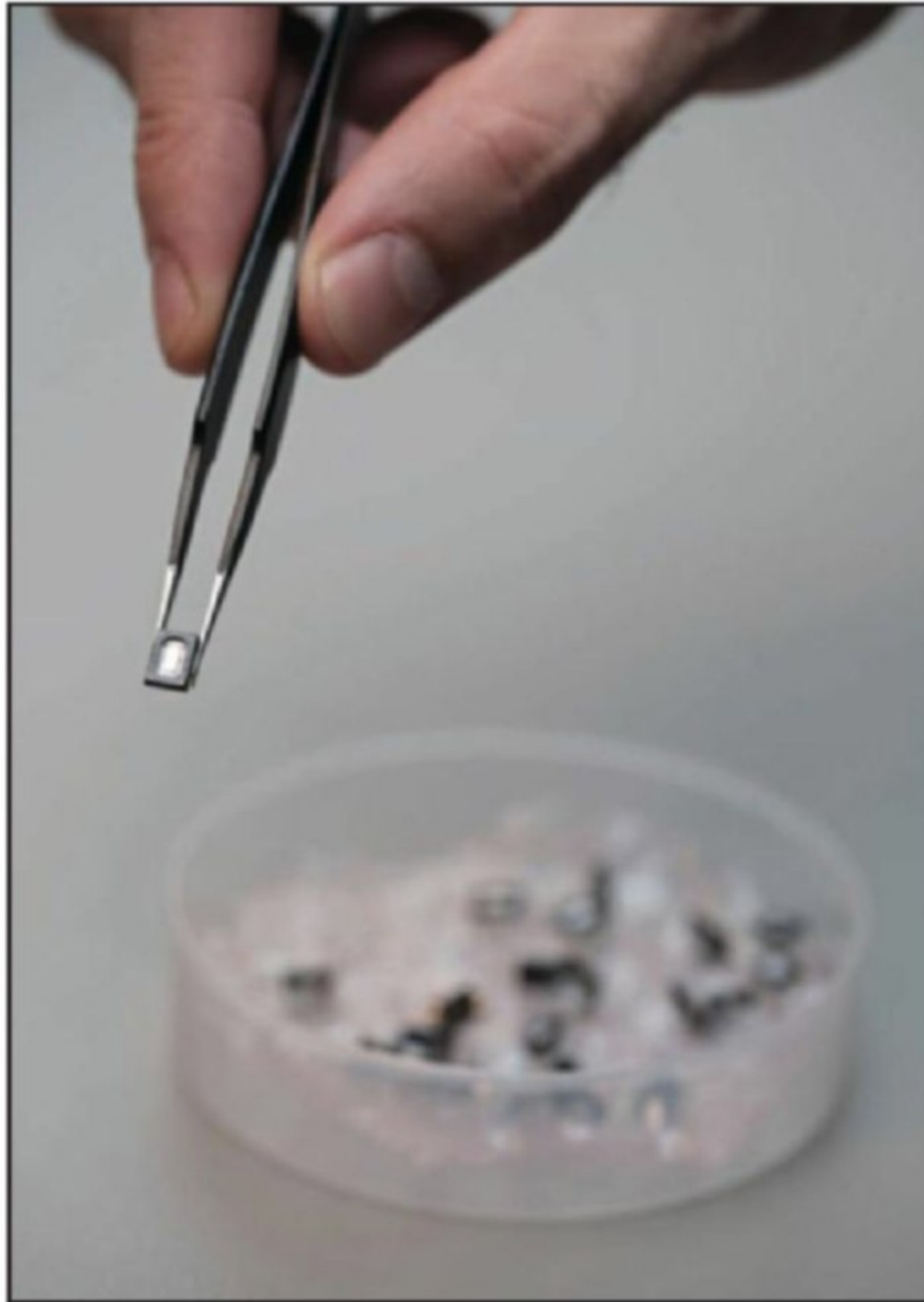
Israel-based Audio Pixels has succeeded in showing functional prototypes of its MEMS loudspeaker and is now focused on advancing aspects related to mass production and commercialization of MEMS loudspeakers. Audio Pixels pursued an entirely new technique to generate sound waves directly from a digital audio stream and the company holds innovative patents in the fields of electromechanical structures, pressure generation, acoustic wave generation and control, signal processing, and packaging.

The silicon chip that can be used either as a stand-alone speaker or cascaded in any multiples of the same chip in order to achieve required performance specifications. This modular paradigm is comparable to "parametric speakers" such as phased arrays as well as simply using more transducers for increasing the dynamic range. Audio Pixels technique is called Digital Sound Reconstruction, a theory introduced by Homer Dudley at Bell labs in the late 1930s.

Audio Pixels and ICsense, a European ASIC design company, recently announced that they have entered into a comprehensive agreement directed to support the production needs of Audio Pixels' patented low-cost MEMS-based digital speakers.

Still, the most advanced initiative—now commercially available—is the one from Austrian company USound GmbH. In 2017, STMicroelectronics (ST) and USound announced a collaboration on the industrialization and production of the world's first miniature piezoelectric MEMS actuators for smart audio systems in portable devices. The patented

USound's current MEMS microspeaker generation already offers more than 50% extended battery life for wireless earbuds compared to systems with electro-dynamic speakers. The soon-to-be-available next-generation devices will integrate an innovative energy recovery audio amplifier that USound states will create the most advanced and compact micro audio system on the market.



microspeaker technology from USound aims to replace commonly used balanced-armature and electrodynamic receivers for handsets with a small piezo-MEMS actuator.

Manufactured using ST's industry-leading thin-film piezo-electric (TFP) technology, these actuators improve scalability and cost while assuring lower power consumption and heat dissipation in hearables and smartphones devices, without compromising audio quality.

Resources

Amazon.com, Inc., amazon.co.uk/music/unlimited/hd

Audiovisual and Integrated Experience Association (AVIXA), www.avixa.org/META | www.avixa.org/IOTA

"Global Home and Professional Studio Report," "Worldwide Home Audio Market Report," and "2019 Headphones Market Report," Futuresource Consulting, www.futuresource-consulting.com

M. Klasco and S. Tatarunis, "Microspeakers' Anatomy—How to Design Audio Systems with Very Small Drivers, *Voice Coil*, May 2010
<https://www.audioxpress.com/article/microspeakers-anatomy-how-to-design-audio-systems-with-very-small-drivers>

"Pro Speakers Market - Global Outlook and Forecast 2019-2024," Arizton Advisory Service, <https://www.arizton.com/market-reports/pro-speaker-market>

Sony Corp. of America, www.sony.net/360RA

xMEMS Labs, <http://xmems.com>

These extremely small speakers, the thinnest in the world, also enable wearable applications in over-the-ear headphones, or Augmented-Reality/Virtual-Reality (AR/VR) headsets.

The MEMS microspeakers are simpler to manufacture, more reliable, and more economical in high volumes. The new devices also eliminate the usual trade-offs between magnet size, air volume, and sound quality that complicate both the design and integration of traditional, conventional electro-mechanical miniature speakers that contain a magnet and balanced-armature mechanism.

USound's current MEMS microspeaker generation already offers more than 50% extended battery life for wireless earbuds compared to systems with electro-dynamic speakers. USound's MEMS speakers are being put to real-world tests in application designs such as the Proteus speaker array and the Fauna audio glasses, a wearable design. The Proteus was the first concept designed by USound to demonstrate the possibilities of its MEMS devices, creating a fully-fledged driver made from 80 microspeakers.

USound's MEMS silicon microspeakers technology is based on a membrane mounted on a number of cantilevers built from PZT (lead zirconate titanate). When voltage is applied to this material, the crystals in the PZT expand or shrink making the membrane vibrate. Multiple cantilevers create a translation of the extension and shrinking, pushing a piston up and down. This replaces the traditional coil moving in a magnetic field as found in conventional speakers. Finally, mounting a membrane and a plate closes the acoustic shortcut and increases the effective area, thus enabling extended bandwidth sound generation.

With this approach, less heat is produced as well as less sound distortions due to fewer moving parts, creating an overall more robust system—in contrast to conventional speaker systems.

Another example of a MEMS loudspeaker research comes from the German Fraunhofer Institute for Digital Media Technology, which has already shown a new MEMS piezoelectric speaker with an area of just 4 mm x 4 mm and a thickness of just 0.4 mm. This project combines with intelligent signal processing and is targeting essentially new headphones, hearables, and hearing aids.

More recently, xMEMS Labs, a MEMS solutions startup headquartered in Los Altos, CA, announced a novel MEMS speaker for earphone applications that showcases not only the benefits of MEMS technologies with small size and low power consumption, but also a scalable design that will enable applications in many markets. The company claims to have developed a complete MEMS process that reduces manufacturing complexities. 

Stetron

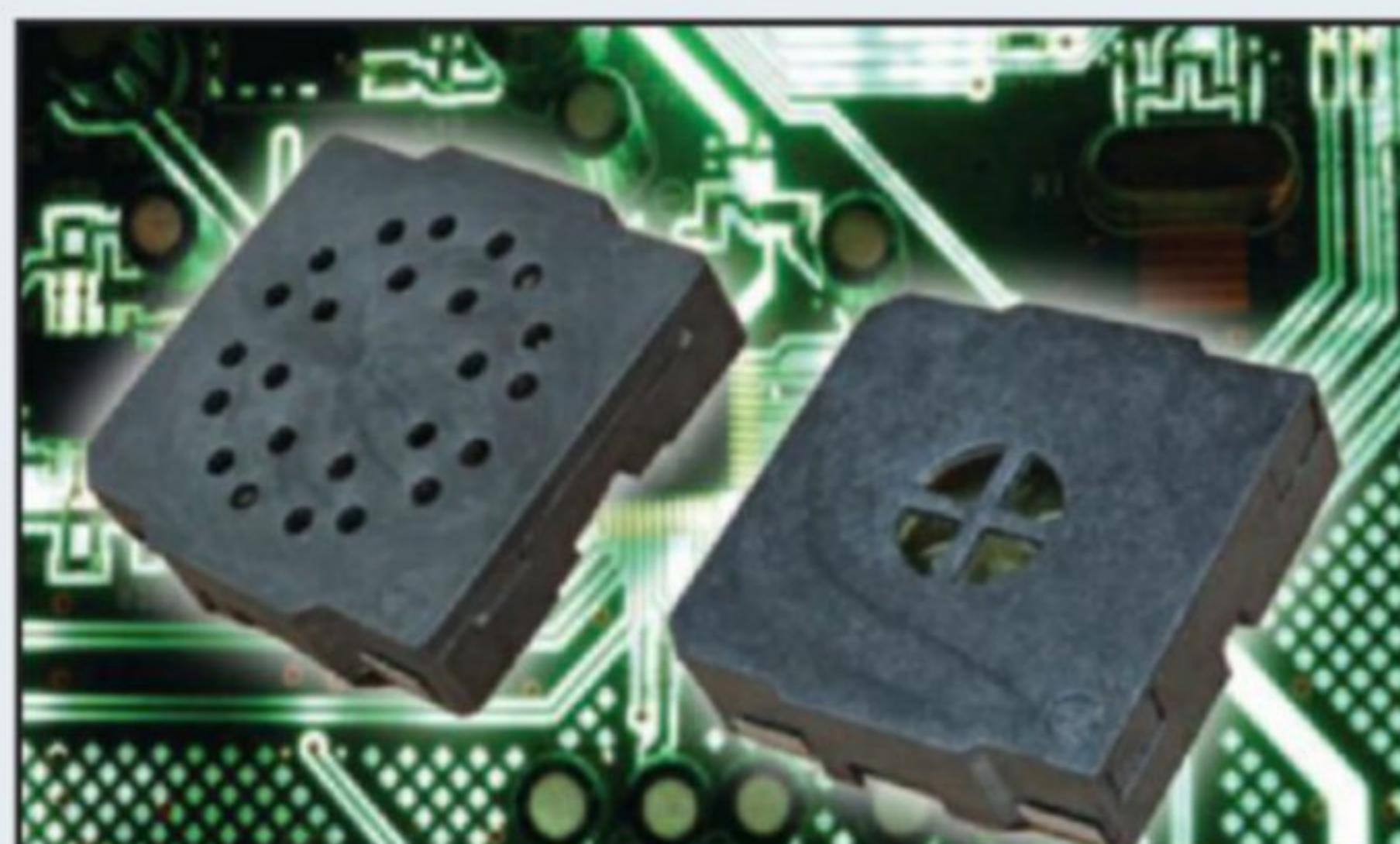
Need Small, Low-Power, Water-Resistant Speakers?

Mobile CE devices, two-way radios, telematics, Bluetooth devices, and so forth—there are many possible applications where a water-resistant speaker is needed. For that purpose, Stetron has developed a new low-profile 36 mm Water Resistant Speaker (Part number U0036004NM22AR) featuring low total harmonic distortion (THD), high output sound pressure level (SPL), high power handling capability and low F_0 .

As a specialist manufacturer of custom audio solutions, Stetron has a demonstrated expertise in optimizing engineered components including microphones (ECMs), dynamic receivers and loudspeakers. And Stetron's Asian-based manufacturing coupled with North American engineering, allows the Buffalo, NY, company to deliver technically advanced products at competitive prices.

As another example of the company's ability to create specialized solutions, Stetron announced the development of a 520 Hz low-frequency miniature loudspeaker designed especially for the Fire Alarm and Protection Industry. Featuring a slim design with a Mylar cone, which renders it water resistant, the Stetron D0057008NM127AR low-frequency driver was designed to be used in UL217 approved products, complying with all the performance requirements stipulated in section 84 of this standard.

This low-profile driver is able to achieve high SPL, with a small footprint of only 2.25" (57 mm), and is able to respond to 520 Hz. It also requires a very low current, making it suitable for battery-powered devices. Despite its physical size, this 57 mm × 17.5 mm



Microspeakers designed for mobile applications



The new Stetron U0036004NM22AR water resistant speaker

driver has a high output SPL of minimum 91.2 dB measured at 1 W/1 m and a rated input power of 2 W RMS.

Recently, Stetron also introduced several new surface-mount microspeakers designed for portable or hand-held product applications as well as products exposed to harsh temperatures.

Measuring just 13 mm × 13 mm, 15 mm × 15 mm, and 20 mm × 20 mm, all the new Stetron microspeakers are wideband, low profile, surface mount, and rugged, able to generate high output SPL levels.

The microspeakers—Part numbers s1313008sm40cmr, s15150087sm40cmr, and s2020008sm45cmr—are able to sustain high temperatures up to 85°C and low temperatures of -40°C.

Established in 1965, Stetron offers specialized speaker drivers in sizes from 10 mm to 150 mm and dynamic receivers, with high levels of consistency, manufactured on fully automated lines. Stetron is also able to offer engineering design services and create custom loudspeaker enclosures.

www.stetron.com

TeXtreme

Thin-Ply Carbon Diaphragms

Thin-Ply Carbon Diaphragms (TPCD) is a new category of loudspeaker diaphragms enabling clarity, smoothness and sensitivity by combining pistonic motion with controlled break-up. The vision is to engineer speaker diaphragms that sound like you want them to sound. The secret is the ability to engineer the diaphragm's behavior.

The ideal speaker cone should be stiff, light, and well damped. In reality, that is a difficult combination to achieve, typically forcing cone solutions to either prioritize pistonic motion but accepting uncontrolled break-up (hard cones); or prioritizing damping but accepting diaphragm break-up in the audible range (soft cones).

TPCD enables pistonic motion with controlled break-up, offering a combination of clarity, smoothness, and sensitivity. Clarity and sensitivity are enabled through the combination of high stiffness and ultralight weight, providing pistonic motion at a wide frequency range. Natural and smooth sound is enabled by controlling the break-up behavior through the concept of distributed break-up. Distributed break-up is achieved by eliminating the symmetric and big break-up modes, replacing them with smaller, local, and distributed break-up modes. That means that big peaks and dips of the frequency response are removed.



Not only are the break-up modes distributed, the way they appear can be optimized and controlled. By varying the way the thin-ply carbon layers are arranged, the properties and behavior of the diaphragm at different frequencies can be optimized. Much like the tone of a guitar string can be tuned by adjusting the tension, the TPCD can be tuned by adjusting the combination of fiber angles. That means that the TPCD diaphragm can be engineered for a specific and desired behavior and sound signature—the TPCD diaphragm can be engineered to sound like you want it to sound.

The vast flexibility of the TPCD technology enables it to be used for a number of cone and dome types, for instance tweeters, midranges, full ranges, woofers, compression drivers, headphones and microspeakers.

www.textreme.com/markets/audio

CUI

Waterproof IP67 Micro Speakers

CUI's Audio Group from Tualatin, OR, a subsidiary of CUI Global Inc., was founded in 1989 and is an electronic components manufacturer specializing in a diverse range of product technologies. The company's interconnect, audio, motion and thermal management groups provide solutions across a range of market segments, including specialized speakers and transducers.

The company launched several waterproof microspeaker models, with Ingress Protection (IP) ratings of IP67, to its CMS product family. Housed in compact packages measuring as small as 15 mm x 11 mm with profiles as low as 2.5 mm, these waterproof speakers are ideal for portable and handheld electronic devices subject to the moisture and environmental contaminants found in industrial and outdoor applications.

The rectangular frame models feature sound pressure levels (SPLs) ranging from 90 to 93 dB at 0.1 m, nominal inputs of 0.7 W, resonance frequencies from 550 Hz to 800 Hz, and impedance ratings of 6 Ω or 8 Ω . All series further offer spring



contact, solder pad, or wire lead mounting styles and are constructed with neodymium magnets and mylar cones for added durability.

The CMS series speakers are now available in spring contact, solder pad, and wire lead mounting options with prices starting at \$1.22 per unit at 500 pieces through distribution (OEM pricing available under request).

www.cui.com/waterproof-speakers

Earthquake Sound

The Cinénova Clarté LCR

For decades, Earthquake Sound has attended numerous trade shows. While at the shows, Joseph Sahyoun, founder and president of Earthquake Sound, takes the time to look at some of the products on the market and he noticed a pattern when it came to home theater Left, Center, Right (LCR) speakers or more specifically ribbon LCRs.

Several manufacturers made unfounded and outlandish claims while asking astronomical prices for their ribbon LCRs. Sahyoun was disappointed with the performance and price of ribbon LCRs in the market and set out on what would be a several year-long development project to produce the ultimate ribbon LCR that ticks all the boxes without an outrageous price. Sahyoun has always believed that you should not have to be a millionaire to experience audio euphoria. After years of creating prototypes and testing, Sahyoun hit the jackpot and the Cinénova Clarté was born.

The Cinénova Clarté is Earthquake's newest addition to the infamous Cinénova family of high-fidelity products. Thanks to its no-expense-spared design, it features a combination of four 2" precision ribbon drivers and paired with three 5.25" carbon fiber mid-bass drivers. The end result is an LCR that has a genuine life-like sound reproduction that brings the concert or movie straight to any living room. The carbon fiber cone woofers provide a distortion-free mid-range while the ultra-fast ribbon tweeters add remarkable sound quality with pinpoint accuracy. The top-tier drivers are perfectly balanced by an elaborate second order crossover network design that allows the Clarté to achieve a truly flat response curve and seamlessly blend the audio between the drivers.



Each Cinénova Clarté LCR speaker uses four 2" precision ribbon tweeters and three 5.25" carbon fiber mid-bass drivers in a sonically neutral cabinet design with a 40 Hz to 20 kHz frequency response. The design is rated for 600 W maximum power handling at 8 Ω .

The Cinénova Clarté is available in both in-wall and on-wall versions that were designed to be vertically or horizontally mounted. This allows users to position the speakers for left, right, and center channels offering plenty of installation flexibility for new or pre-existing home audio systems. The drivers can be covered with a magnetically attached grille after the installation.

Depending on one's needs, multiple Clartés may also be arranged to create array setups for music or home theater systems. The in-wall version also features Earthquake's Turn-N-Lock Mechanism making installation quick and easy—using the provided template to cut the mounting hole, the installer only needs to connect the speaker wires, and turn the screws to secure the Clarté to the wall.

www.earthquakesound.com

Beyma

LEX Subwoofer Family Expansion

In 2017, Spanish renowned pro audio transducer manufacturer Beyma introduced its new 18LEX1600Nd, a long excursion, high power handling and low distortion 18" subwoofer. A truly unique product, the 18LEX1600Nd development included an optimized mechanical design combined with an optimized electro-acoustic design demanding extensive simulation, prototypes and measurements until the desired targets were achieved. And the industrialization phase drove the entire team to validate a consistent product manufactured in Beyma's semi-automated line.

The new 18LEX1600Nd has been designed for performing better in vented cabinets and loaded horn designs, and offer a top competitive weight/performance unit thanks to an optimized finite element analysis (FEA) neodymium magnet motor structure, which leverages the brand's exclusive Maltcross cooling technology.

The 18" subwoofer responds from 35 Hz to 1 kHz with a program power rating of 1600 W (AES) and 3200 W (program) at 8 Ω . More importantly, its combination of Mechanical Mirror Suspension System (MMSS) technology, exclusive Neck Coupling Reinforcement (NCR) membrane, double silicon spider, 4" DUO double layer in/out copper voice coil, and aluminum demodulating ring, provide the 18LEX1600Nd with an extended controlled displacement ($X_{max} \pm 14.5$ mm), and 65 mm peak-to-peak excursion before damage, optimized for direct radiation and band-pass subwoofer applications. All while being extremely competitive in price.

After the success of the 18LEX1600Nd, Beyma realized that it had to take advantage of the features of this speaker, to expand the family, and benefit from this development and features.

With this philosophy Beyma's engineering team developed the 18LEX1000Nd and 15LEX1000Nd subwoofers, following the demand for a performance closely-matched to the 18LEX1600Nd, but with a lower power capacity. Currently, 1000 W subwoofers are a standard in the industry and they are widely used, but Beyma wanted to offer something different from the speakers already found in the market.

Apart from the power handling and excursion capabilities, the other main key target for the LEX family is to lower the final system's overall weight. That is why the design of the new subwoofers was based on a 3.5" motor structure, which thanks



The concept and design of Beyma's successful 18LEX1600Nd subwoofer inspired its engineering team to expand the research toward even more competitive weight/performance ratios.

to the Beyma-patented Maltcross technology, is able to deliver the performance and power handling of a 4" coil, but using a 3.5" coil. This makes everything more compact, generating an improvement in weight and price.

The Maltcross cooling technology—developed several years ago by Beyma—enables the subwoofers to handle more power with a smaller coil diameter, so the designers can think in smaller coil designs with a smaller motor structure, reducing the weight but increasing the sensitivity and also keeping an adequate power handling for a target application.

In the development of new Beyma LEX subwoofers, apart from the traditional Thiele-Small (T-S) parameters in the small signal domain, a lot of the design efforts focused on the large signal domain where the nonlinearities of these drivers will appear. While T-S parameters are standard and are measured with "comfortable" conditions for the speaker, in the "real world" of a subwoofer application, it's a whole different story. Beyma's engineers measured and exposed the drivers to different signals and conditions to confirm the parameters and performance of the speakers, looking for consistency of the basic or fundamental parameters during its life and in the long term. The final results are speakers with greater linearity and consistency over time.

www.beyma.com



The expanded LEX subwoofer family offers an improvement in weight and price, increasing the sensitivity and also keeping an adequate power handling for the target applications.

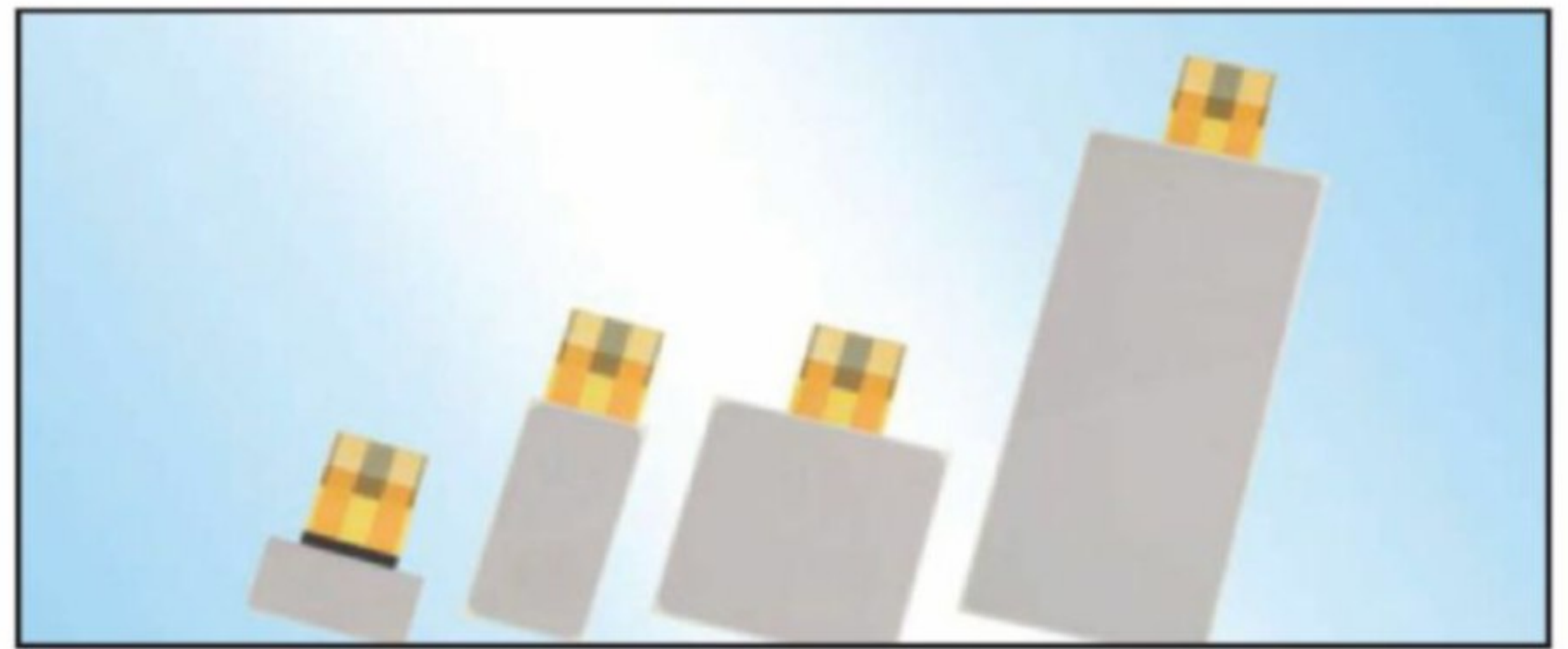
TDK

PiezoListen Ultra-Thin Piezo Speakers with a Wider Dynamic Range

TDK Corp. has expanded its line-up of ultra-thin PiezoListen speakers with new types, featuring a wider dynamic range and higher sound pressure level (SPL) in the low-frequency range, even at voltages up to 24 V_{pp}. The PiezoListen series of piezo speakers now includes the PHUA6630 wide dynamic range speaker with an operating frequency range of 400 Hz to 20 kHz. This product has a footprint of 66 mm × 30 mm and complements the existing PHUA3030 speaker, which was introduced in May 2019. Also new in the series are two new sound positioning speakers with an operating frequency range of 1000 Hz to 20 kHz.

The miniature PHUA2010 features dimensions of just 20 mm × 10 mm, while the PHUA3015 speaker measures 30 mm × 15 mm. Samples are available for all types.

With a thickness as low as 0.49 mm the PiezoListen type PHUA3030 is the world's thinnest speaker that is able to cover a wide frequency range from as low as 400 Hz up to 20 kHz. Thanks to its form factor, PiezoListen can turn nearly any surface such as a display screen into a speaker. In addition, it is easy to integrate into existing applications without sound emission holes, enabling



flexible acoustic solutions that require no changes to the exterior designs.

The development of the low-profile, high-displacement piezoelectric speakers was made possible by TDK's patented multilayer and material technologies. As the demand for advanced multimedia continues to grow, TDK will enable completely new integrated acoustic solutions and thus contribute to the evolution of application design with its PiezoListen range. Applications such as speakers for televisions, tablets, notebooks, smartphones, and other devices, as well as positioning speakers in surround systems benefit from this technology that is able to produce sound from nearly any display or surface.

www.tdk.com

Materion Truextent

BEX4008/4016 & Headphone Drivers

Materion's Truextent material is a premium acoustic beryllium foil used to make domes and cones for high-end loudspeakers and headphones. Recognized around the world by OEMs and ODMs, Truextent foil creates superior speaker domes and cones for crystal-clear sound reproduction.

The Truextent BeX4008/4016 precision replacement diaphragm is engineered to fit a variety of large-format compression drivers, including JBL and Radian models. Each diaphragm provides high efficiency and clarity of sound with extended bandwidth compared to the factory original or other aftermarket diaphragms. All Truextent BeX diaphragms deliver superior reliability, repeatability and value.

The diaphragm domes are formed from Genuine Truextent brand Acoustic Beryllium bonded to pure aluminum edgewound voice coils with a Kapton former utilizing proprietary bonding methods.

Truextent Acoustic Beryllium is the lightest and stiffest of all commercially available dome materials giving the diaphragm a low overall moving mass, greatly improving responsiveness, while remaining rigid over a wide frequency range. Additionally the BeX has the highest speed of sound through the dome, and thus the highest break-up mode.

The Truextent BeX suspension is constructed from PEEK



Materion's Truextent Beryllium foil is used by Focal in its top-of-the-range Utopia headphone diaphragm.

rather than a one-piece metal dome and surround, enhancing the reliability and eliminating metal fatigue that can destroy traditional driver products. The PEEK surround is designed to provide excellent damping for smooth low distortion linear output with excellent transient response.

Headphone Drivers

Beyond speaker drivers assemblies for loudspeakers Materion is also creating domes starting at 6 mm drivers and going up to 50 mm drivers for in ear monitors and headphones. This means manufacturers can still have the increased durability of grain

structure, higher damping, and an extra usable octave of resonance free high frequency sound on the go.

Providing a superior material option for the highest end of the market, Materion's Truextent Beryllium foil is used by Focal in its Utopia headphone diaphragm. Using Beryllium for acoustic components minimizes audible ringing and distortion heard from traditional metal diaphragms (e.g., titanium or aluminum). Beryllium enables an extra octave of useable, non-resonant high-frequency energy—pushing breakup modes beyond the audible band.

<http://truextent.com>



MISCO

Bold North Audio Expands in 2020

Earlier in 2019, US speaker manufacturer MISCO unveiled a new version of a legendary driver. The new MS10-W, marketed under the new MISCO brand, Bold North Audio, is effectively a direct replacement for the famous white driver on Yamaha's long-discontinued NS10 studio monitors! The MS10-W includes a lightweight white flat profile seamed paper cone with a white paper convex dust cap, and compliance is provided by a 12 mm wide coated cloth surround and a 3.25" diameter cupped cloth spider.

While other companies have attempted to make drop-in replacement woofers with white cones, MISCO is the first to precisely match the sonic character of the original Yamaha woofer. For the MS10-W, the MISCO engineering team researched all aspects of the original woofer and materials, choosing to use a seamed cone to match the performance of the original Yamaha (while other companies previously designing "direct-replacements" for the NS-10's did not) and the Klippel measurements have demonstrated that this was critical to carefully match the original driver's behavior and understand how to closely match the results in actual auditions.

MISCO's goal was not to improve upon the NS-10's mid-woofer performance but to exactly match its sound better than any manufacturer has done. The target was a drop-in replacement driver for the NS-10M Studio, the product iteration most often found in recording studios. The proof is in the measurements and the listening. The Bold North Audio (BNA) replacement driver is identical to the Yamaha original.



The Bold North Audio MS10-W is an exact match to the NS-10's mid-woofer performance and a drop-in replacement driver for the still extremely popular NS-10M Studio monitors.



New products from Bold North Audio are scheduled for launch in 2020.

Bold Expansion

Now, MISCO has announced that it would add several SKUs to the high-performance BNA brand in 2020. BNA was created to meet the needs of branded high-end hi-fi speaker system manufacturers, pro audio companies, recording studios, and advanced DIY hobbyists. All BNA products are Klippel Verified (designed and 100% tested with Klippel equipment) and assembled in MISCO's St. Paul, MN, factory.

The current BNA MS10-W is assembled in MISCO's factory under the supervision of the engineers who designed it. Every single production unit is tested using Klippel quality control apparatus to ensure exceptional unit-to-unit consistency. The MS10-W earned "Klippel Verified" status, which indicates the state-of-the-art Klippel equipment has been used from design through to end-of-line QC.

The MS10-W is packaged and sold individually, but MISCO highly recommends that NS-10 owners replace both mid-woofers at the same time. Driver suspensions and other parameters change over time such that a new production driver cannot match the sound of an older driver. For the best imaging and sound staging it is essential that left and right speakers be as closely matched as possible. Replacing both mid-woofers with MS10-Ws, in essence, brings the speaker system back to the sound of a new pair of NS-10s (assuming the tweeters and crossovers are working correctly).

New BNA products planned for early 2020 release are a 1.1" (29 mm) soft dome tweeter, a 6.5" (16.5 cm) high-excursion, low distortion XBL2 mid/woofer, a sonic-match replacement tweeter for the Yamaha NS-10 studio monitor, and a studio monitor speaker system. Product details and pricing will be revealed in early 2020.

www.miscospeakers.com

Measuring Amplifier Harmonic Distortion Below -140 dBc

At lower frequencies (<100 kHz), higher speed op-amps and fully differential amplifiers (FDAs) become essentially unmeasurable using typical passive filter and spectrum analyzer techniques. Recognizing that the spurious add as little voltage sine waves, an innovative technique was developed to allow measurements below even -140 dBc to be made from 1 kHz to 1 MHz.

By
Michael Steffes
(United States)

Typical approaches try to provide a no harmonic input test signal then remove the fundamental from the device under test (DUT) output spectrum before applying it to a spectrum analyzer using passive filters. Imagine running an op-amp in an inverting gain of -1 condition and applying a test input frequency with no filtering. If that original input signal is then passed around the inverting DUT and summed in an inverting summing stage, the original input signal (and its harmonics coming through the DUT) can be cancelled at the summing junction—leaving only the remaining DUT-generated harmonics.

Critical to this is treating the harmonic addition as a V_{pp} addition and not a power addition. Then, with fine scale gain matching and a tuneable all-pass

passive filter in the bypass path, the fundamental being applied can be driven to zero at the summing amplifier output along with the source harmonics. At lower frequencies, the high-frequency amplifiers being tested are very flat on gain response and are linear phase (constant delay).

Figure 1 shows the full test set-up, including the summing amplifier also implementing the second-order low-pass poles of a third-order Butterworth filter at 5 Mhz (which is shown in Figure 8 of the article written by Xavier Ramus).^[1] One error in Figure 1 is the impedance feeding out of the DUT into the summing junction should be 500 Ω to get the reported 9.5 dB residue gain in the filter stage (see Figure 10 of Ramus' article).^[1]

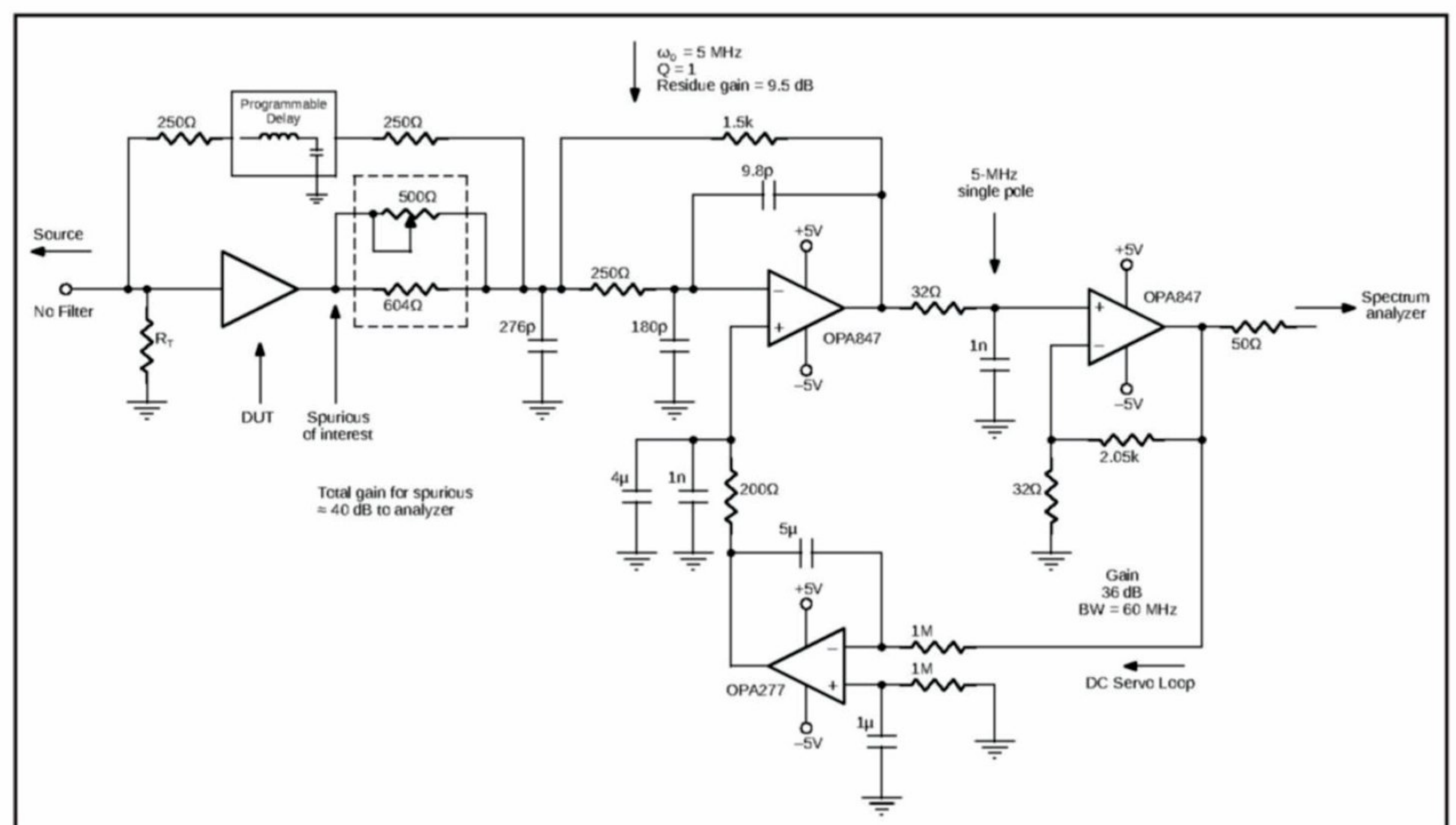
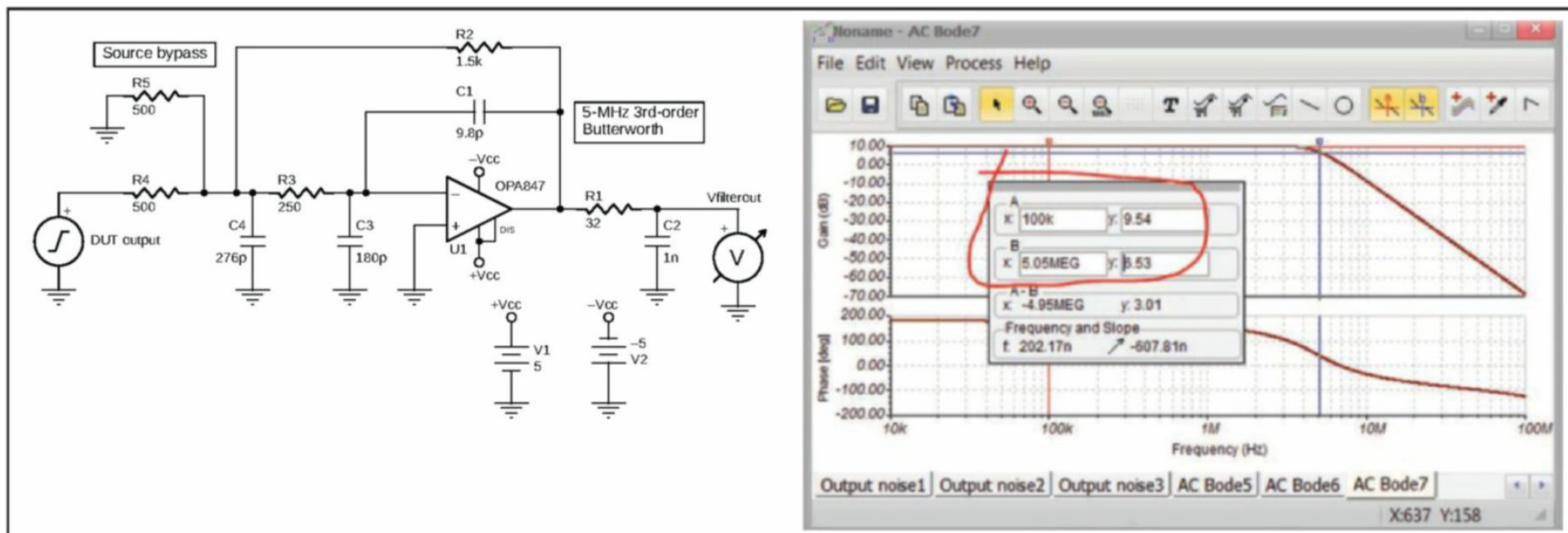


Figure 1: This is the source spectrum cancellation approach to measuring very low device under test (DUT) harmonic distortion.



The Key

The key to this approach is to present a “perfect” inverting summing operation in the “residue” amplifier. Using the 0.85 nV/√Hz input noise, 3.9 GHz gain bandwidth product, Texas Instruments OPA847^[2] achieves nearly perfect summing in the <3 MHz intended measurement range (third harmonic for a 1 Mhz test input). Considering only the input test fundamental frequency, adjusting both summing amplitude and propagation delay is essentially getting current cancellation for that test frequency (and the source harmonics) at the 276 pF summing node. The residual signal passed on is then only the added harmonics produced by the DUT.

Once we get the fundamental and its harmonics cancelled at the summing junction, the rest of the test circuit includes:

1. Implementing a third-order 5 MHz Butterworth filter to limit the broadband noise heading toward the spectrum analyzer.
2. The DUT spurious gets a gain of -3 V/V in the active filter stage. That 9.5 dB gain gets another 36 dB gain in a non-inverting OPA847 output stage.
3. The output 50 Ω matching resistor inserts a 6 dB loss to the spectrum analyzer for a net gain of 39.5 dB for the DUT harmonics.
4. With this much gain, a low-frequency servo loop (OPA277 feedback) centers up the final OPA847 output on ground with a high-pass pole far below the intended measurement frequency range.

There are some added nuances in the multiple feedback (MFB) active filter stage. While the 500 Ω DUT output resistor sets the gain for the DUT harmonics, it is the parallel combination of the two 500 Ω's summing into this node that is used for the active filter RC solution. The nominal frequency

response from the DUT output to the input at the final OPA847 gain stage shows this is achieving a 9.5 dB gain with the desired 5 Mhz F_{-3 dB}.

Normally, you would not apply a decompensated op-amp to a multiple feedback (MFB) design due to the direct feedback capacitor taking the noise gain to 1 at higher frequencies. The added C3 capacitor at the inverting node to ground in **Figure 2** shapes the higher frequency noise gain up to:

$$1 + (180/9.8) = 19.4 \text{ V/V}$$

at higher frequencies, keeping this implementation stable.^[3]

To extend this approach to even lower distortion levels (<-140 dBc), while the DUT is operating at an inverting gain of -1 V/V (noise gain of 2), adding another resistor to ground on its inverting node can keep the signal gain at -1 V/V while artificially increasing the noise gain—reducing the loop gain artificially raising the DUT output distortion terms. Once measured, those can be re-referenced to a simple -1 V/V condition by reducing those measured harmonic distortion terms by the difference between the actual noise gain (in decibels) and 6 dB. This approach is limited by the signal bandlimiting that will also be happening in the DUT. It would be best to limit this artificial noise gain increase to where the DUT bandwidth is no less than four times the expected maximum HD3 frequency to be measured. 

Figure 2: This schematic and accompanying graph show the active filter stage frequency response for the DUT output harmonics.

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[2] TI OPA847, “Wideband, Ultra-Low Noise, Voltage Feedback Op Amp with Shutdown,” Texas Instruments, <https://www.ti.com/lit/ds/symlink/opa847.pdf>.

[3] M. Steffes, “Design Methodology for MFB Filters in ADC Interface Application,” TI Application Note, February 2006, <http://www.ti.com/lit/an/sboa114/sboa114.pdf>.



Fresh From the Bench

Cambridge Audio Edge W Power Amplifier

Amplification to Live With

By
Oliver A. Masciarotte
(United States)

Photo 1: Cambridge Audio's Edge W power amplifier is remarkably detailed. The sound is finely balanced, with a musical depth and clarity that comes through with every note.

For a few weeks, two *audioXpress* reviewers, Oliver Masciarotte and Stuart Yaniger, tested Cambridge Audio's Edge series power amplifier, the Edge W, which the British manufacturer introduced in 2018 as part of its 50th anniversary celebrations. *audioXpress* submitted this design, which claimed to be the culmination of the brand's audio engineering expertise, to extensive listening and measurement sessions.

In years past, I had thought of Cambridge Audio as an uninspired CE manufacturer cranking out unassuming, mid-level kit. I knew it had a distinguished legacy, but the demos never got my juices flowing. In October of 2017, all that changed.

That fall, I stepped into Cambridge Audio's room at Rocky Mountain Audio Fest (RMAF) and was greeted by a bunch of eager hi-fi guys waiting to share a story. For my show coverage, I wrote that they had been working "on a new range of higher end electronics aimed at elevated performance. With a minimal signal path, intelligent biasing and no caps to degrade the audio, their prototype component line is a departure from their normally budget-priced offerings. The stereo power amp is joined by a modern, all-inputs-welcome preamp that also builds some fresh thinking into the design and layout. Paired with a set of Bowers & Wilkins 805 D3 stand mounters, it was the most lifelike presentation I've yet heard from these guys, very promising." Last year, that promise was fulfilled.

The Edge W is part of a triad of Edge-badged products—the Edge A, the Edge NQ, and the Edge W. The \$5,000 Edge A is an integrated amp, the \$4,000 Edge NQ is a network-enabled preamp, and the \$4,000 Edge W is purely a power amplifier (see **Photo 1**). One might guess the line's name derives from "the cutting edge" but, in fact, the products are named after Gordon Edge, a founder and the company's

Edge W

Cambridge Audio USA
1913 N. Milwaukee Ave., Chicago IL 60647
www.cambridgeaudio.com
Price: US \$4,000

Ancillary Equipment:

Speakers—PureAudioProject Trio15 Heil (\$4,495, www.pureaudioproject.com)
ProAc Studio SM100 (\$3,400, www.proac-loudspeakers.com)
Isolation—IsoAcoustics GAIA II (\$300, www.isoacoustics.com)
Source—Amarra Luxe 4 macOS (\$99, www.sonicstudio.com/amarra)
Conversion—Mytek Digital Brooklyn DAC
Amplification—Lindell AMPX (\$1,600, www.lindellaudio.se)
Cabling—VUE Digital VU-3 (USB), Soundstring GEN II Beta 2-22S (XLR), AntiCable Level 3.1 (speaker), Soundstring GEN II Digimax-18 (power)

first head of engineering. Edge, with Peter Lee, created the sleek, sexy, and innovative P40, a 20 W integrated amplifier that jump started the company. One of the aspects of the P40 that allowed for its slim packaging was the pioneering use of a toroidal transformer in a production audio product.

The Edge W's Topology

I asked the current Engineering Project Manager, Nick Brown, for some details about the Edge W's operation and construction. Assuming the product was some sort of advanced Class AB, I asked if the amp transitions from Class A to Class AB. For many listeners, an ability to deliver 2 or 3 Class A watts before it moves into the Class B regime is one of the key aspects of an amp's sonic signature. Brown replied, via e-mail, that it produces about 3 W into 8 Ω and 1.5 W into 4 Ω in pure Class A mode. The Edge W develops 100 W RMS into 8 Ω , and 200 W into 4 Ω . He went on to elaborate that, "one of the Class XA's major benefits is to reduce the crossover distortion even after the transition from Class A to AB operation, compared to a high-biased Class AB stage delivering similar Class A power level. This is because of the better inherent linearity of the output stage when it's working with normal Class AB current as opposed to high bias current to deliver more watts in Class A condition."

As it turned out, the Edge W leverages Cambridge Audio's XA technology, an in-house riff derived from work done for the company by amplifier "Yoda" Douglas Self. His XD or Crossover Displacement design actively displaces the crossover point away from the traditional zero crossing. The result? Less heat and power draw

that a pure Class A design and lower distortion than traditional Class AB.

Cambridge Audio boasts that there are only 14 components in the signal path (see **Photo 2**). That path is a combination of discreet and integrated components, which keeps the cost reasonable. Brown mentioned that, while the power output stage uses discrete output devices, the signal path does include an op-amp. A total of eight Sanken transistors per channel drive the output. The multi-layer circuit board employs a full ground plane, while the components populating it were all chosen for sound quality and optimal performance.

When asked for specifics on the XA topology, Brown started with the basics, stating that in traditional Class AB amps, crossover distortion happens each time the signal crosses zero amplitude. "With Class XA, the crossover point is shifted away from zero, and thus crossover distortion only happens for signals larger than a given threshold." In the Edge W, that threshold happens at about 3 W.

"In practical terms, Class XA consists of a tuned constant current generator added to the power stage. This extra current shifts the crossover point away from 0 V. Distortion is much more detectable by the human ear at low sound level, while 3 W can generate a discrete amount of sound pressure to 'hide' crossover distortion. Also, at normal listening levels, Class XA can be over 200% more efficient (compared to a) high-biased Class AB able to deliver the same Class A power," explained Brown.

Since I was told the amp is DC-coupled, in this modern age that implies a servo'd design. Cambridge Audio uses servos to inject an equal and opposite



Photo 2: Although minimal in its exterior design, the Edge W has some complex audio engineering inside. To remove the barriers between you and your music, Cambridge Audio engineers have minimized the number of components in the signal path.

amount of DC into the signal to cancel out any detected DC offset, rather than using blocking capacitors. This is significant for its effect on low-frequency purity and transient response. Brown explained that capacitors, "...as components, have the biggest detrimental impact on the sound." He said this is because not only do they have wide tolerances, but they are also microphonic. They transform mechanical vibration into unwanted electrical noise and modulation. Removing them results in a cleaner signal path less susceptible to distortion.

Harkening back to the P40, the Edge W uses not one but two, magnetically opposing toroidal transformers. Brown mentioned they "...also apply a silicon steel screen tape around each transformer to reduce stray fields, then any that is left is cancelled out by the opposing arrangement. The result is that far less

electromagnetic radiation interferes with the sensitive input stage of the amplifier." The effect works at all volume levels because each transformer is delivering the same current, so they are "in harmony," cancelling out each other's electromagnetic field throughout the whole power range.

The Edge W also utilizes a further third transformer for the preamp stage—the power amp and pre power supplies, all linear, are separate from one another. Though fuses are avoided in the audio path as they generally degrade performance, several fuses are used in the power supply circuitry as required by international safety regulations. The Edge W also integrates a full set of protection against DC offset, over-current, clipping and over-temperature. A relay will physically disconnect the speakers in case of a detected fault. With an oscilloscope attached to the speaker leads, I couldn't even get it to clip before wanting to run screaming from the room.

I mentioned earlier that cost savings was part of the mandate to keep the product affordable. The Edge was designed and engineered in the UK by Cambridge Audio's team of in-house engineers. Being one of the first audio businesses to invest in China in 1994, the system is assembled there, where Cambridge has a dedicated team overseeing engineering tasks, QC and best practices. The Edge W is built like the proverbial brick house, with clean lines and a forward-facing wrap of anodized dark gray metal possessing only an illuminated power switch and engraved logotype.

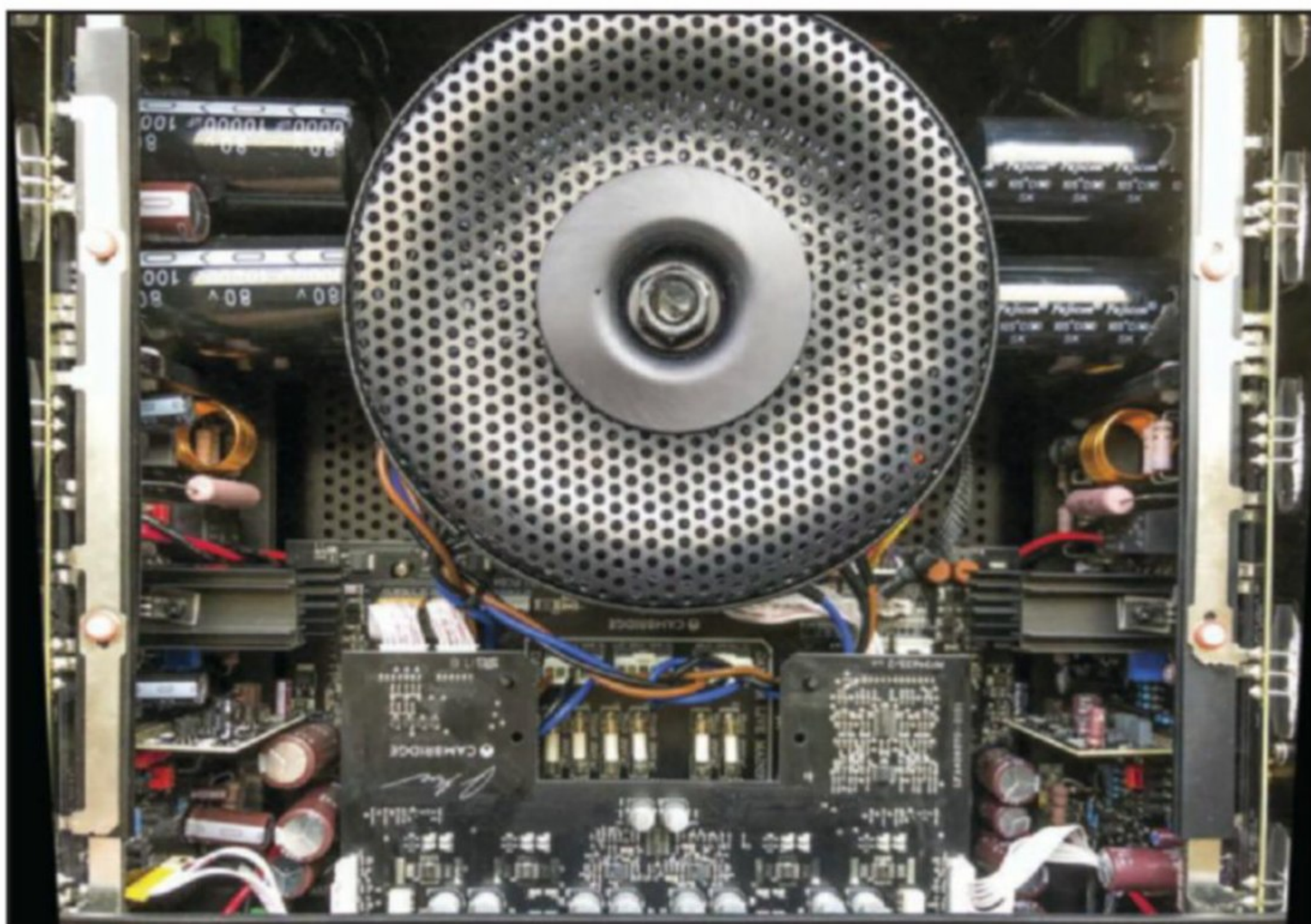


Photo 3: The Edge W has a very robust construction and sophisticated industrial design. (Photo courtesy of Stuart Yaniger and Cynthia Wenslow)

The Class XD Design

For those interested in the Class XD Design, here is a bit of information from the abstract of Patent GB2424137, which was awarded to Douglas Self:

"The crossover voltage of a Class B amplifier is displaced by sinking an extra current to the lower supply rail through a displacer circuit so that (the) upper output device... is always conducting at low input signal levels. The amplifier operates in Class A at low signal levels and in Class B at high signal levels. The displacer circuit may comprise a resistor, a constant current source, or a controlled current source. The output devices may be bipolar transistors, FETs, or thermionic valves. The crossover voltage may be displaced by 1 to 10 volts."

More information on Class XD design can be found in Douglas Self's *Audio Power Amplifier Design Handbook*, Focal Press/Elsevier 2009.

Listening to the Edge W

As my reference e22 Mark II DAC was at the factory for some special modifications, I relied on a straight path from my Mytek Brooklyn into the Edge W via balanced Soundstring GEN II Beta 2-22S cables. For comparison, I used two identical Soundstring GEN II Digimax-18 power cables for the Edge W and my Class A Lindell AMPX reference (see **Photo 3**). At the time of the review, my Level 3 speaker cables had reached the end of their service life, so I reached out to Paul Speltz at Anticables, who provided an upgraded set of Level 3.1 cables.

After a burn-in period, the new pair exhibited improved resolution and mo' betta bottom, so I felt I was ready to dig into the subjective portion of the review. The Trio15 Heils from PureAudioProject I reviewed last September were still in-house, awaiting a HORN1 upgrade, and my good friend Steve Severson lent me his ProAc SM100 stand mounters. Both are low distortion, neutrally voiced speakers, and the low efficiency ProAcs were a good test of the Edge W's beefiness. In their own way, both speakers were perfect for this evaluation.

All well and good, this clever XA displacer

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arrangement, but how does the Edge W actually sound? In short, like nothing. More exactly, like nothing is interfering between you and your music. The Edge W is a straight wire with gain, delivering performance that handily competes with budget busting competitors boasting five digit price tags.

For a dose of real-world acoustic environment and atmosphere, I turned to the Feenbrothers Play Dave Brubeck [Sound Liason: 352.8 kHz]. This live, one microphone recording, was made with a Josephson C700S feeding a Merging Technology Horus. The production is minimalist and the clean, spacious presentation was conveyed with an ease and depth simply not painted as well with lesser electronics.

Another supernaturally realistic recording comes from Morten Lindberg, who I finally got to meet at the recent 2019 Audio Engineering Society (AES) convention in New York. His Hoff Ensemble recording "Polarity" [2L: 352.8 kHz], like the Feenbrothers album mentioned earlier, is free of extraneous production. Through the Edge W, I was joyfully enveloped in an authentic, palpable and well delineated sense of place.

Whether it's the pile driver bottom of Visax's

"Monster" [TIDAL: 44.1 kHz], or Massive Attack's ominous synth bass in the track "Angel" [Qobuz: 44.1 kHz], Edge W's ferrous-free grip takes no prisoners. Listening to this amp, I simply cannot understand why anyone would want a capacitor or inductor sticking their nonlinear noses between listener and their tunes.

It's not just the low end where this more than 50 lbs. beast shines. The published bandwidth spec is less than 3 Hz to 80 kHz, ± 1 dB, and Stuart Yaniger's accompanying tests bear this out. One of my favorite tracks to experience a stripped down production cleanliness is J.S. Ondara's "Lebanon" from his *Tales of America* [Qobuz: 192 kHz]. Ondara's Tracy Chapman voice plus guitar, trap set and chorus, is a fun fest of folky flavors, served up without flaws by Cambridge Audio.

The production credits accompanying my DVD-Audio rip [WAMO: 192 kHz] of Linda Ronstadt's "What's New" reads like a who's who of audio production—Peter Asher as producer; Nelson Riddle lending his arranger and conductor chops; George Massenburg engineering and mixing at The Complex in Los Angeles, CA; and Doug Sax at his mastering desk. The Edge W effortlessly communicated Ronstadt's powerful voice, the velvet smoothness of Riddle's strings and Massenburg and Sax's tasteful helming.

The Verdict

The ProAcs, being studio monitors, are honesty personified except at the bottom half octave, and the Edge W is as uncolored and exemplary as those little two-ways. When comparing the AMPX to the Edge W, the latter throws a much wider soundstage, has more precise transient response, an airy extended top end and a much tighter, more extended low end. I can effortlessly drive the AMPX into tasteful clipping, not so with the Edge W. Both amps exhibit a low distortion clarity, dare I say sweetness; not a euphonic flattery but a truth I enjoy more than any "warm," even harmonic distortion signature. I love my AMPX but the Edge, she really got the edge!

I've seen and heard super-amps at shows, you know the ones. With few exceptions, they are European. They cost well upward of \$20,000, and they define the state-of-the-art in amplifier construction, if not excess. Giant tubes, giant meters, and giant casework with an equally gargantuan price tag. I do have a pair of \$10,000 (each) Class D amps in-house and, frankly, the Edge W flattens them without even noticing the bloody corpse in the rearview mirror. If you're on the hunt for upscale performance at an affordable price, and want to save money on AC and HVAC bills, then I cannot endorse the Edge W enough... simply highly recommended!

Keep reading as Stuart Yaniger puts the Edge W through his own set of listening tests.



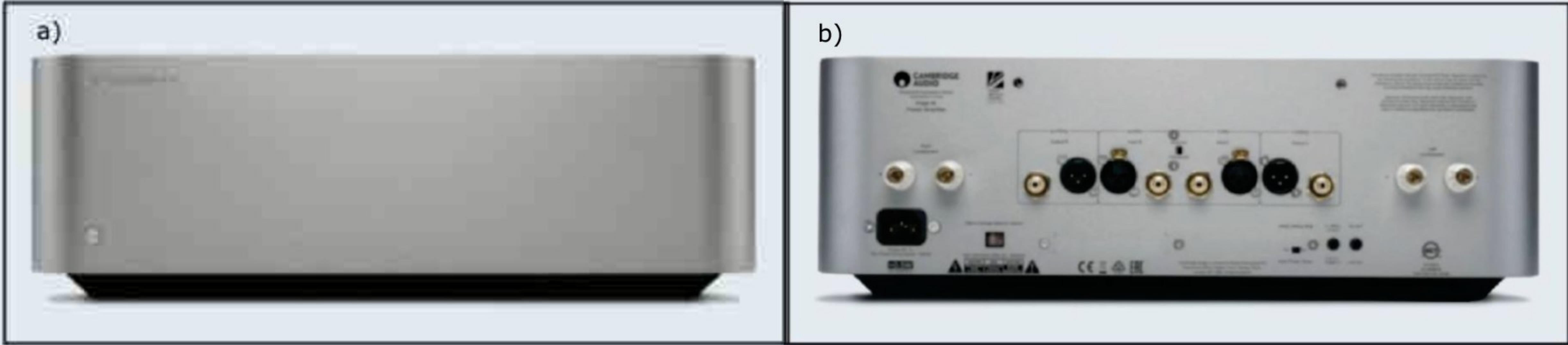
Photo 4: For my test setup I paired the Edge W, in combination with the Brooklyn DAC and Trio15 Heil open baffles, which offers an unsullied, slightly dry presentation.

About the Author

O. A. Masciarotte has spent 40 years immersed in tech, working worldwide on facilitation, marketing, and product development. His writings include *To Serve & Groove*, a book covering file-based music for the home and well over 100 articles. At present, Masciarotte is CMO and co-founder of MAAT, a developer of pro audio hardware and software solutions.

The Measurements

When it was his turn to test Cambridge Audio’s Edge series power amplifier, the Edge W, Stuart Yaniger discovered some very smart engineering.



Cambridge Audio’s Edge W power amplifier has a clean muscular look with a sleek new logo (a). The Edge W also has an added bias voltage to a traditional Class AB design, reducing distortion to inaudible levels. This unique configuration also produces less heat and is intended to make the entire system much more efficient.

By
Stuart Yaniger

Thanks to my millennial daughter, I have become aware of a genre of entertainment known as “unboxing videos.” For people older and/or even less hip than I am, it is exactly what it sounds like, someone receiving a product and going through the process of getting it out of its packaging. For reasons that escape me, YouTube videos of unboxing various items get millions of views.

If you have had an ambition to create an unboxing video and want it to run for over an hour, you need the Cambridge Audio Edge W power amplifier. I have never, in my 50 years doing audio electronics, seen packaging this elaborate. There is an outer box, which is cut open. This allows you to slip out an inner box with removable handles and clamps. With the handles removed, the inner box can be removed to reveal yet another box, with a large block of sculpted foam as the top. Remove the sculpted foam and you find... another box with a zippered fiber bag. Unzip the bag and there’s several more pieces of fiberboard and a silicone gasket. Remove them and you are finally in a position to liberate the amplifier from the lower box.

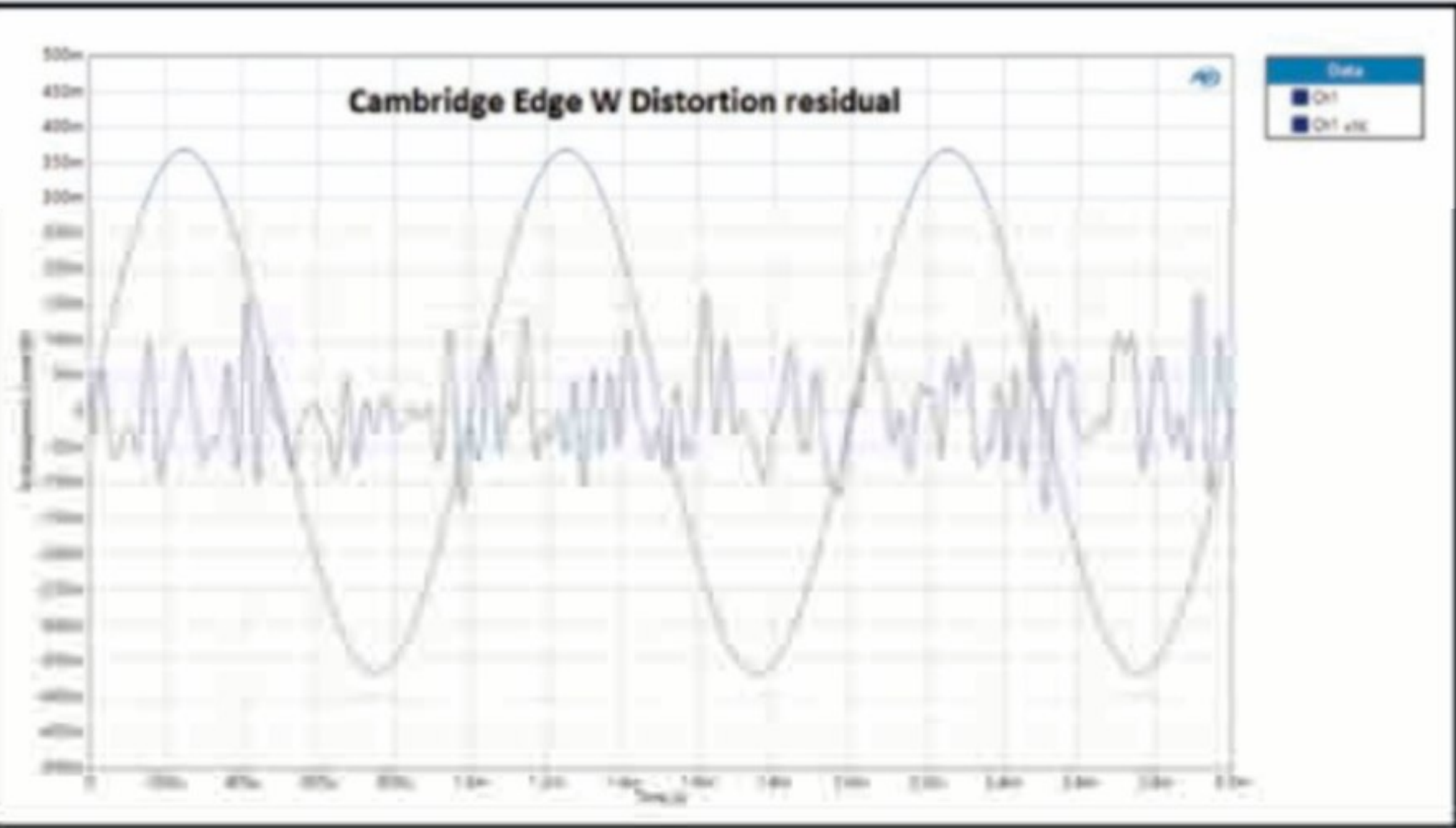


Figure 1: The waveform of a 250 mV sine wave shows no trace of crossover distortion. The residual (shown at $\times 1000$ magnification) is noise-dominated.

Suffice it to say that this beautiful piece of equipment is unlikely to suffer shipping damage. Nor did it in my case.

The Measurements

As usual, my measurements were performed with an Audio Precision APx-525 analyzer and various homemade dummy loads. When the Edge W is plugged in, it goes into Standby mode, where power consumption is a negligible 0.5 W or so. Depressing the power button starts the turn-on cycle, which caused the lamps and appliances in my lab to briefly dim; the power supply reservoir is clearly rather large!

I began the measurements by connecting 8 Ω dummy loads and letting the amplifier come up to operating temperature over an hour. After that time, I used an IR thermometer to check the temperature of the heatsinks (the hottest part of the amplifier), which was stable at about 52°C. I then ran the standard Fast Thermal Cycling (FTC)

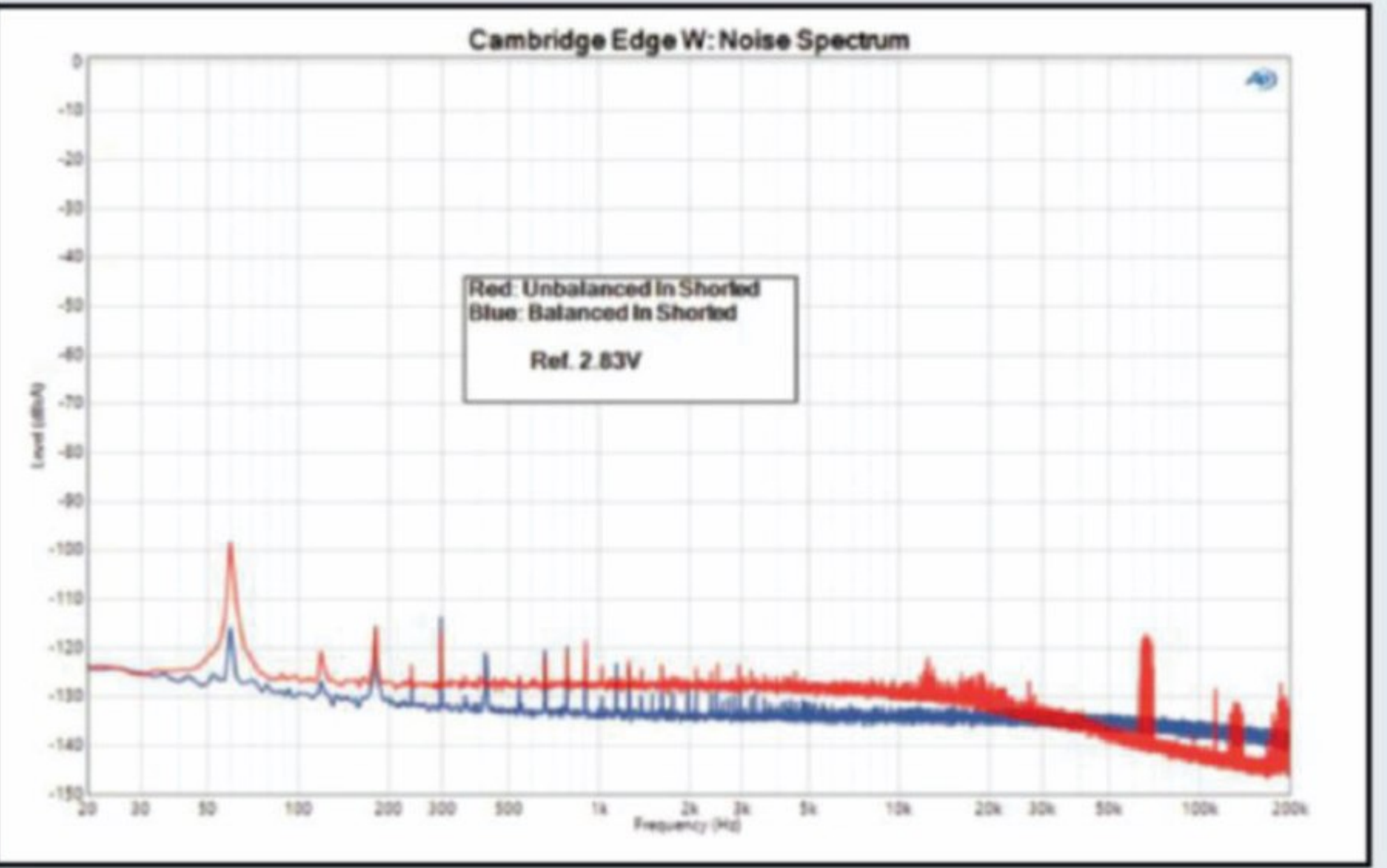


Figure 2: The spectrum of the amplifier’s output with a shorted input shows very low noise, with the worst case (60 Hz in unbalanced mode) being -99 dB down from the 1 W reference.

preconditioning cycle, with 33 W (1/3 rated power) from each channel at 1 kHz into the 8 Ω loads for an hour. The heatsink temperature rose to about 60°C, which is hot to the touch, but not to the dangerous point. In use, I would ensure that these amps get at least moderate ventilation, and the temperatures will stay within reason.

The rather small temperature shift between idle and one-third power suggests that the output stage is biased a bit rich. I connected a meter across one of the output stage emitter resistors to check this and found that the idle current was about 220 mA per device, with 50 V power rails. There are five output transistors of each polarity (10 total) per channel, so the total idle current is about 1.1 A for 110 W of idle dissipation per channel. No wonder this amp runs warm! That idle current implies Class A operation up to about 10 W into 8 Ω .

Looking at a 250 mV 1 kHz sinewave and the distortion residual (see **Figure 1**, distortion residual magnified 1000 times), it is evident that there's no visible crossover distortion, the residual being noise-dominated. A low-level sine wave like this is a very sensitive test for crossover artifacts. If it were present, crossover distortion would be indicated by spikes in the residual coinciding with the signal zero-crossing. Many of Cambridge Audio's amplifiers use an output configuration that it calls Class XD, where a current is injected into the output stage transistors' bases to displace the zero crossing, and if this is used here, it certainly does its job.

After preconditioning, I checked the basics in both balanced and unbalanced mode. In balanced mode, the voltage gain was 22.25 dB (or about 13 V/V). In unbalanced mode (switch selectable), the gain was 28.3 dB (or about 26 V/V). DC output offset voltages with inputs shorted were below my meter's ability to accurately measure. This suggests a design using some very low offset servos, which functioned flawlessly through the testing. Input impedance was 56 k Ω in unbalanced and 102 k Ω in balanced mode. Source impedance at 1 kHz and below measured 22.5 m Ω , rising to 55 m Ω at 20 kHz, equivalent to a damping factor of 145. These source impedances are low enough that frequency response variations due to the amplifier with loudspeaker loads will be absolutely insignificant.

The noise spectrum in both balanced and unbalanced mode with inputs shorted and referenced to 1 W into 8 Ω (2.83 V) is shown in

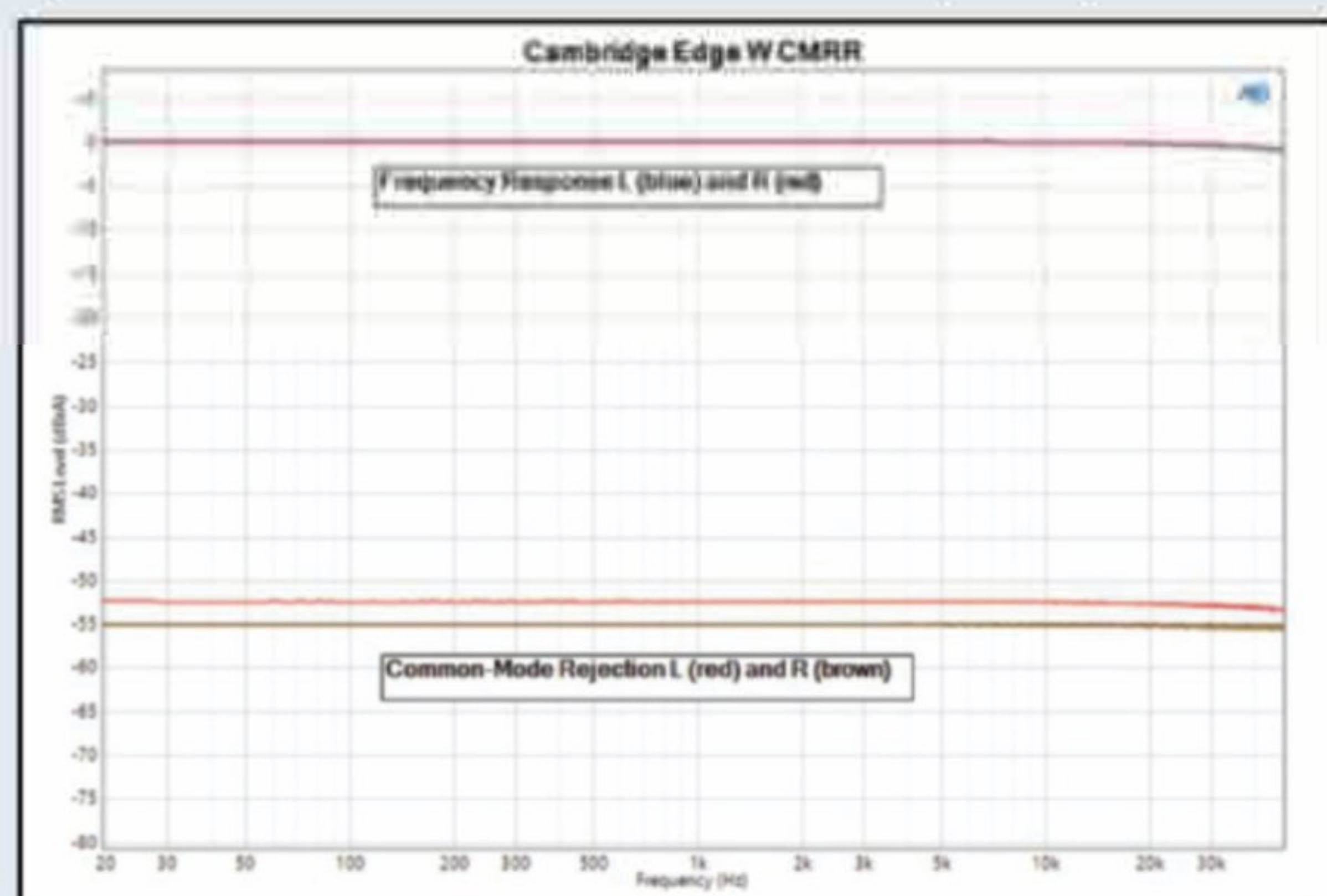


Figure 3: The Cambridge Edge W's frequency response is dead flat (upper curve), and the common mode rejection (CMR) in balanced mode referred to the input is better than -70 dB.

Figure 2. In unbalanced mode, the noise was dominated by a 60 Hz component at -99 dB. In an absolute sense, this is not a large number (32 μ V), but for an amplifier of this quality, it's slightly larger than expected, though of no significant audible consequence. The balanced mode shows nearly an order of magnitude lower hum.

This became a somewhat larger issue when the Audio Precision was used as a signal source: the 60 Hz component increased to -80 dB, which is potentially getting into the range where it could possibly be heard in a very quiet listening area with highly sensitive speakers. With my lab speakers (rated at 90 dB/2.83 V/m), the hum

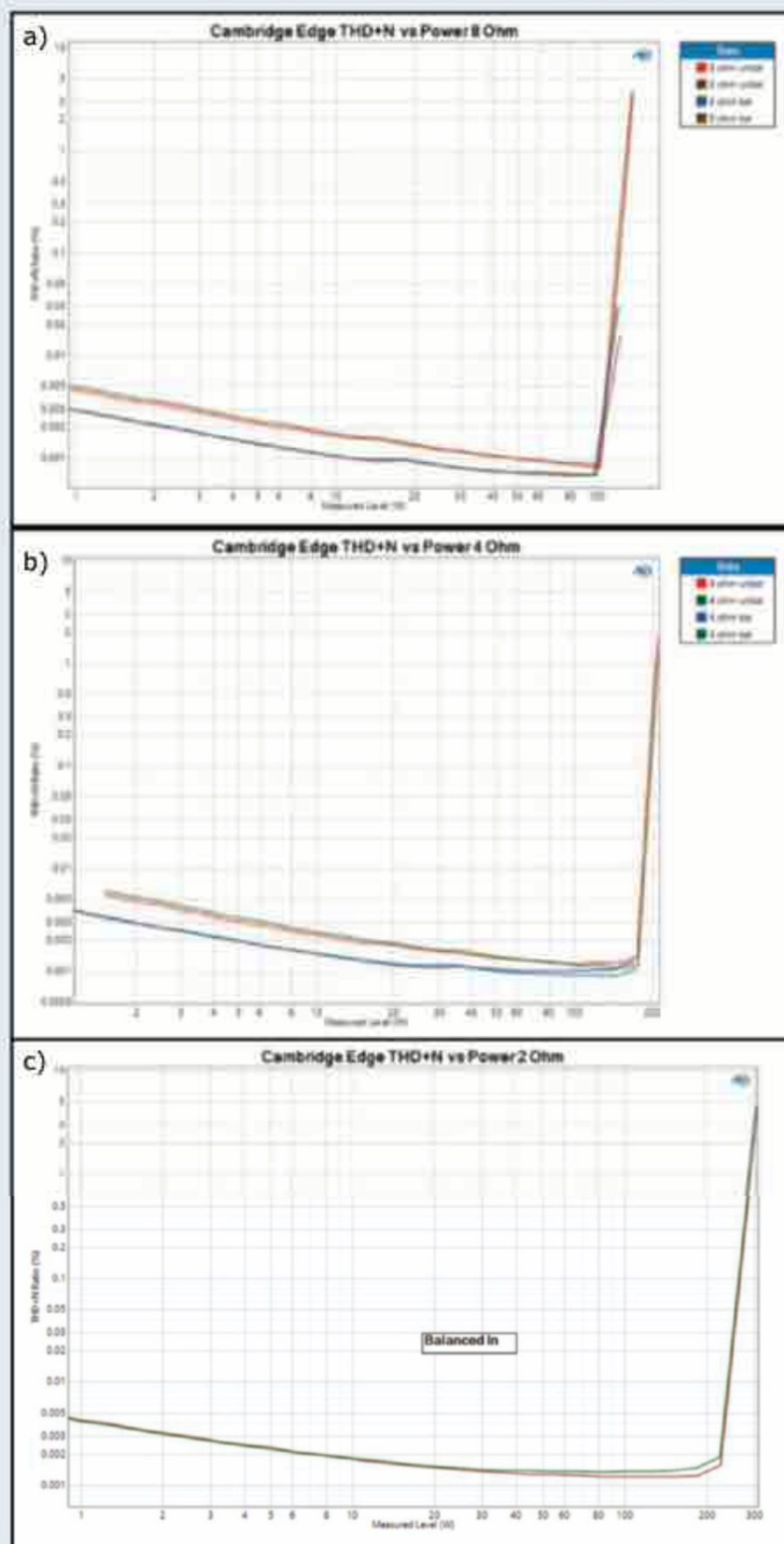


Figure 4: Distortion versus power at 8 Ω (a), 4 Ω (b), and 2 Ω (c) shows the Edge W comfortably meeting specification.

was absolutely inaudible. To be fair, this is not any more hum than is usually found in this class of power amplification, but it stuck out to me because the other measurements were so excellent that seeing one that's normal is a bit jarring!

I tried several alternative grounding schemes between the Edge W in unbalanced mode and the Audio Precision, including running them from separate branches of our house power, but could not get significant improvement. Inserting a Jensen JT11P-1 1:1 input transformer into the line between the Audio Precision and the

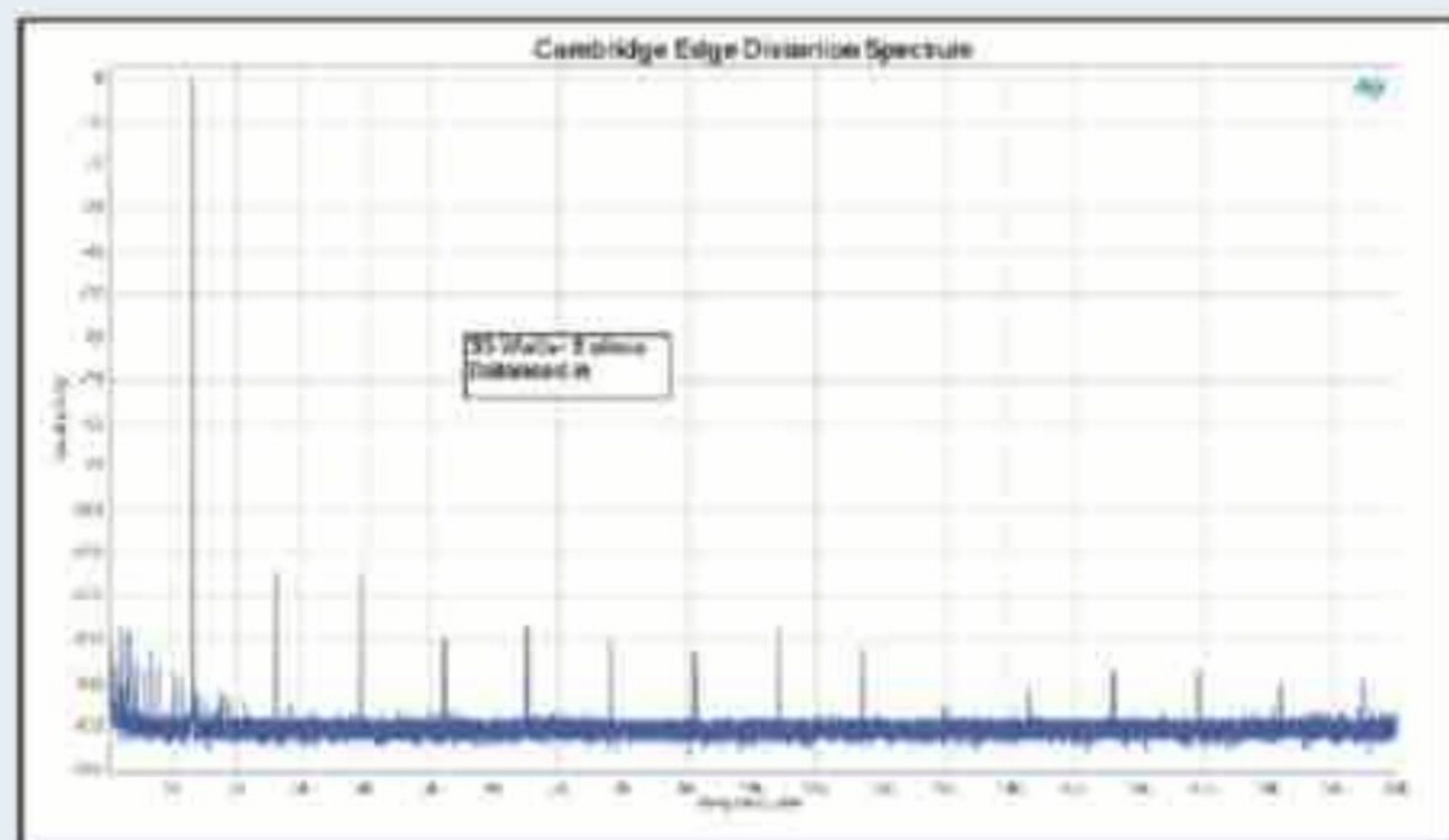


Figure 5: The distortion spectrum of a 1.3 kHz sine wave at 33 W into 8 Ω challenges my measurement capability.

Edge W to achieve galvanic isolation drastically reduced the measured hum to the same levels seen with a shorted input.

The signal-to-noise ratio (SNR) is specified as better than 93 dB referenced to 2.83 V out, with nothing said about balanced or unbalanced inputs. That's almost exactly what I measured using the balanced inputs and a 1 kHz test signal. The unbalanced inputs did not measure as well, and were limited to -80 dB due to the 60 Hz hum loop.

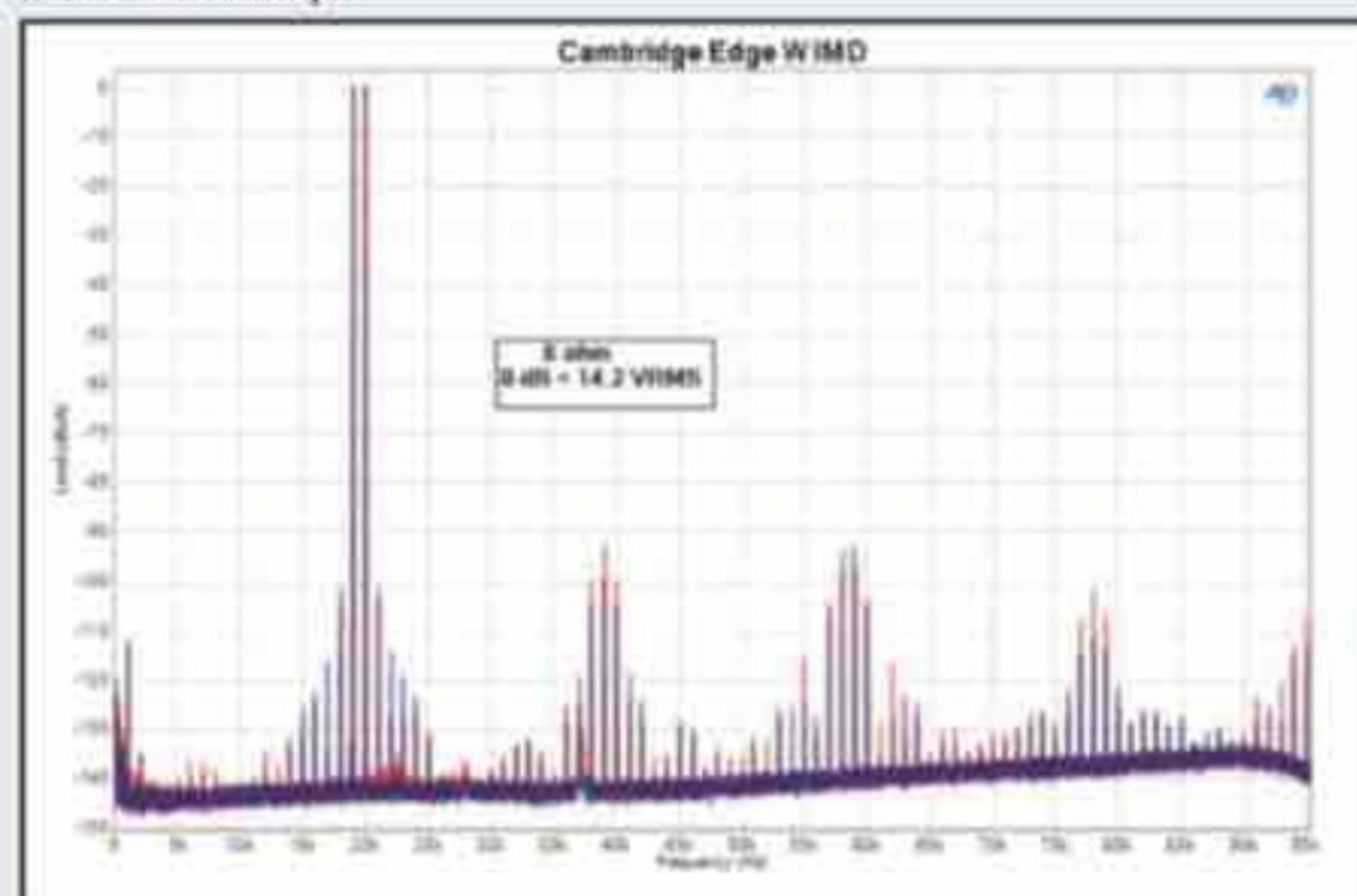


Figure 6: The intermodulation distortion of a mixed 19/20 kHz signal is remarkably low.

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Using the balanced inputs, I ran frequency response sweeps in both differential and common mode (see **Figure 3**). We see that the differential mode response is dead flat in the audible range; I measured the -3 dB frequency at over 100 kHz, so this amplifier easily meets its 80 kHz specification. On the low end, my limit was the 10 Hz high-pass filter of the analyzer, with the Edge W's response still being flat to that frequency.

Ideally, the common mode response of a balanced input should be zero (i.e., infinite common mode rejection ratio). The lower curves show the output-referred common mode rejection (CMR) of the Edge W's balanced circuitry at -52 dB and -55 dB for the left and right channel, respectively. In order to convert this to the more usual input-referred figures, the gain must be subtracted, so the actual CMR is -74 dB and -77 dB, which are decent figures, and sufficient to knock down any reasonable level of induced hum at the inputs to levels below audibility. The CMR is relatively constant with frequency, an indicator of good input circuit design.

Using 8 Ω , 4 Ω , and 2 Ω dummy loads, I ran sweeps of distortion versus output power (see **Figure 4**). At 8 Ω , the distortion rise began at 100 W, with the distortion reaching 1% at about 120 W. Power at 4 Ω is nearly doubled, with the distortion rise beginning at about 180 W and 1% being reached at over 200 W. At 2 Ω , the Edge W delivers more than 200 W. This amp should have no problem driving difficult loudspeaker loads. Note that the balanced distortion is somewhat lower than unbalanced, though both results are quite

good. Some of the difference could also be attributable to the input transformer I used to knock out the hum loop in unbalanced mode.

As one can infer from the power measurement data, where the Edge W really shines is distortion. **Figure 5** shows the distortion spectrum of a 1.3 kHz sine wave at 33 W out into an 8 Ω load. All harmonics are below -115 dB (0.00017%), hugging the residuals of the APx525 analyzer I use, so the Edge W's distortion likely well below that spectacular figure. The spectrum of the intermodulation distortion (IMD) torture test (19 kHz and 20 kHz, equal magnitude) with two 14.2 V sines into 8 Ω is shown in **Figure 6**; taking into account the summed signal, the first order IMD is at nearly -120 dB and the worst sidebands are at nearly -110 dB. One would expect that a multitone measurement would look pretty good, and one would be correct (see **Figure 7**). This is the spectrum of a 40-tone signal at 1 V per tone, and other than the previously noted hum components, there's no distortion components bigger than -112 dBV (2.5 μ V). This is excellent and impressive performance.

The Edge W's Secret

One clue to why this amp works so is revealed in **Figure 8**,

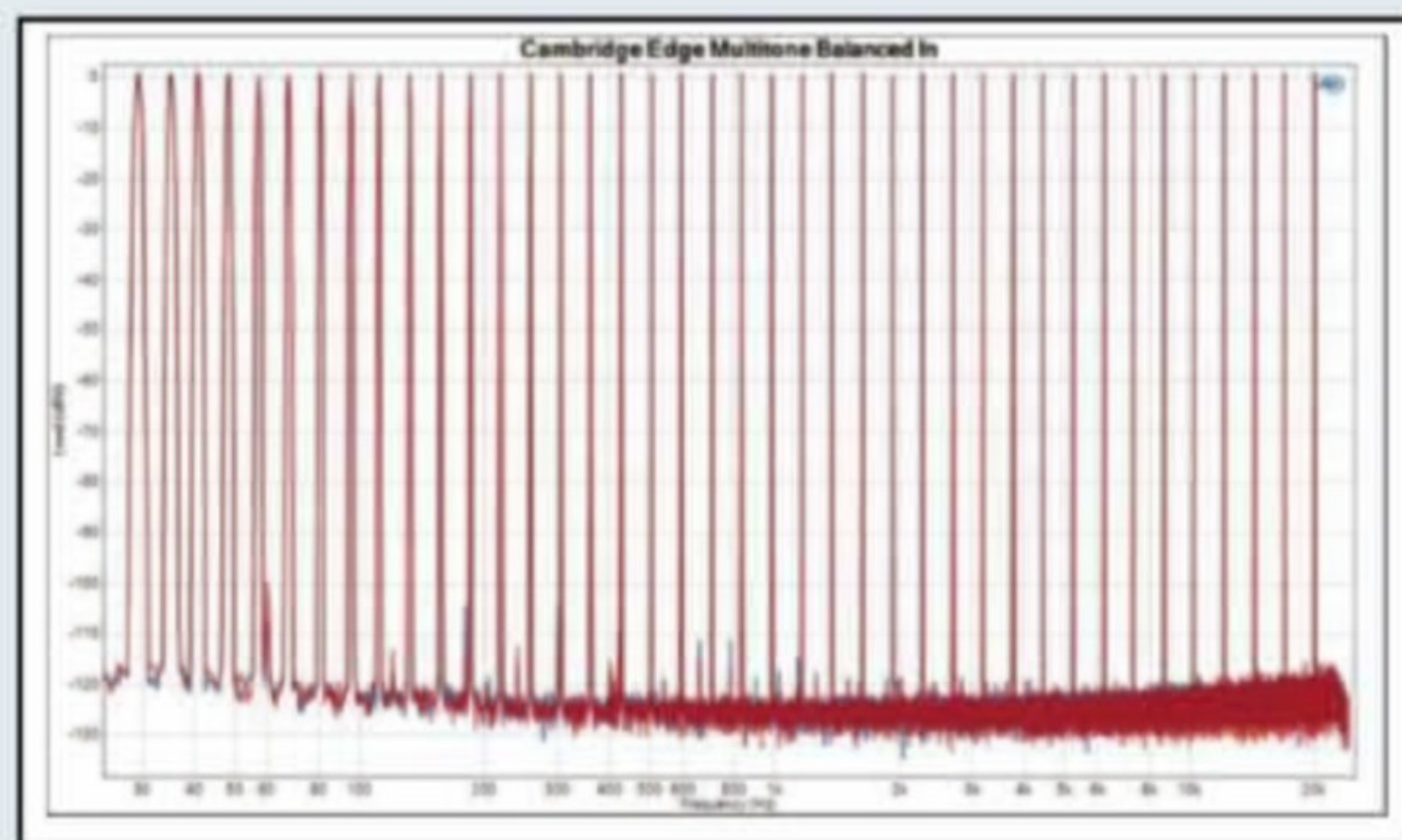


Figure 7: A 40-tone signal's very clean spectrum is a demonstration of the Edge W's low harmonic and intermodulation distortion.

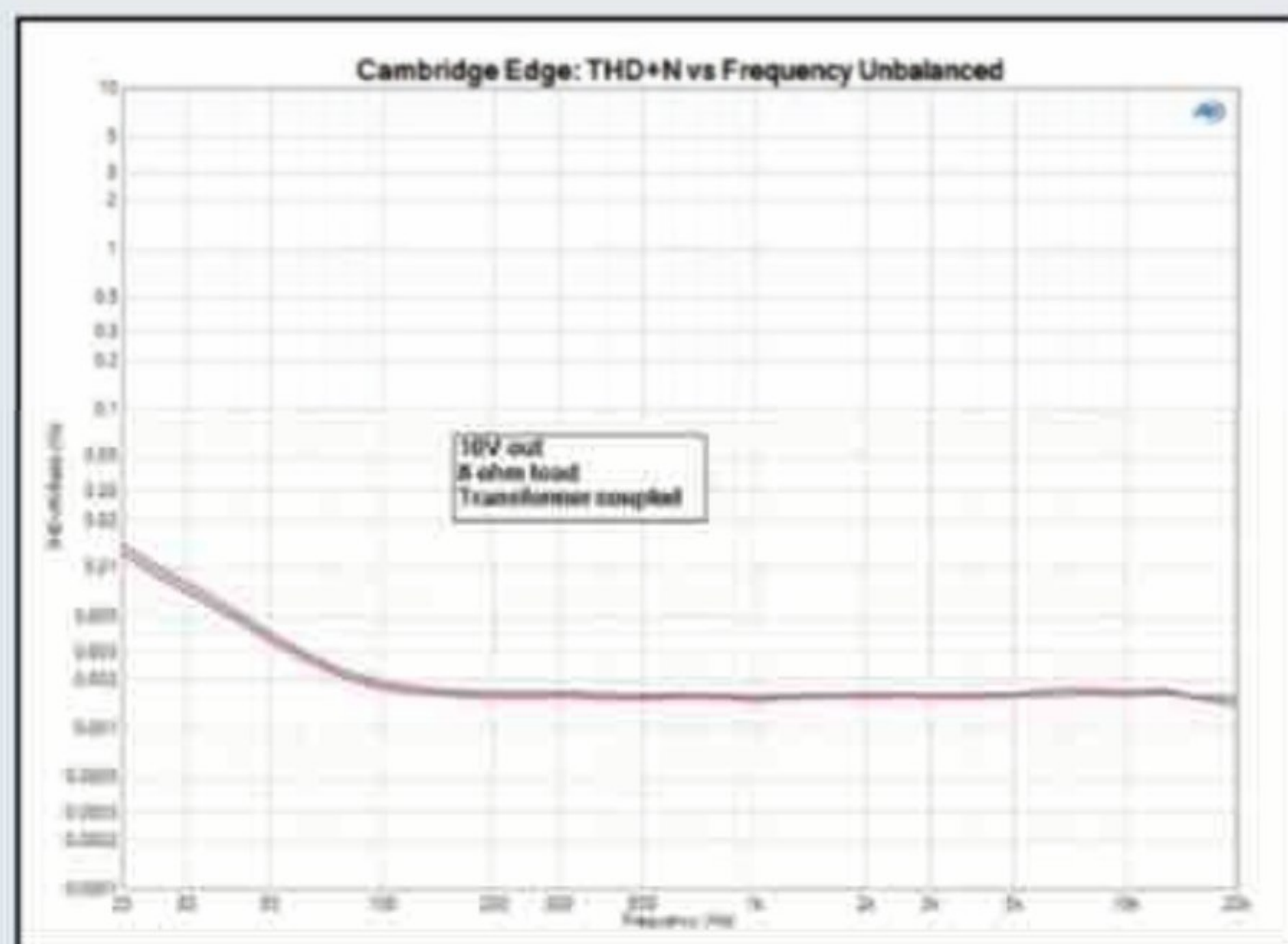


Figure 8: Distortion versus frequency at 12.5 W into 8 Ω is very low and constant, indicating well-engineered feedback and compensation. The rise at low frequencies is a test setup artifact (see text).

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a plot of THD+noise versus frequency at 10 V out into an 8 Ω load. Ignore the rise in distortion at low frequencies—this is caused by the transformer I had to insert to break up the unbalanced input's ground loop—but note that the distortion does not rise at higher frequencies. This suggests some very smart engineering of the feedback and/or feedforward, since in most other amplifiers, the distortion in the top octave starts to rise because of the roll-off of open loop gain and consequent reduction of feedback. In summary, aside from the ground loop problem in unbalanced mode, these are superb measurements.

In Use

Once I finished the measurements, I used the Edge W in my main stereo system to drive the midbass, midrange, and tweeter section of my speakers. These speakers are not a particularly onerous load with impedance staying at 4 Ω or above at any frequency, but they aren't particularly efficient, either. Much to the distress of the other inhabitants of the house, I tend to listen at levels approximating live music.

In use, the Edge W seemed sonically to do exactly what I want a power amp to do: make a small signal larger and as little else as possible. The sound never showed any hint of coloration or strain. Quiet and neutral are the best things one can get out of a power amp, and the Edge W in balanced mode did exactly what it should, being an efficient servant and not getting in the way of the music.

Conclusions

The Cambridge Audio Edge W is a gorgeously constructed unit whose measurements show a lot of attention to the engineering. The distortion is low enough to challenge my measurement capabilities, the amp had no problems with low load impedances, source impedance is very low, and the bandwidth is far better than anything you'll ever need. In balanced mode, which is how my stereo system is wired, the measured performance correlated with my subjective impression of complete neutrality and lack of unwanted adventure.

The only real shortcoming is the sensitivity of the unbalanced inputs to ground loops, and this may be a function of which equipment the Edge W is paired with and details of the grounding arrangement. It may or may not be an issue for you, and even at its worst it may not be actually audible in your setup unless you have very sensitive speakers and a very quiet listening area.

Except for the single issue with minor hum at the unbalanced input in my particular testing set-up, it was a delight to have the Edge W on my bench, and I'd be perfectly happy to have it in my system for a long, long time to come. 

Resources

"Class XD Explained," Cambridge Audio,
<https://www.cambridgeaudio.com/usa/en/blog/class-xd-explained>





[te·tra·kɔɪl]

Te-tra-co-il

(noun)

1. Eighteen Sound's unique transducer technology, integrating four voice coils in a dual gap motor structure, reaching unmatched power handling and acoustic output, reduced power compression, and improved cone control at Xmax.
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Practical Testing of Voice-Controlled Audio Devices

By
Steve F. Temme

(Listen, Inc.)



This article discusses tools and techniques that are available to accurately measure the audio performance of voice-controlled and connected devices under the many various real-world conditions they may be used. It covers basic acoustic measurements such as frequency and distortion response, which have always been carried out on conventional wired systems, and the more complex real-world tests that apply specifically to voice-activated devices, along with the techniques and standards that may be used.

Although smart speakers, hearables and automotive infotainment systems are often categorized as different types of devices, they all have one thing in common—voice control. Furthermore, they all present similar challenges and complexities of test. These include many possible interfaces ranging from hardwired to wireless (Bluetooth, cloud-based), smartphone, voice (“Hey Siri,” “OK Google,” “Alexa”), and in the case of automotive, even USB, memory stick, and CarPlay/Android Auto.

Testing these devices also involves considerable signal processing both on the record side (e.g., beamforming, background noise filtering, and voice activity detection) and on the playback side (e.g., loudness, compression, equalization, and active noise cancellation). This results in changes to their characteristics according to real-world conditions such as the physical environment, playback levels, and background noise, so it is often necessary to test such devices at different levels and different environmental conditions.

Their multifunctional nature means that there are also many aspects of the device that may need to be tested, ranging from voice recognition to music playback or even operation as a hands-free telephone, telephone headset, or hearing aid. Voice recognition usually needs to also be tested with different languages, levels, and environmental conditions (e.g., with and without background noise, different signals etc.)

Most of the tests discussed in this article are relevant to all smart devices including smart speakers, hearables, and automotive infotainment, and some hearable-specific additional tests are also detailed. Finally, the article presents a check list of the test-system functionality required to fully characterize voice controlled audio devices.

Basic Acoustic Tests: Wired and Wireless Measurements

Basic acoustic tests measure the device under static conditions to measure one or multiple microphones and speakers or earphones characteristics such as frequency and impulse response, distortion, and for stereo headphones, left/right tracking. Usually, both wired and wireless performance are measured for comparison. This requires a calibrated input (e.g., a headphone amplifier or auxiliary input) for wired responses and a calibrated Bluetooth interface for wireless measurements.

In addition, a calibrated mouth simulator or source speaker, equalized for a constant sound pressure output level versus frequency, and a calibrated measurement microphone or ear simulators are required for both send and receive acoustical measurements. Although a single measurement microphone usually suffices for smart speaker tests, a head-and-torso simulator (HATS) is recommended for hearables and infotainment systems measurements, since it has both a calibrated mouth and ear simulators to truly represent a typical listener. However, for

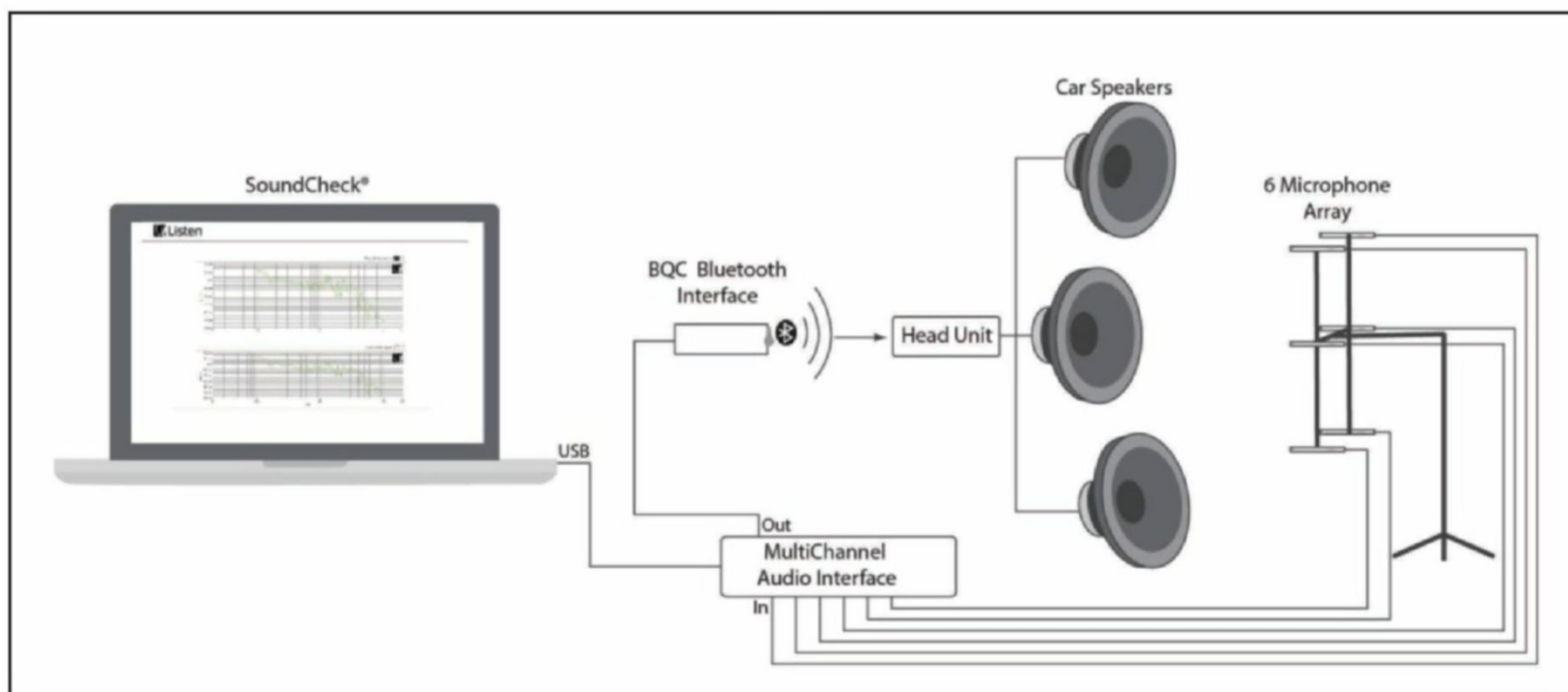


Figure 1: The test configuration for basic wired and wireless infotainment measurement

hearables, ear couplers offer a lower-cost option, and for automotive testing, an array of six measurement microphones is often substituted (see **Figure 1**).

There are currently no standards for testing most smart devices. That said, standards and guidelines exist for many devices that are functionally very similar (speakers, headphones, hearing aids, telephones, etc.), so it makes sense to borrow from these. Such standards include International Electrotechnical

Commission (IEC) and European Norm (EN) for loudspeakers and headphones; Institute of Electrical and Electronics Engineers (IEEE) for headsets; IEEE/ Telecommunication Industry Association (TIA)/ International Telecommunication Union (ITU) for telephone tests; American National Standards Institute (ANSI) and IEC for hearing aid standards; and European Telecommunications Standards Institute (ETSI) for background noise.

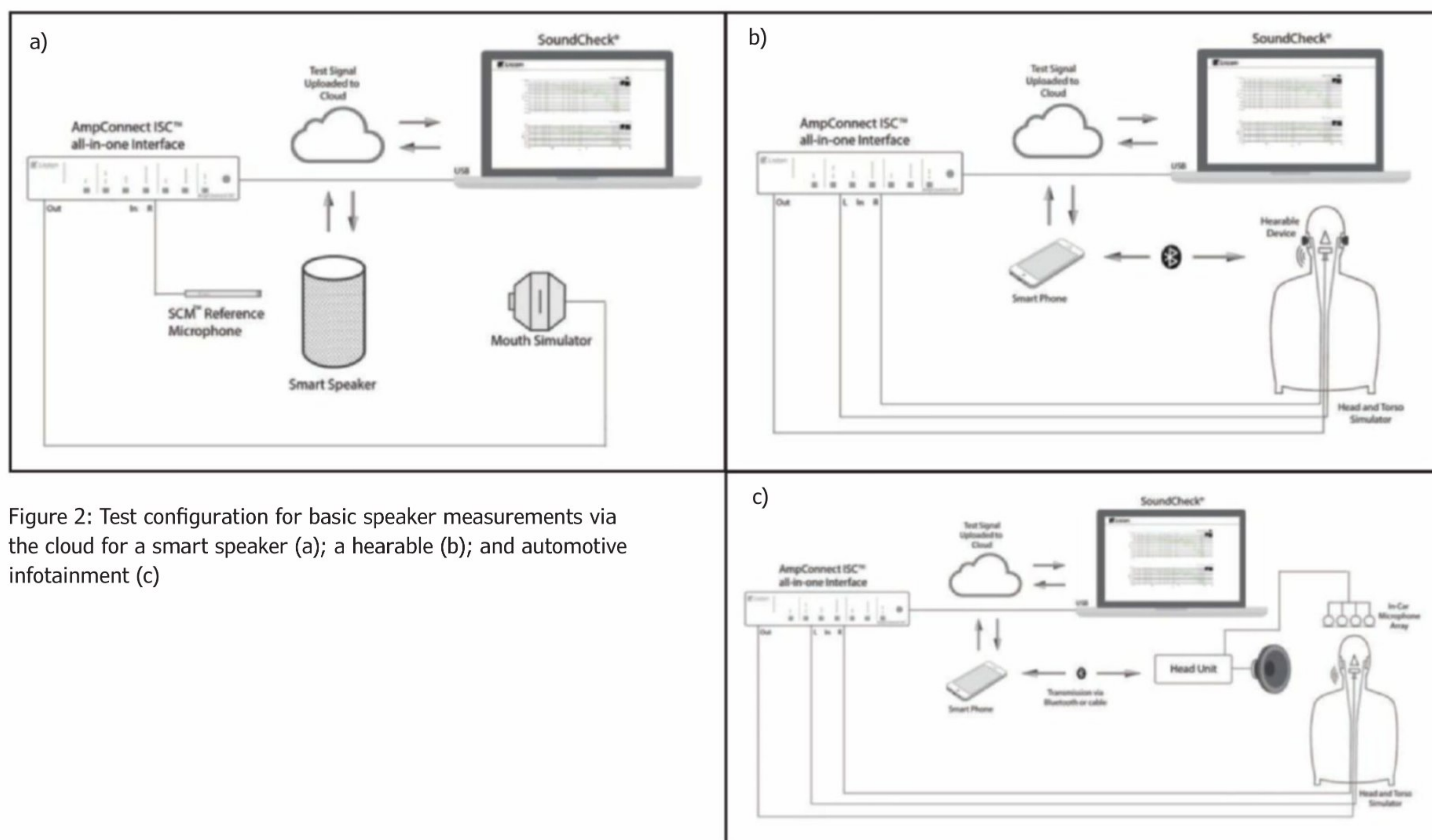


Figure 2: Test configuration for basic speaker measurements via the cloud for a smart speaker (a); a hearable (b); and automotive infotainment (c)

Open-Loop Measurements

Although, on the face of it, playback and record measurements via the cloud look simple, there is much going on “under the hood” of the measurement system. One of the main challenges that must be overcome is correcting for sampling rate error. Traditional measurement systems rely on the device under test (DUT) having a synchronous input and output. Voice-activated hearables, infotainment systems, and smart speakers are intrinsically open loop, which means that although the device can record or playback a signal there is no synchronous signal path. This introduces the possibility for sampling rate error. In other words, the device may record a signal to a file with a sample rate of 44.1 kHz, but it may have in-fact been recorded at a slightly different sample rate (e.g., 44.09 kHz) due to skew in the actual rate of the clock crystal used to drive the sampler (see **Figure 1**).

A similar error can occur when playing back test signals. That is, the test stimulus may be sampled at 44.1 kHz, but due to error in the playback sample rate the file is actually played back at a slightly faster or slower rate. This sampling rate error results in the component tones of the test stimulus being shifted to either a higher or lower frequency. This shift can then lead to measurement errors due to loss of coherence between the stimulus and response signals, and also changes the phase response.

To overcome this sampling rate error, an algorithm is applied that searches the beginning of the response waveform for a “trigger,” which is used to provide a reference point for alignment and shifting of the response to the stimulus signal. This trigger may be a steady state sinusoid at a pre-set level and frequency (see **Figure 2**), or for more robust performance that is less susceptible to false triggers, a log chirp. The signal is then shifted to DC using a heterodyne filter and all other frequencies are filtered out. The output of the heterodyne filter includes the phase information which is used to estimate actual playback or recording sample rate of the response signal. With this information, the entire response waveform is resampled to the correct stimulus sample rate prior to analysis. This frequency shift step corrects for sampling rate error in the device and makes testing straightforward, enabling frequency and distortion to be displayed in exactly the same way as for a conventional wired device.

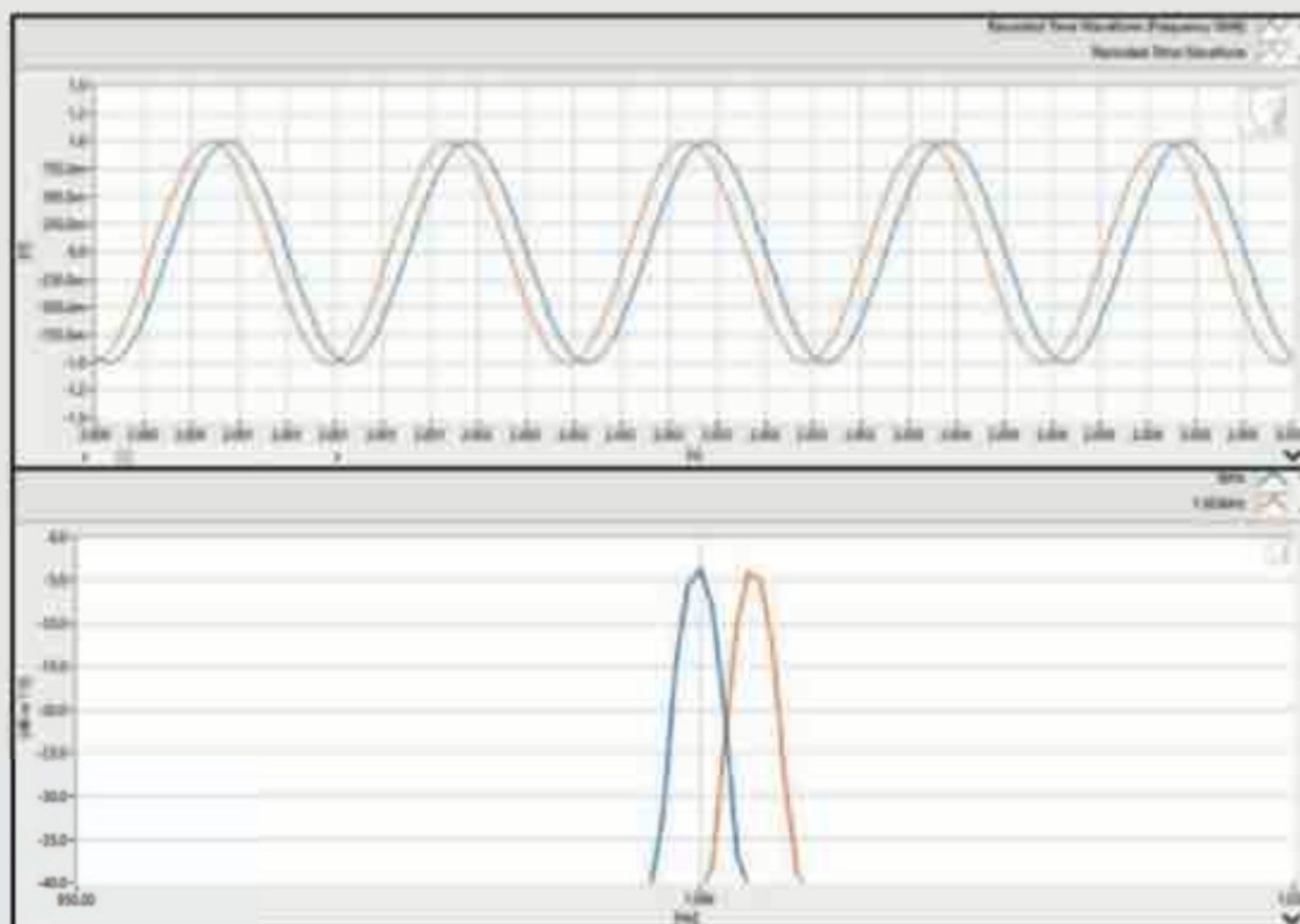


Figure 1: Open-loop measurements with differing sampling rates are shown in the time and frequency domain.

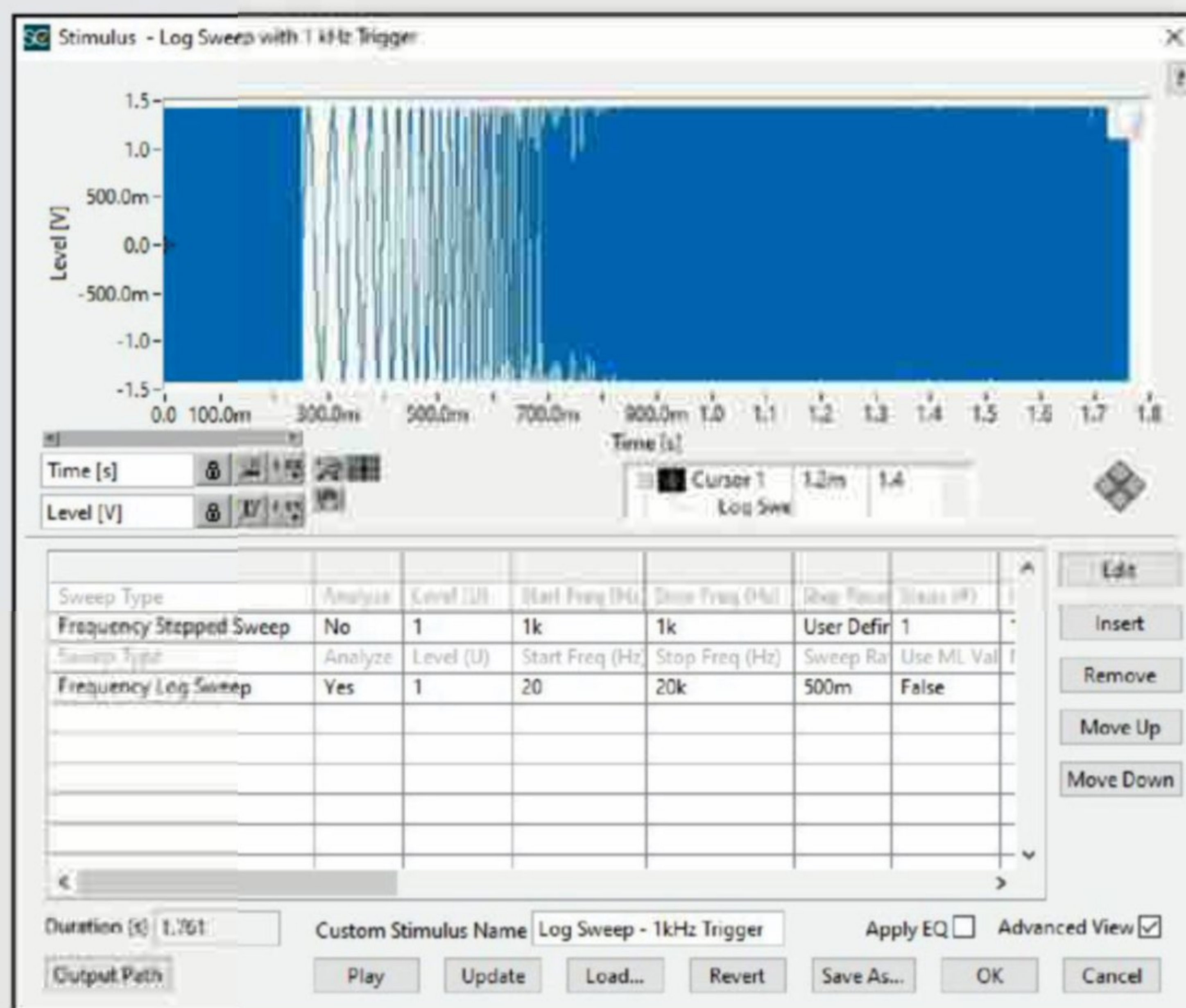


Figure 2: A 1 kHz frequency tone is prepended to stepped sine stimulus to calculate sampling rate and simplify record triggering.

Measurements Via the Cloud

Measurements on smart devices are very similar to standard loudspeaker and microphone measurements, but injecting the stimulus and extracting the recorded signal presents challenges. The only way to do this is like an actual user—via the cloud using voice commands.

An artificial mouth, often contained within the HATS, plays the voice-activation commands to the device's microphones, which are typically connected via Bluetooth to a smartphone and/or a head unit. In the case of a smart speaker, an artificial mouth or small speaker can be used, and a smartphone is not required as the device itself has the Internet connection. The smartphone communicates to the proprietary voice services in the cloud, for example, Alexa Voice Services (AVS), and tells the voice services which uploaded test signal to stream/playback through the smartphone via Bluetooth to the hearable or infotainment system. The test measurement system uses a triggered record to capture the loudspeakers playback signals that can then be analyzed by the test system.

Although the actual physical testing setup is very similar from device to device, the method of wirelessly routing the signal may differ. The test signal must generally be in the cloud to enable playback, but each manufacturer's ecosystem is different in how it plays back from the cloud. Some allow your own recordings to be uploaded, whereas others require them to be on a media streaming platform (e.g., Spotify).

For microphone testing, some systems (e.g., Alexa) allow you to access recordings you have made; others prevent this for security and privacy reasons, which makes making testing those microphones challenging. Each is activated with a different wake word, has a different delay compensation, and records for a different amount of time after it hears the wake word. This must be established for each device under test (DUT), and is usually automated by using record triggering and autodelay.

The physical setups can be taken from the test standards for that particular device. For hearables, the physical configuration for headsets recommended by IEEE 269, in which tests are conducted with the hearable worn on a HATS, is recommended. For smart speakers, since their practical use case resembles speakerphones, the physical geometry recommended by IEEE 1329-2010, which involves testing on a table inside an anechoic chamber, is a good starting point.

In the infotainment world, TIA920 presents a useful test configuration. In all three cases, the artificial mouth, which may be contained within HATS, plays the voice activation commands to the

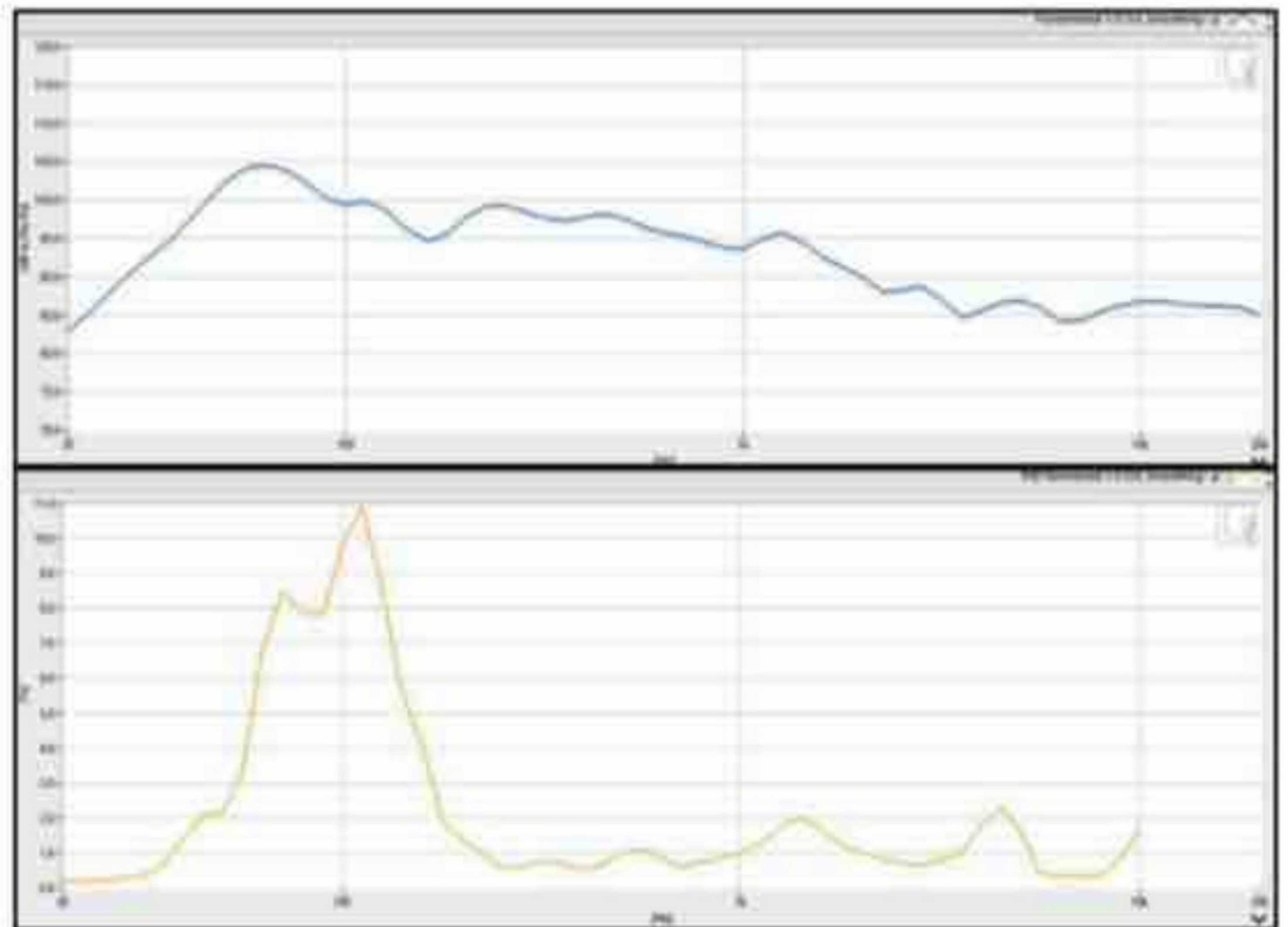


Figure 3: Frequency and distortion responses are shown for an infotainment multi-channel playback system.

device's microphones that are typically connected via Bluetooth to a smartphone, or directly to the cloud in the case of smart speakers. The smartphone communicates to the proprietary voice services in the cloud (e.g., AVS) that tells the voice services which uploaded test signal to stream/playback through the smartphone via Bluetooth to the device. The measurement system uses a triggered record to capture the loudspeakers playback signals that can then be analyzed. Example test configurations are shown in **Figure 2**.

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Playback

When creating a playback test using voice activation, it is ideal to automate the test as much as possible. Wake words and test stimuli should be pre-recorded and uploaded to the music service in the cloud so that they can be triggered by the test system.

The activation signal consists of the system's wake word, followed by a command such "play test signal 1." If a wide range of voice recognition systems from different manufacturers is being tested, it makes sense to record the different voice wake words separately from the playback command so that any combination can be selected.

The test stimulus must also be created and uploaded to the cloud. There is a chance that it might be compressed when it is uploaded, which may introduce some distortion. It can be verified that the MP3 encoding itself does not degrade the test signal by encoding and decoding the test signal, then using transfer function analysis between the original signal and the encoded version. Test signals will be discussed in more detail later, but for basic playback performance testing, a standard stepped sine sweep is preferred.

Once the stimuli and voice commands are created and uploaded, the test sequence simply needs to play the wake word followed by the activation command via the mouth simulator and capture the response with the reference microphones or artificial ears. Stimulus and analysis are then compared as in a traditional measurement to objectively quantify characteristics such as frequency response, impulse response and distortion.

With the test system taking care of all the mechanics of getting the signals in and out of the devices and processing the waveforms, the frequency output and distortion for an earphone, car infotainment system or smart speaker can be displayed in exactly the same way as for a basic wired speaker, or a speaker connected via a Bluetooth interface (see **Figure 3**).

Recording

It is also necessary to measure the microphones or microphone array in the smart device. The setup for this is very similar, except that the wake word and the test stimulus is played through the mouth simulator or source speaker, captured by the device's

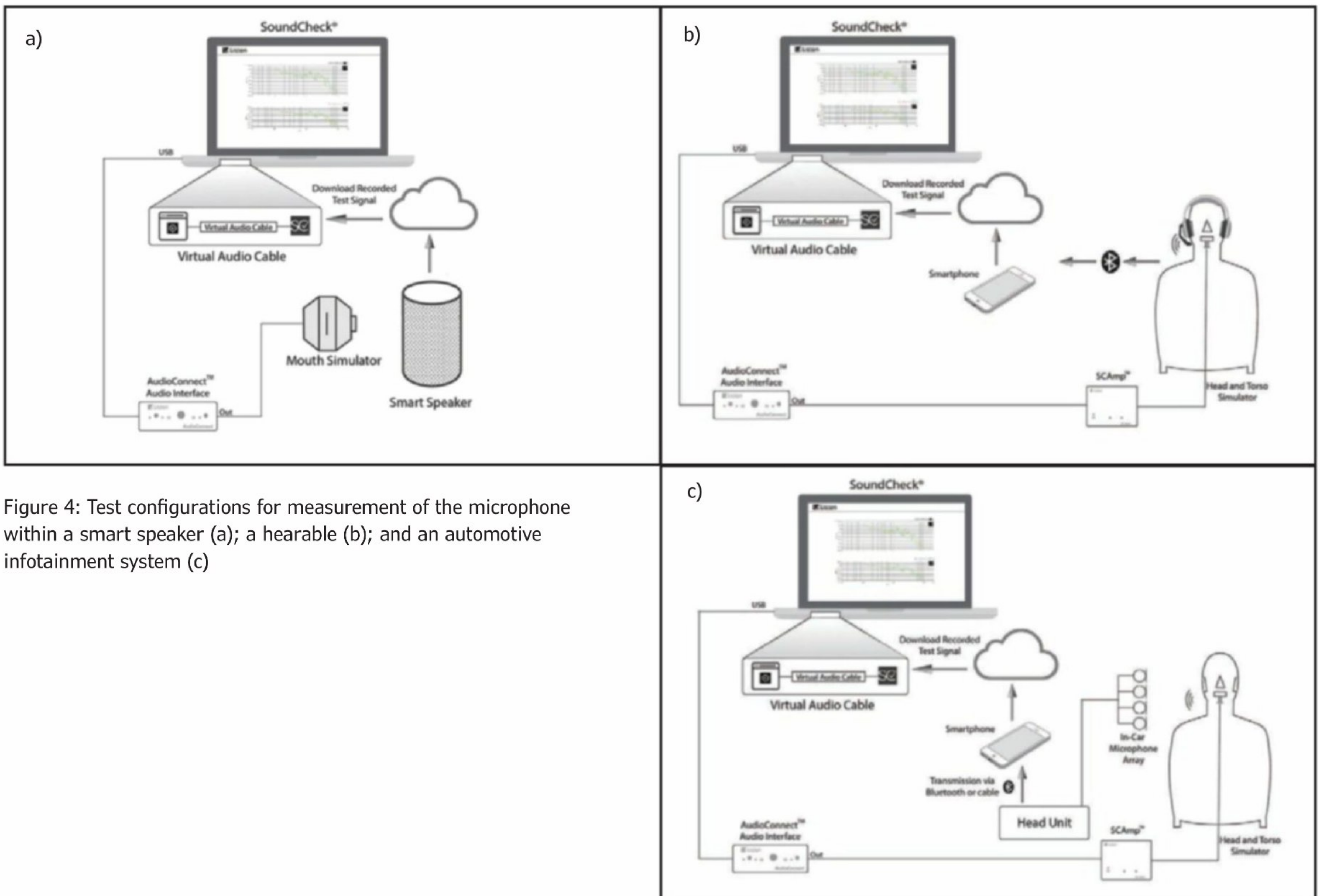


Figure 4: Test configurations for measurement of the microphone within a smart speaker (a); a hearable (b); and an automotive infotainment system (c)

microphones, then the recorded test stimulus is downloaded from the cloud and routed back into the test system, either by recalling the recorded wave file or using a virtual audio cable (see **Figure 4**). Once the signal has been brought into the test system this way, as with the playback test, the appropriate calculations are applied to overcome frequency shift and error and it is analyzed as normal.

Nonlinear Behavior

While the basic tests described earlier are ideal for measuring frequency response and distortion, they barely scratch the surface in terms of the breadth and depth of tests that one might want to carry out on a smart device such as a smart speaker, hearable, or infotainment system in the R&D lab.

Smart devices and systems are generally nonlinear and their response varies according to many factors such as type of background noise, volume of background noise, whether the device is playing music or communicating voice, and more. These considerations bring up a multitude of tests that could yield valuable information about the systems' performance. While the specific tests that are carried out vary according to the manufacturer's

design objectives, some of the test methodologies that might be used, and the industry standards that provide a useful starting point are relevant for the majority of voice-controlled devices.

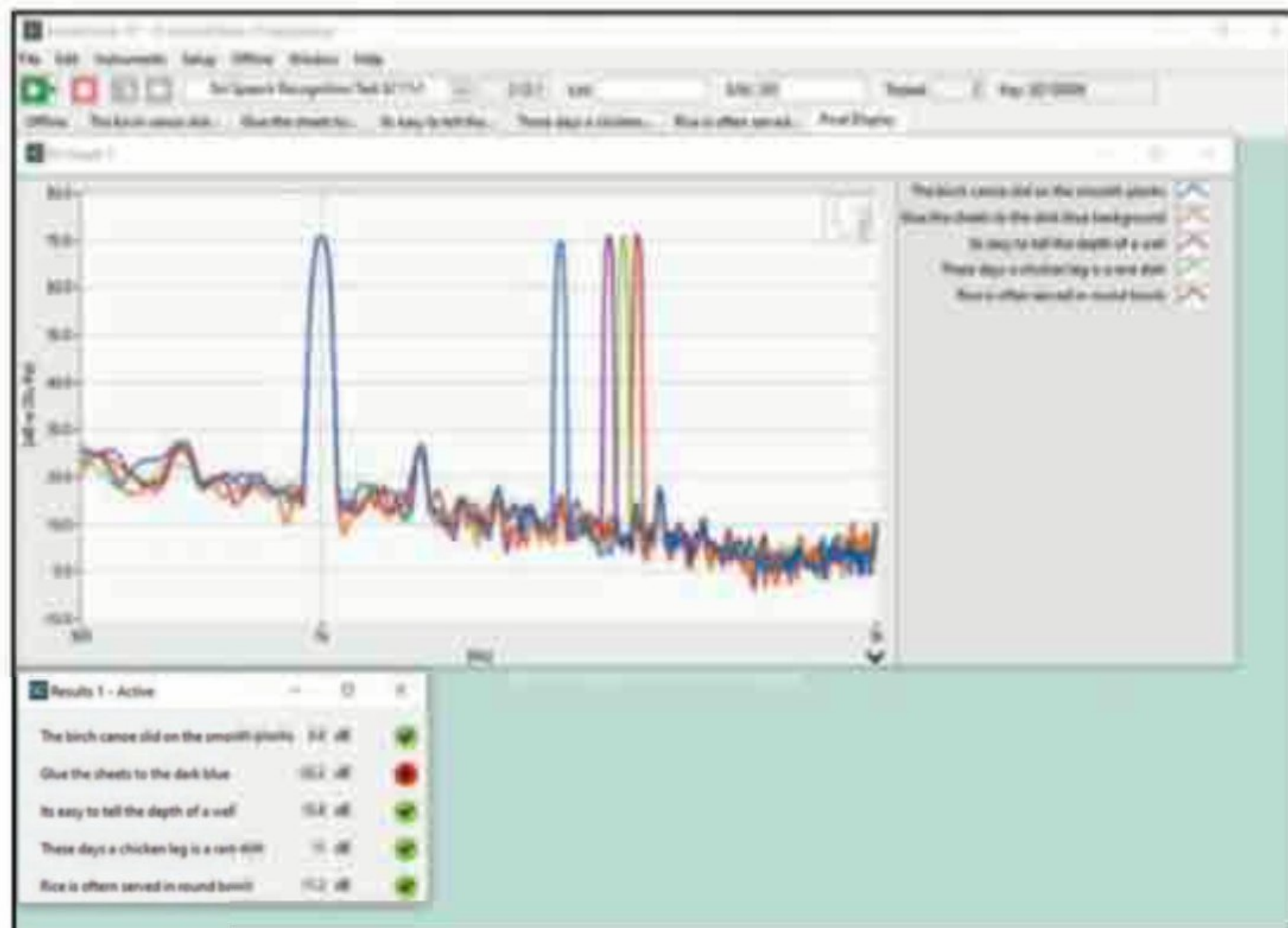


Figure 5: Graphs showing results of end-to-end speech recognition

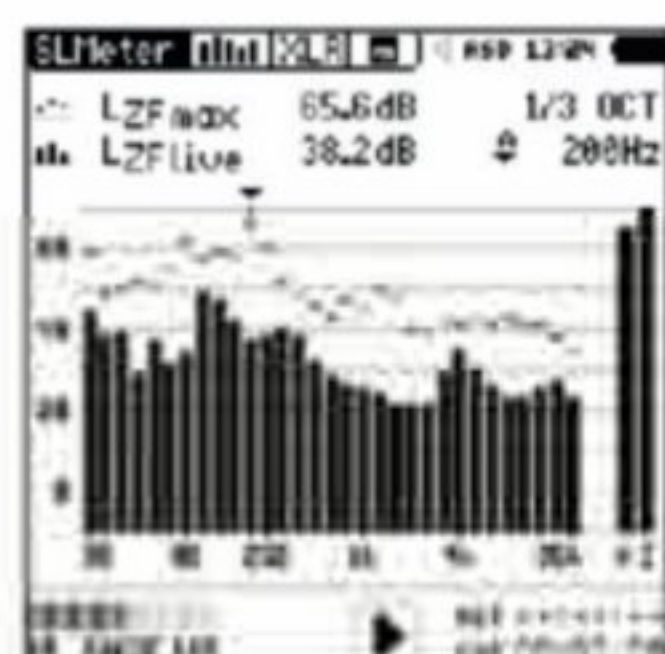
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A significant difference from basic tests is that such advanced tests generally require the use of speech or music as test signals. This means that a test system that can support such test signals is essential and it is also important that the system can measure the active speech level of speech-based signals to ensure calibration and standardization as commonly used in telephony measurements.

Speech Recognition

Speech recognition tests are important for evaluating the system's ability to understand voice commands. The physical setup is the same as in the basic test outlined above. The test method usually involved creating a series of specially titled musical tracks and uploading them to the appropriate cloud service (e.g., iTunes). The track titles are the words or phrases that you want to test, and the actual content of the tracks is a single, dual, or multitone that enables you to identify the track by its audio content.

Generally, "Harvard Sentences"—a collection of phonetically balanced sentences that use specific

phonemes at the same frequency they appear in English—are used for the track titles. These are recommended by "IEEE Recommended Practices for Speech Quality Measurements for VoIP, Cellular, and other Telephone Systems," and are widely used in telecoms voice recognition. Pre-recorded versions of these can be downloaded online, or your own can be recorded (e.g., if you are testing the device with different accents). These track titles are used to request playbacks, and a limit is used to detect whether the correct, or indeed any, signal was played back (see **Figure 5**).

There are many variations on this test, for example, testing different voices and accents, testing an active talker versus an interfering talker or background noise levels, testing with different types of background noise, and so forth.

Background Noise

Background noise is necessary for a wide range of smart system and device tests including voice recognition, signal-to-noise ratio (SNR) optimization of microphone arrays, beamforming directionality studies and more. Where background noise is needed, ETSI standard ES 202 396-1 contains binaural recordings made in different noise environments (e.g., café noise, subway, road noise, traffic noise, sirens etc.). These can be played using a costly dedicated system, or with sufficient expertise, by programming your own playback method.

However, some systems offer integrated ETSI background noise modules, which fully integrate the ETSI standard library with the test system, playing the background noise at calibrated levels, which can be controlled and adjusted as part of a pre-programmed test sequence (see **Figure 6**). Such test sequences may include loops to incrementally increase the volume or change the noise and repeat the test, making it simple to implement a test that increases the background noise by fixed levels until the voice is no longer accurately recognized (see **Figure 7**). In addition to being a very cost-effective approach, it also significantly simplifies the physical test set-up and reduces test development time.

Voice Quality

Smart speakers, infotainment systems, and hearables often function as hands-free communication devices. For such use cases, we can lean heavily on telephone test sequences such as the TIA standard TIA920, which covers both hands-free devices (as in automotive applications and smart speakers) and headset devices (e.g., hearables). In fact, smart speakers and infotainment systems can be tested as a speakerphone exactly as defined in

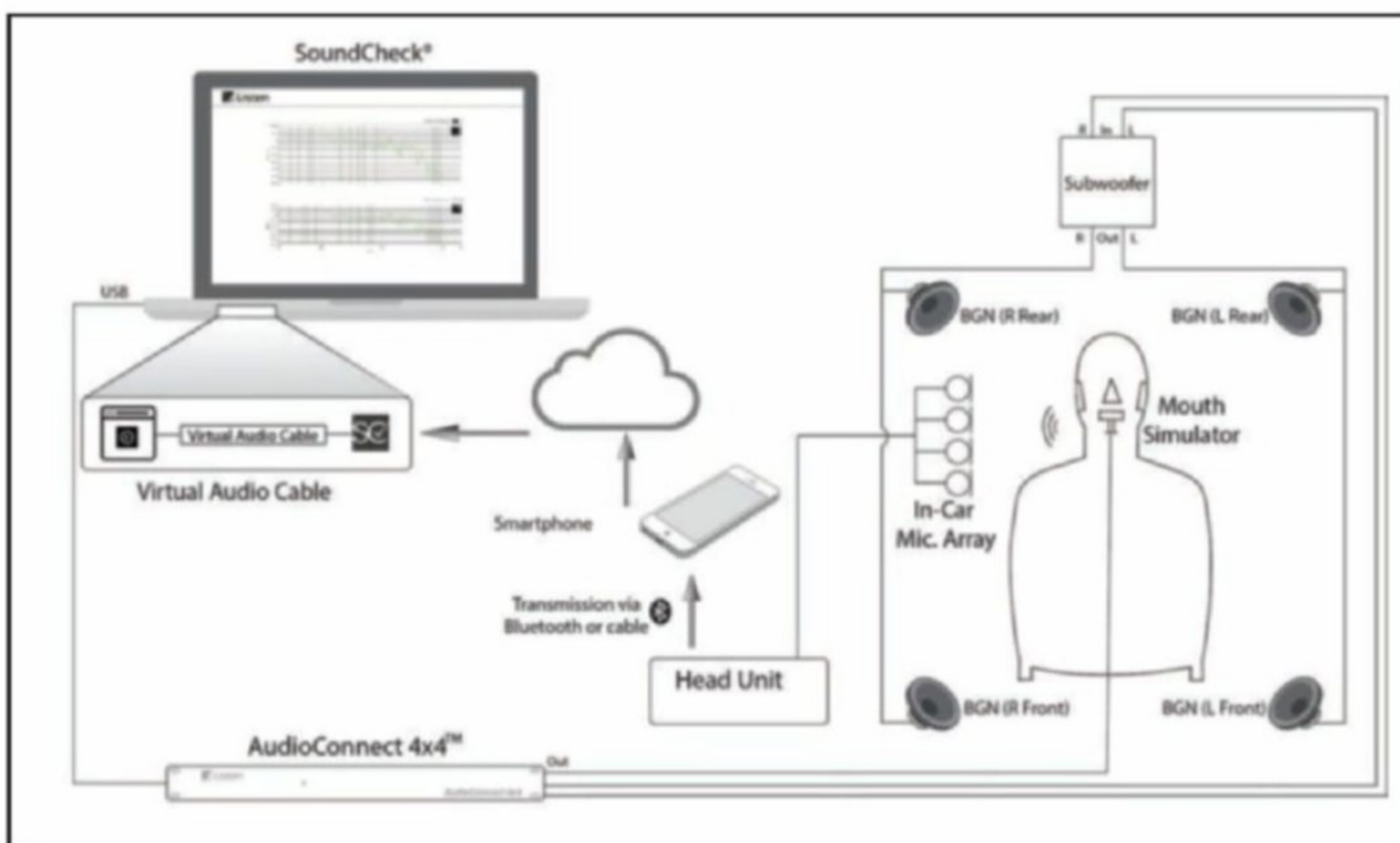


Figure 6: The test configuration is for hands-free measurements with ETSI standard background noises. In a hearable setup, the device's microphone would replace the in-car array, and the signal would go directly from that to the smartphone.

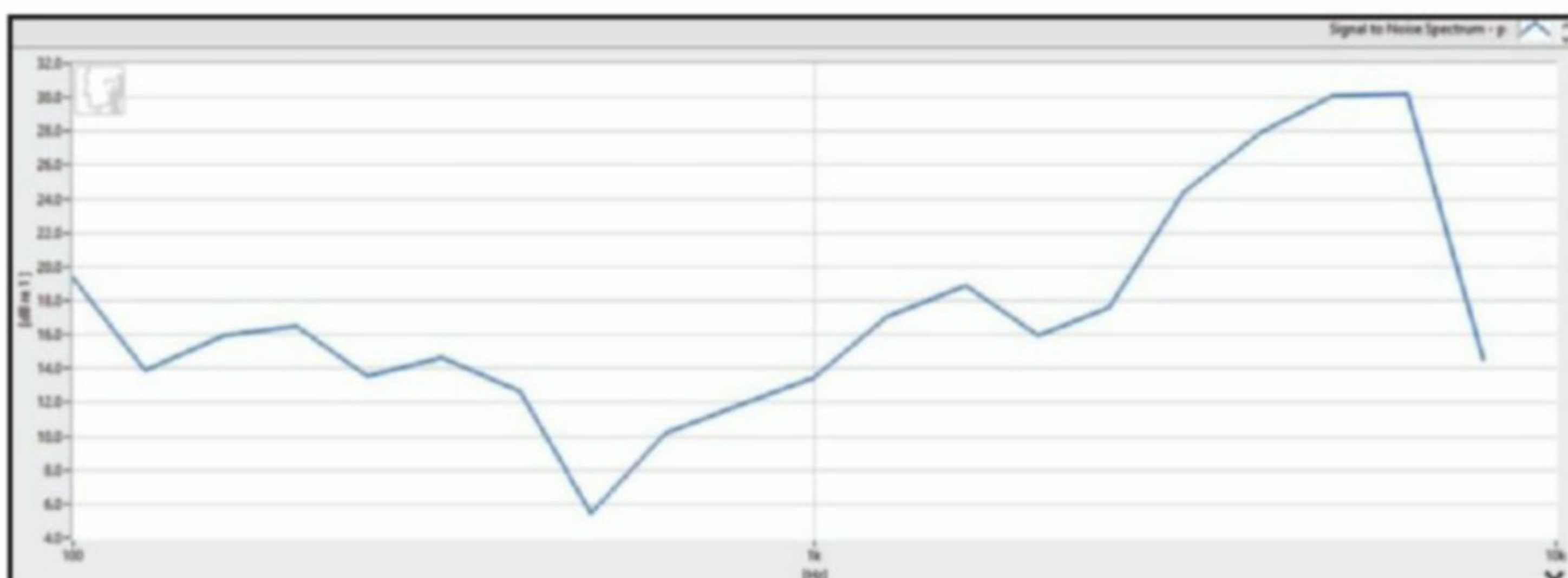


Figure 7: The signal-to-noise ratio (SNR) for a microphone array measured with ETSI standard background noises

the TIA standard, and a hearable can be tested as a headset exactly as defined in the TIA standard.

Although the test setup in each case is identical to the basic acoustic test setups outlined above, the test sequences for measurement to the standard are highly complex and either require considerable expertise in this area to create them, or they may be purchased as an off-the-shelf package.

Noise Cancellation

Although most commonly used for hearables, noise-cancellation tests also have applications in automotive infotainment systems for reduction of cabin background noise. The test configuration for measuring noise cancelling headphones is shown in **Figure 8**. The test procedure can be broken down into three measurements and three calculations. First, the headphones are removed from the head and torso, the noise signal is played, and the spectrum measured through the open ear. This un-occluded ear spectrum is used as the baseline for the noise. Next, passive isolation is measured by placing the headphones onto the head and torso, playing the noise signal again and measuring the noise through the artificial ear.

Finally, the measurement is repeated with the active noise cancellation (ANC) circuit turned on.

These measurements result in three spectra, which are used to calculate the three attenuation parameters. Passive Isolation, which quantifies how much noise is attenuated simply by the headphones being worn, is calculated by subtracting the un-occluded baseline curve from the second measurement where the headphones are in place,

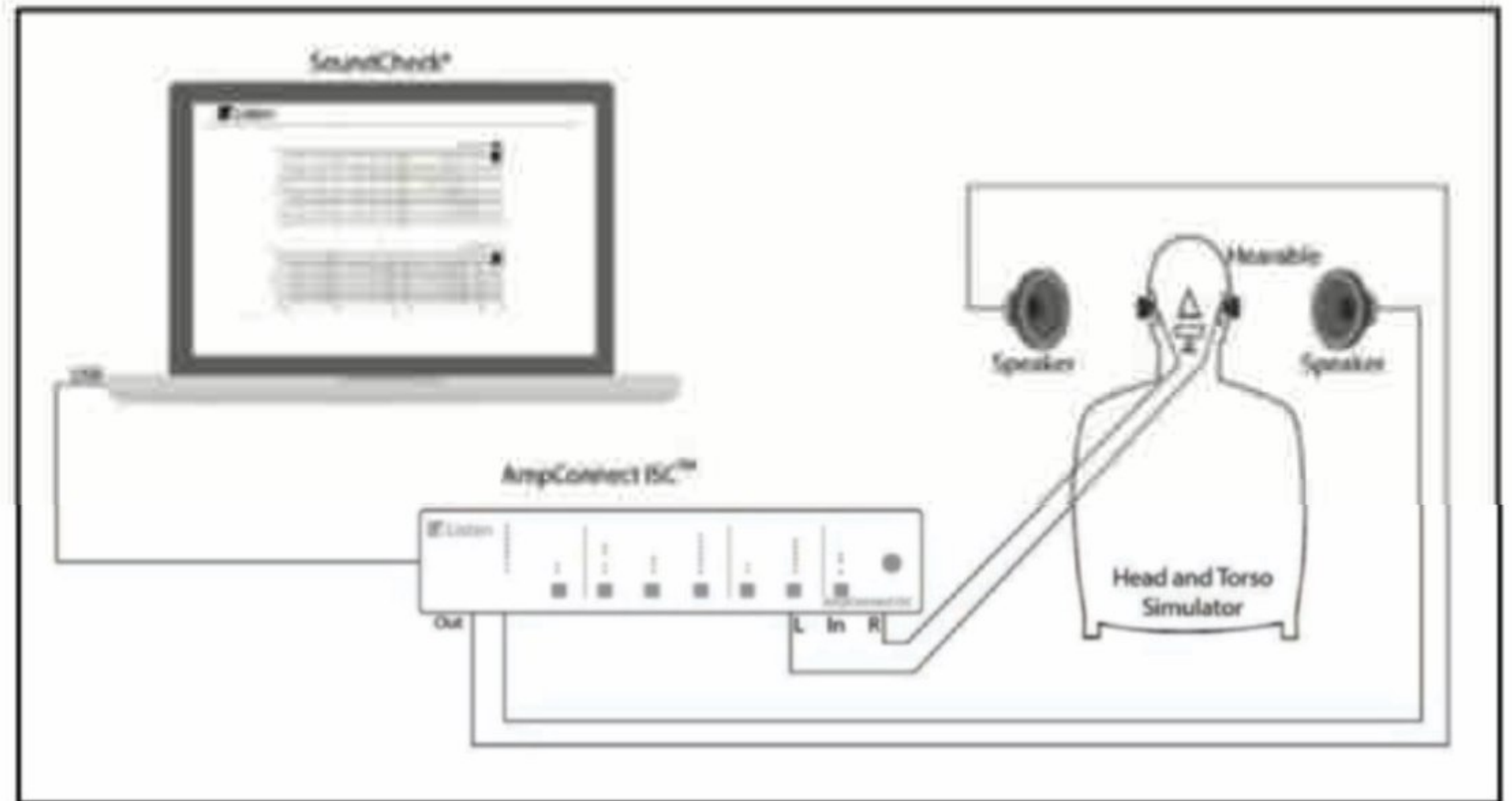


Figure 8: A test configuration for measuring ANC hearables

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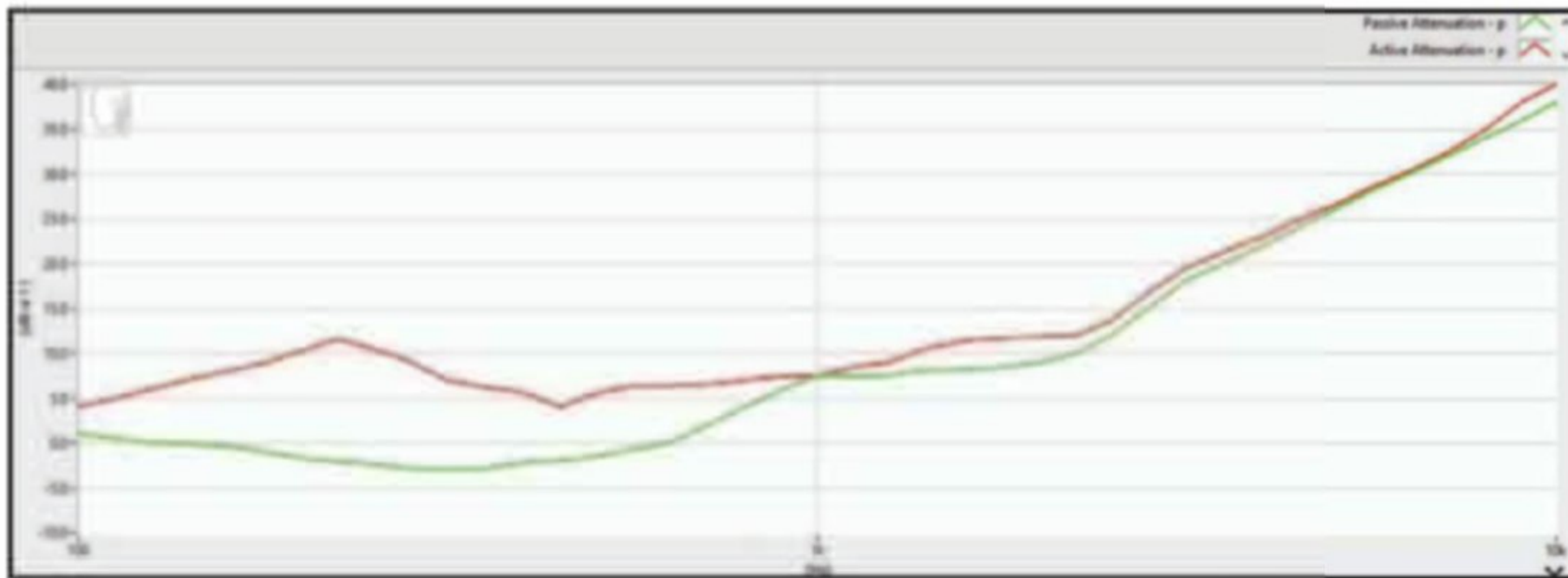


Figure 9: These are typical noise cancelling measurement results.

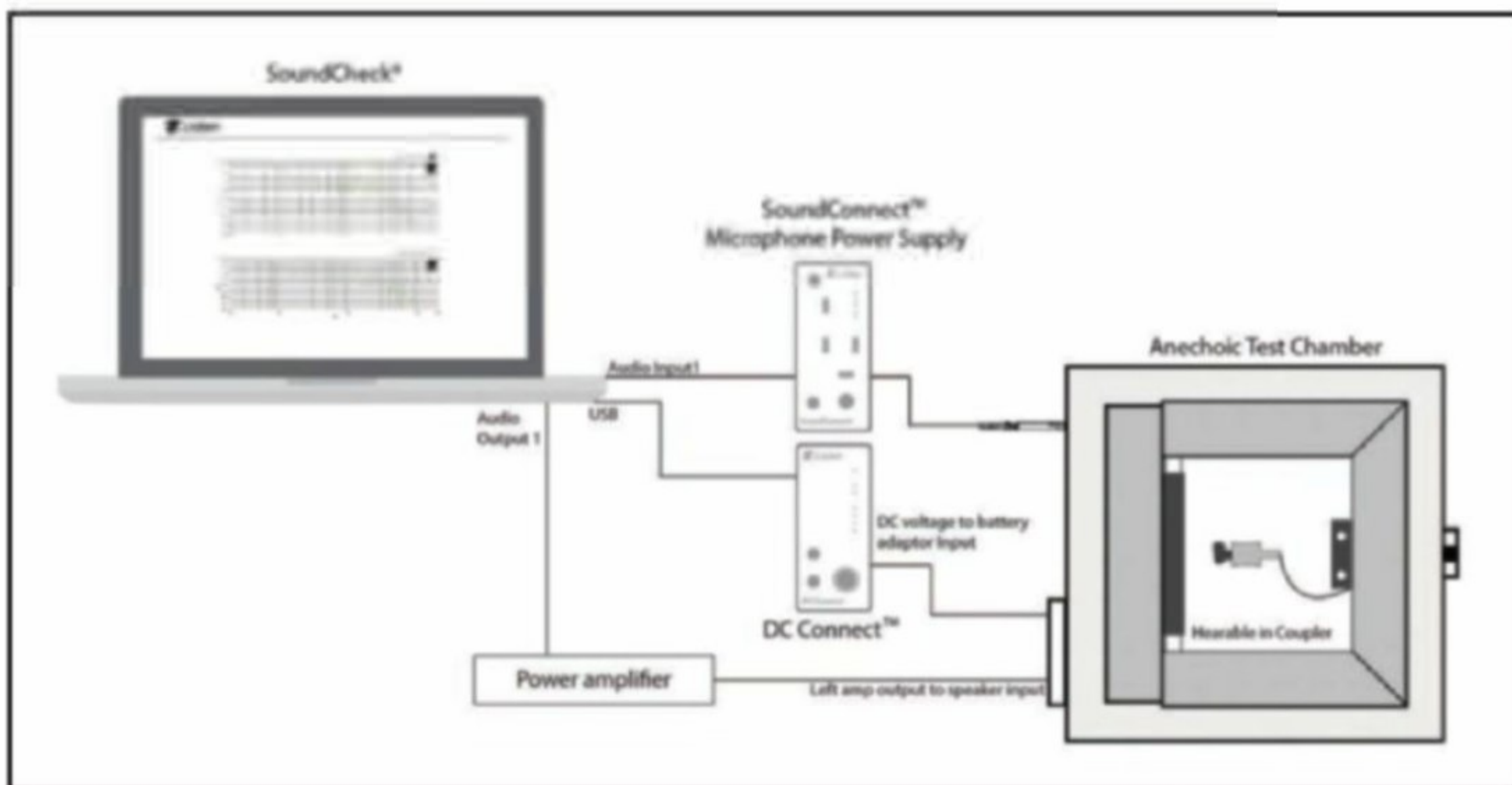


Figure 10: A test configuration for measuring hearable to hearing aid standards

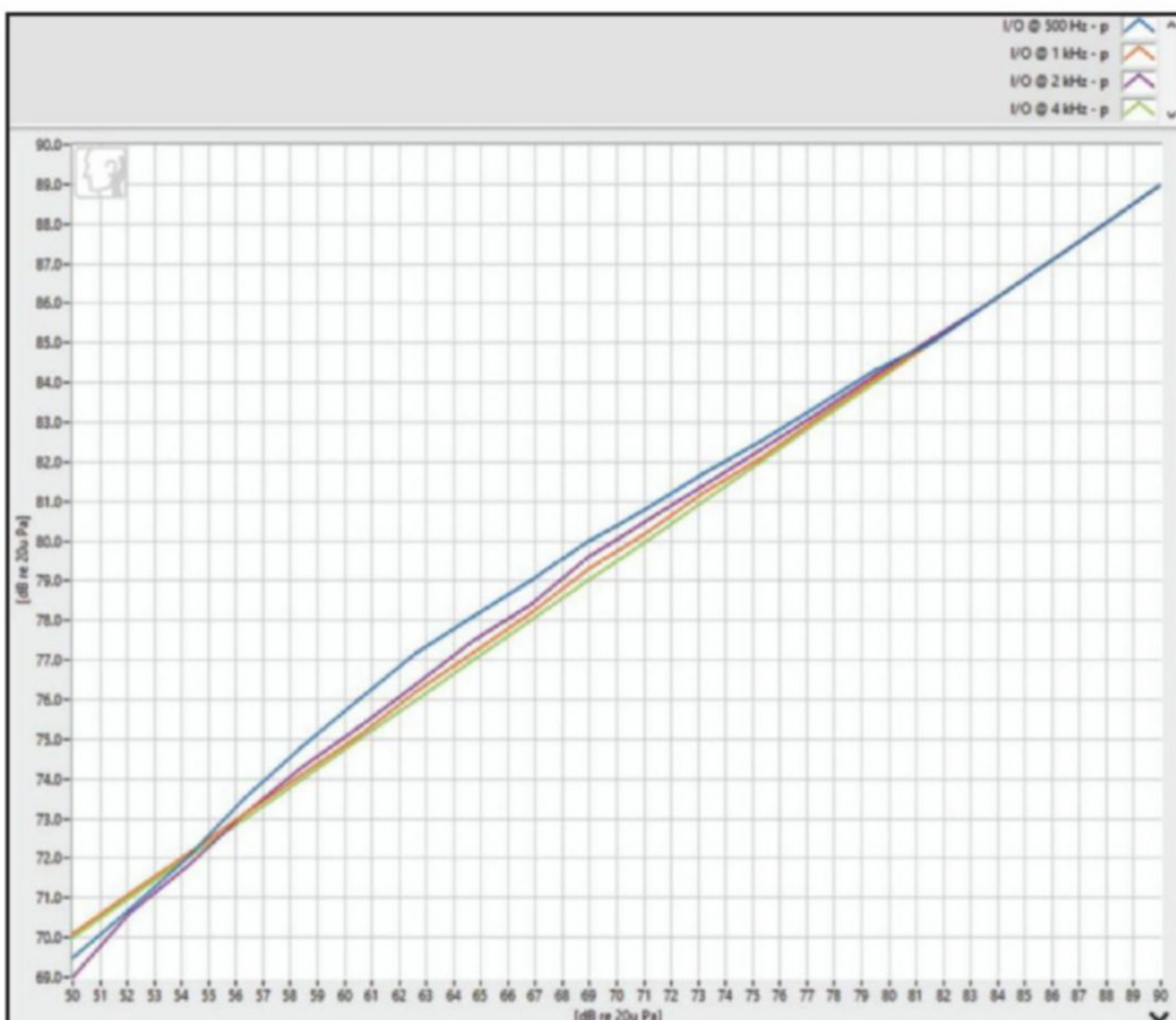


Figure 11: Hearable measurement of Input vs. Output

but noise cancellation is disengaged. Passive isolation can be significant across the frequency band but is most prominent at higher frequencies. Circum-aural headphones and in-ear earphones provide a reasonable level of passive isolation, whereas supra-aural headphones and earbuds typically offer much less.

Active Attenuation, which quantifies how much noise is reduced (or sometimes increased) when the active cancellation circuit is engaged, is calculated by subtracting the second measurement (noise cancellation off) from the third measurement (noise cancellation on). Typically, this will be most prominent in the lower frequencies.

Finally, total attenuation is calculated by subtracting the curve with noise cancellation turned on from the baseline measurement without headphones. This represents the end-user's experience of the device, combining both passive and active components in attenuating background noise (see **Figure 9**).

Often this test is optimized by performing each measurement multiple times and averaging the resulting spectra. This accounts for variations in the fit of the headphones, which for some devices can impact the passive and even the active portions of the noise cancellation performance.

Some ANC headphones allow control of the degree of noise cancelling typically via a smartphone app. This is designed to account for different listening environments (e.g., on a plane with maximum noise cancellation versus walking on a sidewalk next to a busy street where one should be more aware of the environment). In this case, measurements can be configured to show the level of noise cancellation versus frequency for the different environments. Again, this is a case for using a calibrated, simulated background noise system such as that specified by ETSI to recreate the different test environments in a very repeatable way.

Some noise cancelling algorithms actually alter the frequency response of the music being played when the noise cancelling circuit is engaged. This effect can be evaluated by measuring the frequency response using the above methods, both with and without the noise cancellation turned on, and comparing the resulting response curves. Differences, if they are present, will typically be in the 1 kHz to 3 kHz range.

Hearing Assistance

Fairly specific to hearables is the integration of hearing enhancement for personal sound amplification product (PSAP) or hearing aid functionality, and it is desirable to measure

their amplification or compression algorithms along with their corresponding attack and release times. This is usually measured according to ANSI/IEC hearing aid standards in an anechoic test box (see **Figure 10**), where the loudspeaker is equalized to have a flat frequency response and the hearable is inserted into a 2cc coupler or “artificial ear.”

The ANSI S3.22 standard describes the method for testing hearing aids with automatic gain control (AGC) and can be applied to hearables that offer this functionality. The device is placed in an anechoic box, and an amplitude sweep from 50 to 90 dB SPL played through the test box speaker. This sweep is repeated at four frequencies (500 Hz to 4 kHz) in octave increments. The output of the hearing aid is plotted vs. the input on the graph to quantify the gain added for low level signals and compression at high levels for hearing loss compensation (see **Figure 11**).

Conclusions and Test System Recommendations

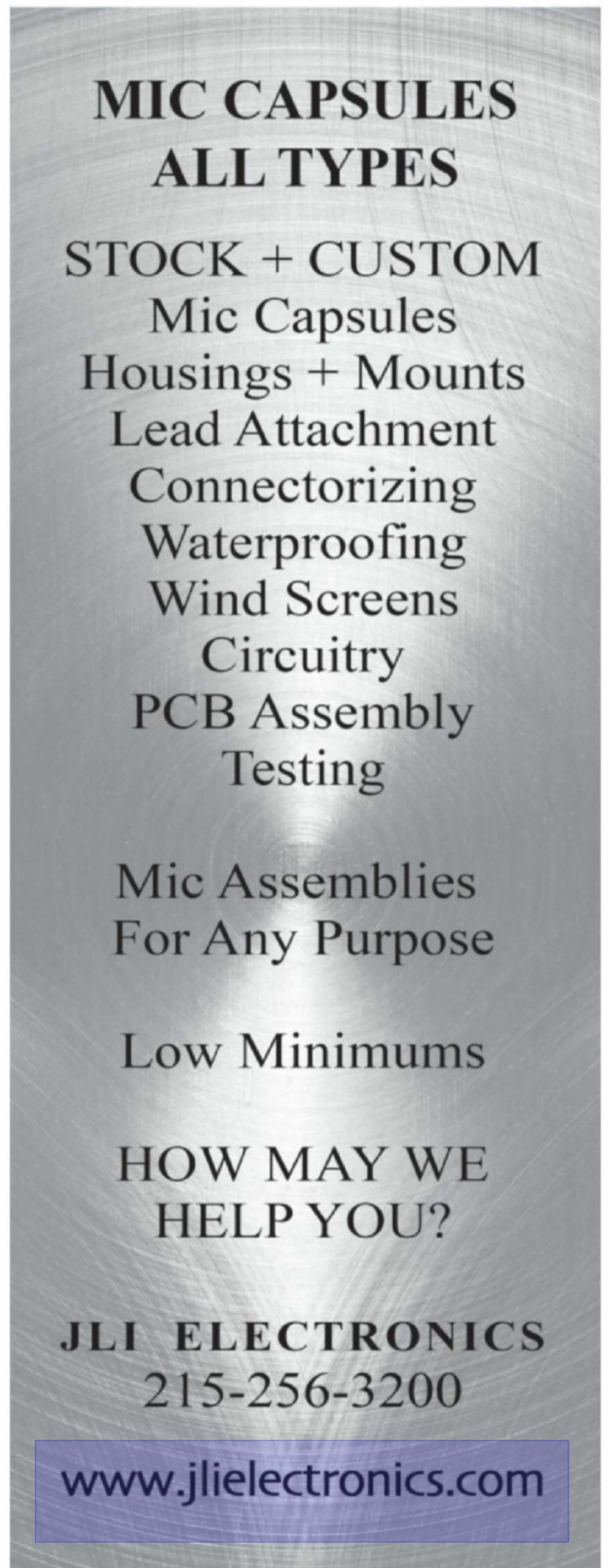
We have described a variety of ways to measure the audio performance of voice controlled audio devices including smart speakers, headphones/hearables, and automotive infotainment systems. These measurements represent the convergence of tests from loudspeakers, headphones, microphones, telephones, hearing aids, and more, and in addition must be made with a complex open-loop signal path, which may involve smartphones, head units and signals stored in the cloud.

When selecting a test system, based on these requirements, a huge number of test stimuli, algorithms, analysis options, and signal paths options are necessary, not to mention many “useful-to-have” features such as integrated background noise and a test configurator that makes it easy to run many iterations of a test continuously. We have compiled a checklist of the functionalities that will likely be needed and should be carefully evaluated before selecting a test system. The check list of functionalities includes:

- Tests both the input and output components of the device (i.e. loudspeaker or earphones and microphones)
- Can create a compound stimulus with frequency and/or log chirp trigger to enable accurate open-loop testing
- Can be set to analyze only parts of the response signal, so that

About the Author

Steve Temme is founder and President of Listen, Inc., manufacturer of the SoundCheck audio test system. Steve founded the company in 1995, and for the past 25 years the company has remained on the cutting edge of research in audio measurement, regularly introducing new measurement techniques, algorithms, and hardware to enable testing of all kinds of devices ranging from basic transducers to complex audio systems. In recent years, Listen’s product development focus has been on creating the algorithms, interfaces and test methods to measure today’s many and varied wireless and cloud-communicating devices. Prior to founding Listen, Steve worked for many years as an acoustic test and measurement applications engineer at Brüel & Kjær, and also as a loudspeaker design engineer at Apogee Acoustics. He holds a BSME from Tufts University, has authored numerous papers on acoustic testing, and has lectured extensively throughout the world.



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BS EN 50332-1 Sound System Equipment – Headphones and earphones associated with portable audio equipment—Maximum SPL measurement methodology and limit considerations

ETSI ES 202 396-1: Speech and Multimedia Transmission Quality (STQ); Speech quality performance in presence of background noise; Part 1: Background noise simulation technique and background noise database

IEC 60118-7: Electroacoustics—Hearing aids—Part 7: Measurement of the performance characteristics of hearing aids for production, supply and delivery quality assurance purposes

IEC 60268-7: Sound system equipment – Part 7: Headphones and Earphones

IEC 61094-4, "Measurement Microphones Part 4: Specifications for Working Standard Microphones," November 1995

IEEE Recommended Practice for Speech Quality Measurements

IEEE 269: "Standard Methods for Measuring Transmission Performance of Analog and Digital Telephone Sets, Handsets, and Headsets"

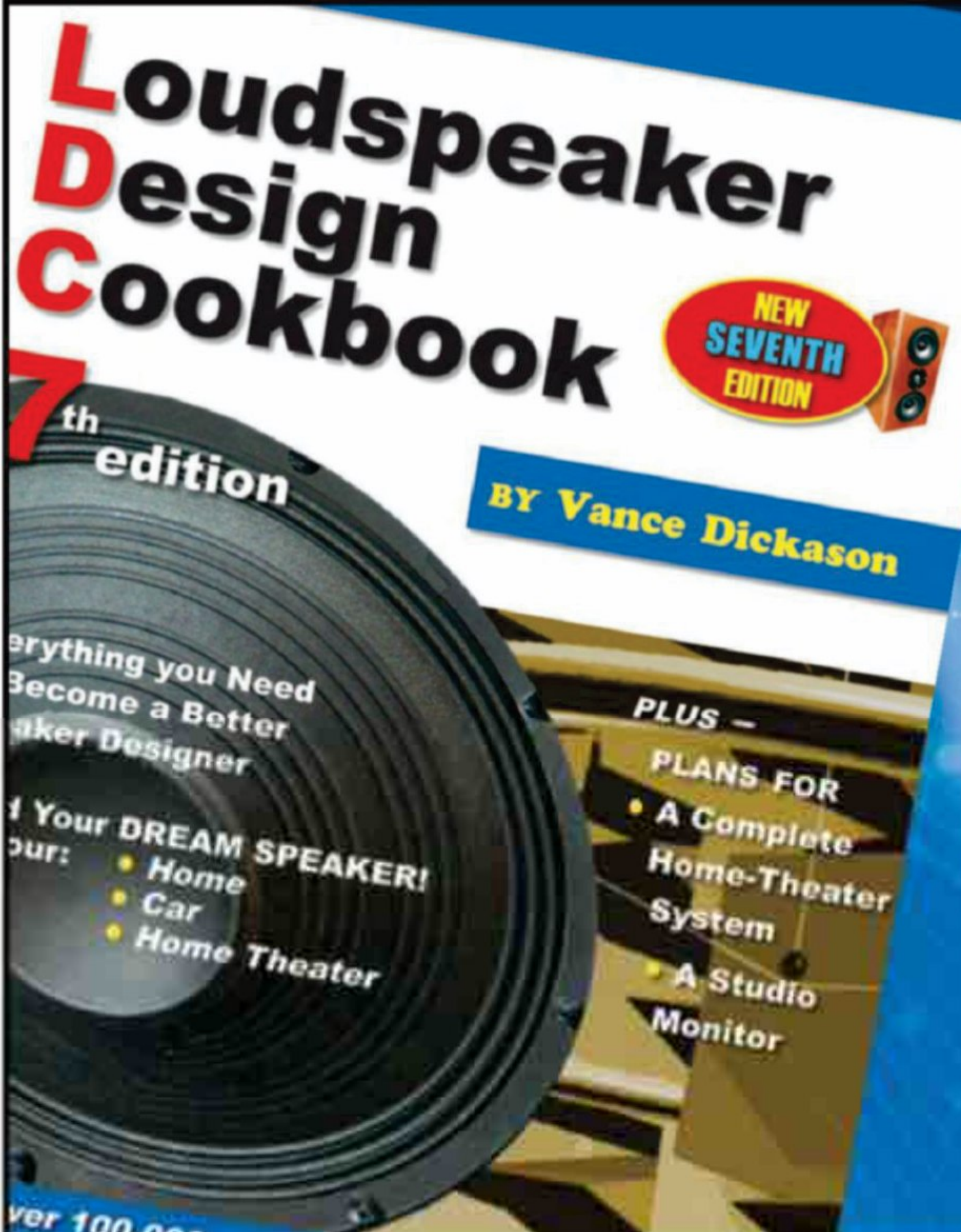
IEEE 1329-2010, "Standard Method for Measuring Transmission Performance of Speakerphones," October 2010

ITU-T Recommendation P.51, "Artificial Mouth," August 1996

TIA 920.130: Telecommunications Communications Products Transmission Requirements for Digital Interface Communications Devices with Headsets

the trigger tone can be eliminated from the analysis

- Able to use speech and music as test signals, calibrate levels, and equalize a mouth simulator (requires the inclusion of active speech algorithms)
- Background noise generation – although this can be generated externally, testing is much simpler and faster if it can be created by the test system and integrated with the test sequence
- Can accept signal via virtual audio cable in order to route in signals from the cloud—native Windows multimedia interface
- Availability of pre-written TIA test sequences if device needs to be tested to telephony standards
- Test programming flexibility – one that allows easy duplication of sub-sequences and loops is extremely helpful when repeating a test at various levels
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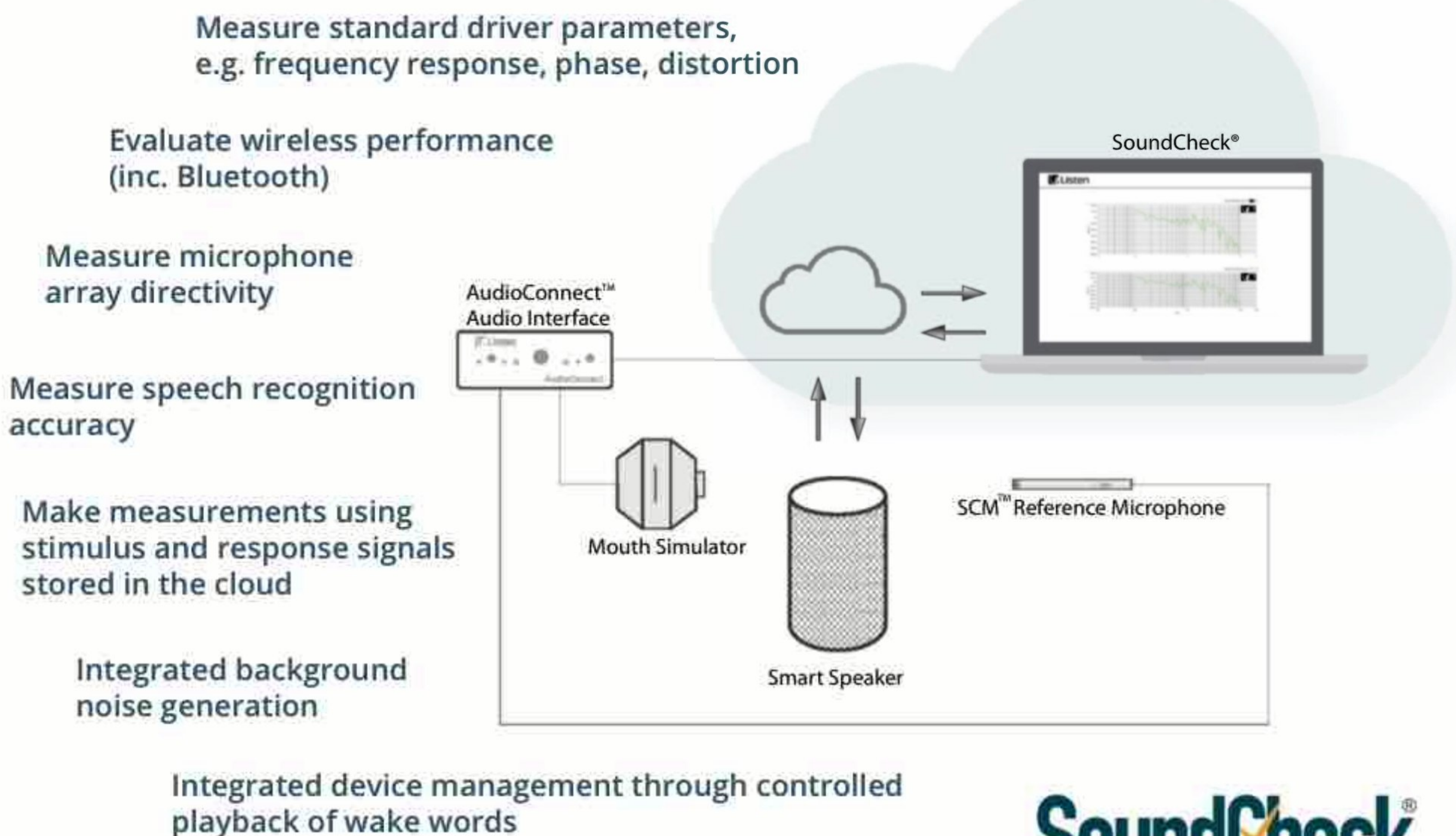
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Implementation is a snap, as the small, 56 mm x 30 mm Hardware Attached on Top (HAT) board fits directly on top of the Raspberry Pi Zero W board (see **Photo 1**). The board uses the same single 5 V supply that is already available on these. On-board power management and voltage boosting from the 5 V Raspberry Pi supply to 20 V ensures sufficient dynamic range and plenty of headroom margin to guarantee an undistorted open and natural sound that outperforms most off-the-shelf wireless speakers (see **Figure 1**).

No additional components are needed—so anyone equipped with a screwdriver can now turn their old loudspeakers into very powerful high-end wireless speakers.

The amplifier is extremely power-efficient and does not add much to the overall power budget of the Raspberry Pi Zero. From a power consumption and battery playback time perspective, the powerful audio amplifier addition is almost “free”—making the solution extremely suitable also for portable applications.

Everybody Loves Audio

While Infineon's MERUS integrated multi-level Class D audio amplifier ICs are rapidly gaining popularity with established audio brands and ODMs, there has also been steadily increasing interest from makers, tinkerers, hobby designers, and DIY enthusiasts who

are eager to build their own high-end audio devices and systems.

The highly power-efficient MERUS ICs are ideally suited for these compact audio amplifications as they can provide very high dynamic output range and the ability to play loud, but at the same time they do not require heatsinks or extensive output filtering.

Very often, however, it has proven too difficult for non-professional users to utilize the new digital input audio-amplifier ICs to their full potential—not because of the amplifier ICs per se—but because of other system components required to build a full-fledged audio system.

With the functions and features we have come to expect in a modern audio product (e.g., stereo pairing, multi-room wireless audio streaming, advanced audio processing capabilities, etc.), until now making a full active audio design solution has just been too complex a task for the average hobbyist to overcome.

The Raspberry Pi Revolution

As the world's third best-selling general-purpose computer, the Raspberry Pi is a phenomenon.

Since Esben Upton and his team revolutionized the computer industry back in 2011 with their first prototype of the affordable and flexible Raspberry Pi computer, it has become a lot easier for everyone to get access to a very powerful and yet inexpensive computer platform.

If you have any prior experience as an electronics hobbyist, chances are high that you already have one of these tiny British-made boards in your drawers somewhere.

Raspberry Pi boards are literally used everywhere in anything and

everything from laptops, household appliances, and robots to thin clients and industrial control systems. Rumor has it that they even use Raspberry Pi boards to run experiments onboard the International Space Station.

As such, the Raspberry Pi Zero W is a fantastic, miniature version of the Raspberry Pi that shrinks the board down to about the size of a stick of gum and packs in Bluetooth 4.0 as well as 802.11n wireless LAN connectivity, all at a very attractive price point of around \$10.

Tapping into a Free Software Ecosystem

Much of the success of the Raspberry Pi is grounded in its use of open-source software. The Raspberry Pi runs Linux on a multi-core ARM CPU, which is the same hardware that you will find in many tablets, smartphones, and set-top boxes and therefore, the operating system that on the Raspberry Pi also uses the same software repositories that powers the entire Linux software ecosystem. This means that you are tapping into the same code base that is running on almost 2 billion Android smartphones and a vast majority of the servers that are powering the Internet.

Building on 20 years of continuous software development you get a lot for free—also in terms of audio functionality. The audio stack on Linux is well-designed and allows applications to stream digital audio uncompressed to the hardware codec that is connected to the I²S audio bus. With the Pi HAT board, Infineon is providing a compatible low-level driver which ensures plug-and-play digital audio with its integrated multilevel Class D audio amplifier IC on the Pi HAT.

What is more, advanced users who want to push the limits of their DIY audio system even further can add digital-signal-processing to their audio systems. The Raspberry Pi platform is compatible with open-source and free-license audio programming frameworks (e.g., Pure Data and JUCE). So it is quite feasible for the advanced user to apply additional functionality and features such as speaker equalization, crossover filters, room acoustics correction or other DSP algorithms.

Making It Easy

Intended for the construction of active loudspeakers and wireless music streaming with minimum size, extremely low power consumption, and best-in-class HD audio quality, the MERUS audio amplifier reference boards are equipped with the MA12070P monolithic Class D multilevel amplifier IC.

Driven from a single 5 V supply and an onboard voltage-boost-converter, the system can provide up

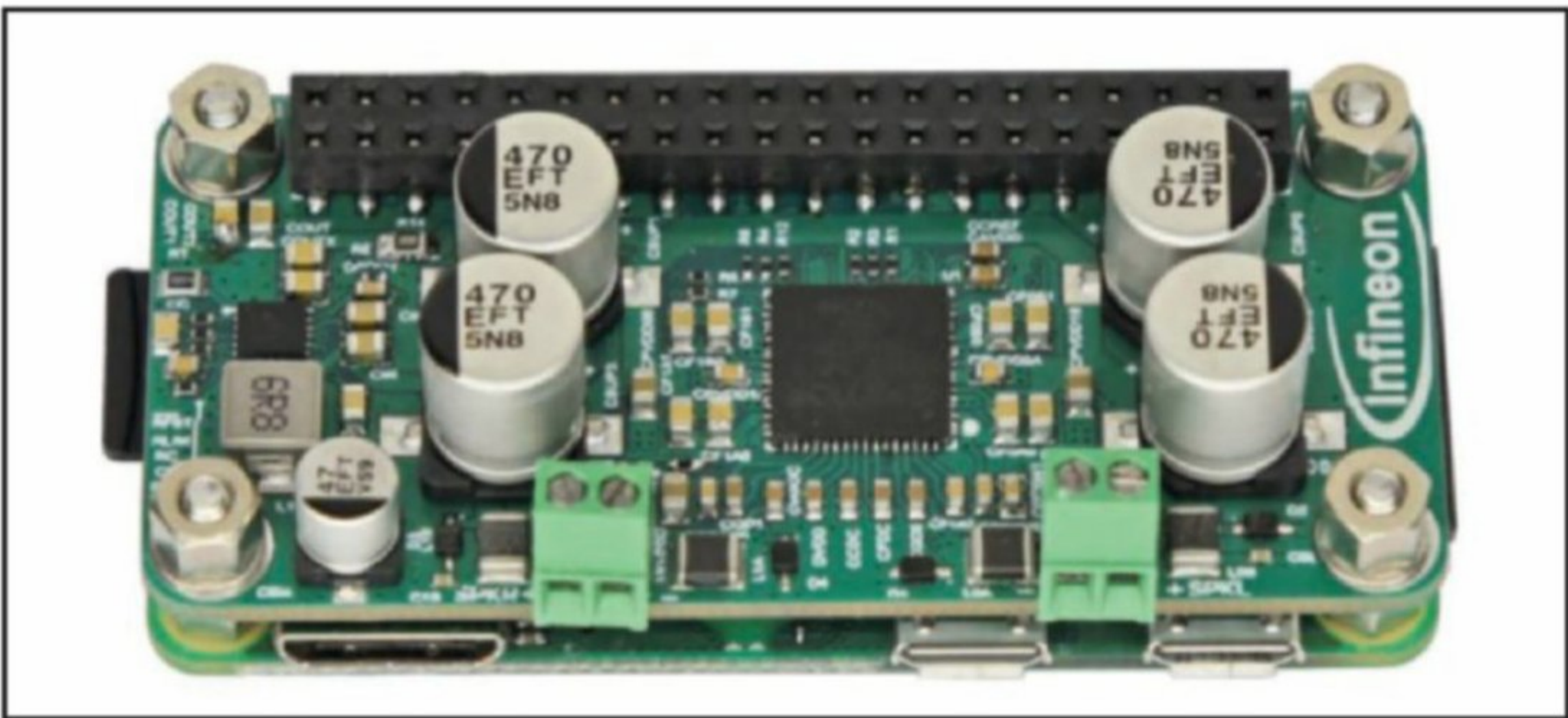


Photo 1: The small, 56 mm × 30 mm Hardware Attached on Top (HAT) board fits directly on top of the Raspberry Pi Zero W board.

to 2 × 40 W peak output power boosted from the Raspberry Pi supply with up to 48 kHz/24-bit music playback.

There is no need for additional power supplies as the MERUS audio amplifier attaches directly on top of the Raspberry Pi Zero W board and is powered directly from its 5 V supply rail—so no additional components required.

The single power supply is easily added by connecting a standard mobile phone charger via the onboard USB connector—and to build a truly wireless and portable system, you can simply add a power bank unit. The output power of the system is related directly to the peak output current of the power bank battery so make sure to use one with enough oomph.

A 5 V/2.1 A with 3000 mAh capacity would be a good starting point. This will give you roughly 10 hours of continuous playback time at 60% output volume from your phone.

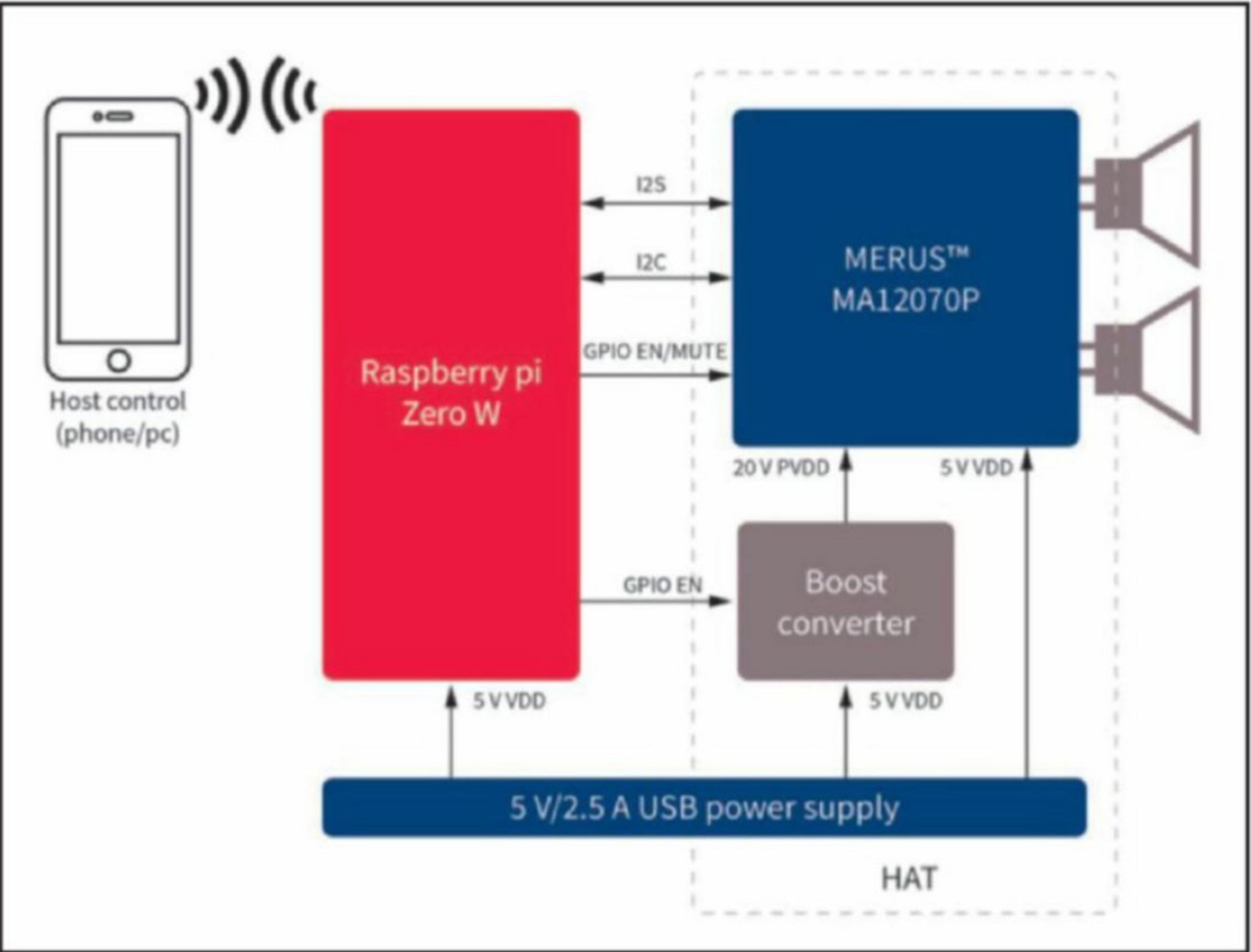


Figure 1: To get the system up running, all you need to do is to download the software on your PC, write it to an SD card which slots into the Raspberry Pi, attach the Pi HAT to your speaker(s) and plug in the power supply.

The effective playback time is dominated by the power consumption of the Bluetooth and WiFi functionality on the Raspberry Pi Zero W board, but there are also numerous possibilities to upgrade the battery for even longer playback time.

To get the system up and running, all you need to do is to download the software onto your PC, write it to an SD card which slots into the Raspberry Pi, attach the Pi HAT to your speaker(s) and plug in the power supply. In a few seconds, you will have a full-fledged high-end wireless Hi-Fi audio system.

The resulting design is compatible with the most popular audio streaming applications for Raspberry

Pi such as Justboom Player, Volumio, Max2Play, and it can be also used in Raspbian as an Airplay client or Bluetooth audio receiver.

This means that you can control the speakers from your phone or PC and pair them up in the configurations you want while streaming with Spotify, Google Music or SoundCloud to access Webradio, Podcasts, music from the cloud, or your own collection from a device anywhere in your network throughout your home or on the move.

Applications

Together, the Raspberry Pi Zero W and MERUS MA12070P Pi HAT amplifier boards make up a very strong combination that we foresee being used in a plethora of DIY audio applications such as smart speakers, soundbars, on-the-go speakers and intelligent home audio systems.

The Raspberry Pi Zero W boards are available from many sources and the Pi HAT Class D audio boards from Infineon are priced at \$29.95 for single quantities and will be available for sale in February 2020 at www.infineon.com/makers. For Infineon's product offering of multilevel Class D audio amplifiers, visit www.infineon.com/audiosolutions. 

About the Author

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

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An Essential Piece of Equipment for the Audio Enthusiast

Heavy-Duty Dummy Load

By
Bruce Brown
(United States)

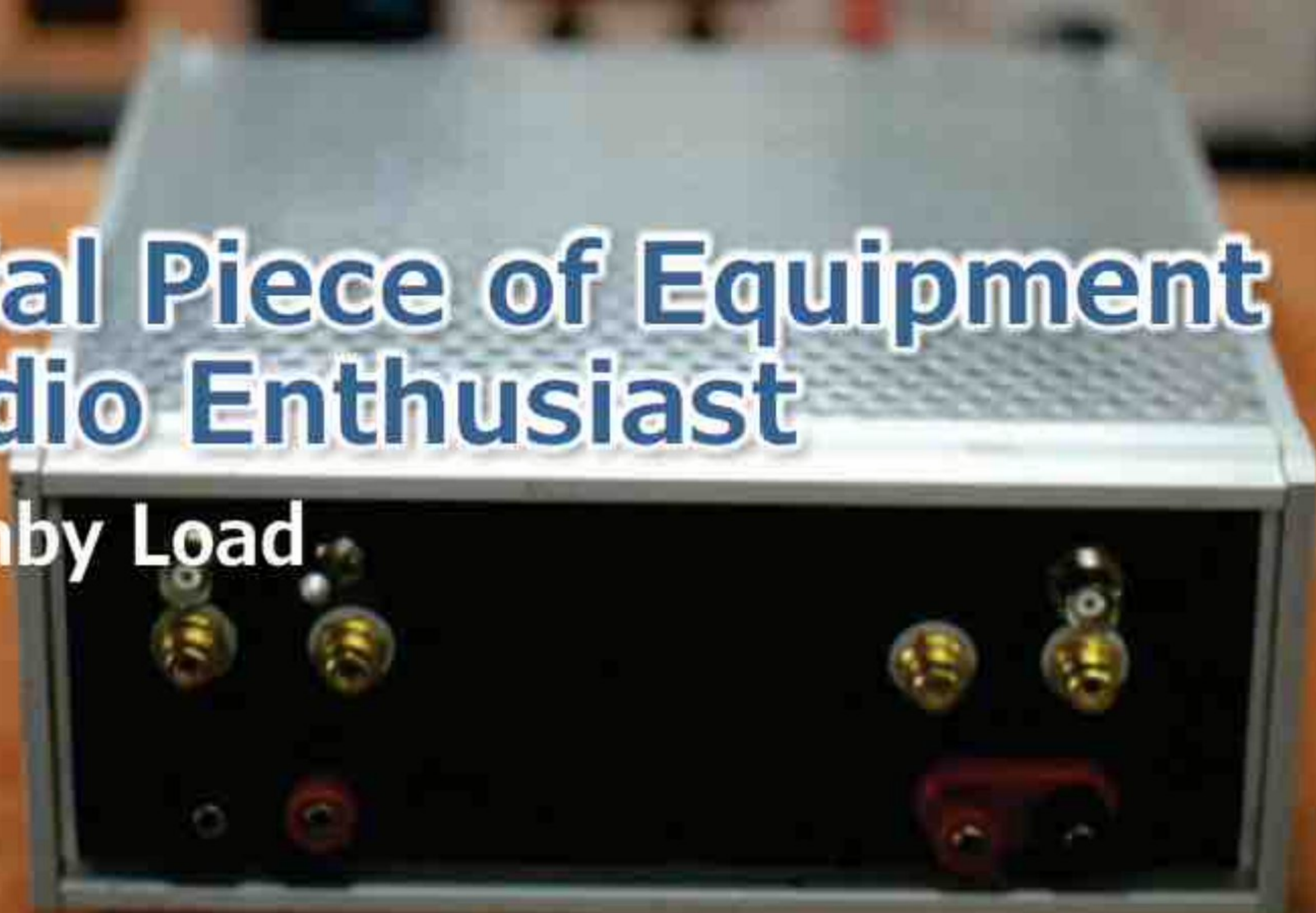


Photo 1: I built a heavy-duty dummy load to test how my amplifier handles large power levels.

If you are a regular reader of *audioXpress*, you might know that Bruce Brown restores vintage tube equipment, as well as select transistor stuff. There are a number of essential tools needed for evaluating and testing audio equipment. These tools include an oscilloscope, a good voltmeter, an audio generator (sine, square, and triangle waveforms) and last but not least a serious dummy load. This article will explain how to build such an essential piece of equipment.

To begin, a dummy (or dummy/dumbie) load refers to a simulated electrical load, frequently used in the testing of audio equipment or radio transmitters. Audio manufacturers commonly use dummy loads to test how an amplifier handles large power levels without driving a set of speakers at very high volume levels.

My Sencore Audio Analyzer has a built-in sine and square wave generator, a dummy load, and

a set of meters to monitor the RMS output of an amplifier. The only issue with it is that it only will measure 100 W RMS per channel. This is perfect for the majority of tube amps I rebuild, design, or restore, but for serious solid-state amplifiers it is far from ideal.

I recently finished rebuilding a Dynaco Stereo 400, and had to cobble together a number of large resistors to load it down. I was considering how to build a serious load center when I came across a flyer from Marlin P Jones and Associates (www.MPJA.com). The company deals in many surplus items as well as new parts and accessories. They had 16 Ω , 120 W resistors in the flyer and I began to formulate a plan to build my own heavy duty dummy load (see **Photo 1**).

The Construction

The construction is fairly simple and I have included a list of parts you can use to complete this project, even if you don't have a good supply of stuff laying around. The total cost of this build is less than \$50, and it is well worth the money.

The only thing that you will have to supply

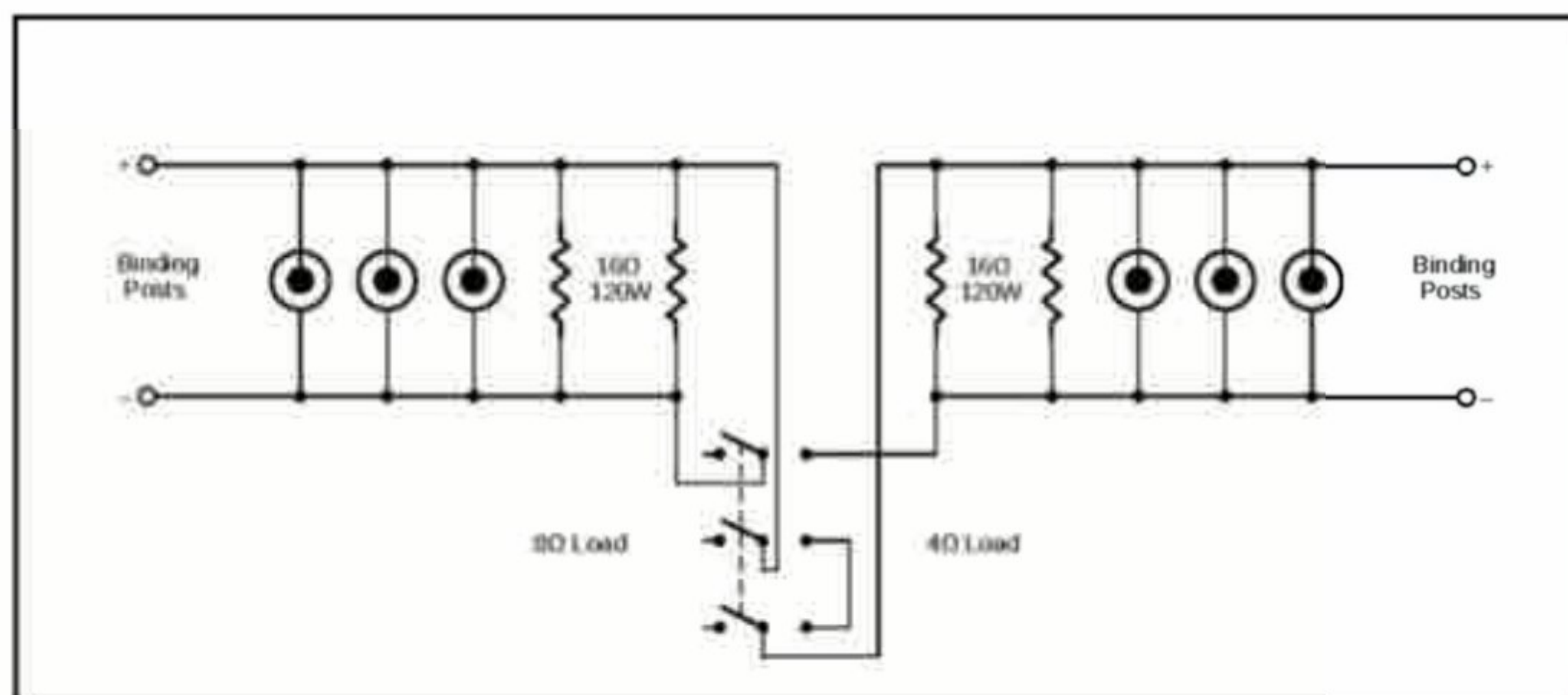


Figure 1: The schematic is quite simple, in one position the two 16 Ω resistors are not connected to the other two. In the second position, they are paralleled with the second two.



Photo 2: You can choose whatever you want for a case, but be sure you have enough space to keep the resistors away from each other so they can get adequate circulation.

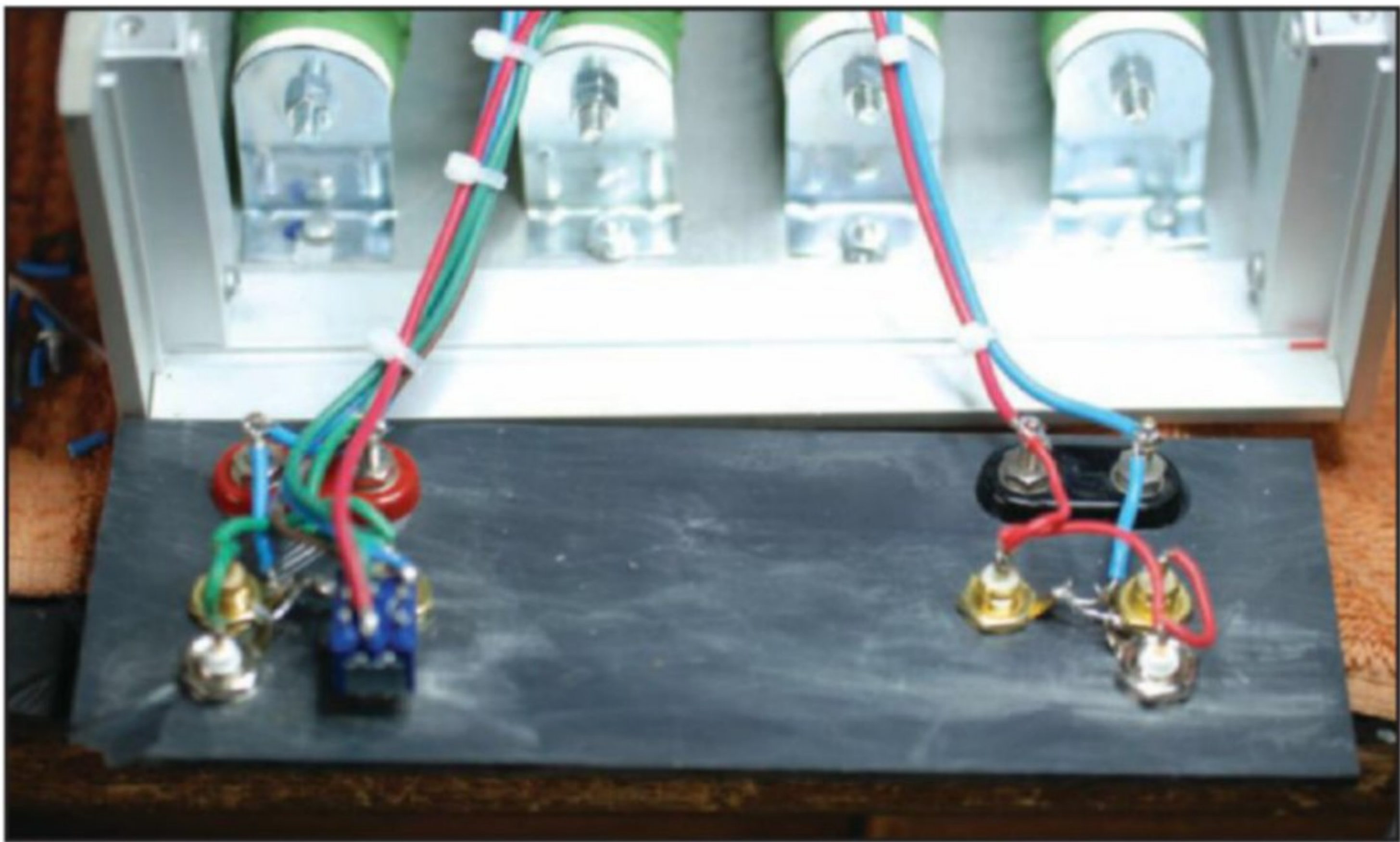


Photo 3: The jacks are all paralleled with each bank of resistors and then I added wiring for the switch.

Parts List

Amount	Parts	Cost	Total Cost
2	Dual binding posts (14492PL)	\$1.95	\$3.90
4	RCA Phono Jacks (33145 and 25522)	\$0.41	\$1.64
2-4	BNC Jacks (20507)	\$1.00	\$4.00
4	16 Ω , 120 W resistors (17789)	\$6.95	\$27.80
1	3PDT Toggler Switch (16076)	\$0.88	\$0.88
			Grand Total: \$38.22

*Not included: Case of your choice

on your own is an enclosure for the unit. If you aren't concerned about aesthetics, you could just mount the resistors on a wooden base and screw an aluminum panel to the front for the jacks and single switch.

In its most basic form, you could just wire two of each of the resistors in parallel for an 8 Ω load. With the addition of a 3PST toggle switch, you can have either an 8 Ω stereo load rated at 240 W or a single 4 Ω load at 480 W (a Mono load).

The schematic (see **Figure 1**) is quite simple, in one position the two 16 Ω resistors are not connected to the other two, in the second position they are paralleled with the second two. Because some stereo amplifiers (Class D and T) can't have a common ground, both banks are completely isolated. When switched to the 4 Ω position, both banks are paralleled including their grounds. This is a 4 Ω mono load!

I am fortunate in that I have tons of stuff around my house that I have collected for years. The case I chose for the dummy load is one I picked up from a surplus store in St Louis, MO. It was originally some kind of test equipment case and it just happened to be the right size for the four resistors. You can choose whatever you want for a case, but be sure you have enough space to keep the resistors away from each other so they can get adequate circulation (see **Photo 2**). You could also substantially increase the power handling ability of the unit by mounting them in a sealed case and filling the case with mineral oil (shades of Heathkit Antenna).

I chose to use binding posts for the amplifier inputs and added jacks for my oscilloscope and BK AC meter. Use as many or as few as you might find necessary, with whatever you find useful in for your situation. The jacks are all paralleled with each bank of resistors and then I added wiring for the switch (see **Photo 3**). The top of the original case was solid, but I wanted more ventilation, so I cut a piece of expanded aluminum to cover it.

Check Your Work

It's easy to check your wiring, simply connect an ohmmeter across each of the binding posts. You should have a reading of about 8 Ω on each set of binding posts. Flip the switch and you will have around 4 Ω at each set of binding posts. This is important: Remember that the 4 Ω load is a mono load! Never have a stereo amplifier connected to both sets of binding posts and have the switch in the 4 Ω mono load! If you do not heed this warning, you might destroy your amplifier.

Using the Dumb Load

So how do you use your new piece of equipment? You will need an AC voltmeter that reads in RMS (most new meters do this, even very inexpensive ones); an oscilloscope (nothing fancy or exotic); and a sine, square, and triangle wave generator (again, nothing fancy). Dan Joffe at UpdatemyDynaco.com sells a very inexpensive ultra-low distortion sine wave generator in a kit form that works very well.

Connect your amplifier outputs to the binding posts, connect your AC voltmeter and oscilloscope to the appropriate jacks, and connect your signal generator to the inputs of the amplifier under test. Turn on amplifier and let it warm up for a few minutes.

Slowly bring up the level on the signal generator and watch the oscilloscope and meter. You should see a nice sine wave and get an indication on the meter. Continue advancing the generator signal until the sine wave starts to flatten. This is the beginning of clipping. It is not a good idea to allow the amplifier to run on a sine wave at full power for a long time. If you have a triangle wave, you can determine the exact level of clipping more

accurately. Once you have established where the amplifier begins to clip, you can read the meter voltage and calculate the RMS watts with the following formula:

$$\text{RMS voltage} \times \text{RMS Voltage/Impedance of the load} = \text{RMS Watts}$$

So if you read 45 V, then 45×45 divided by $8 \Omega = 253 \text{ W RMS}$, which is a pretty hefty amplifier! 

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.

About the Author

Bruce W. Brown has been experimenting with electronics and building stuff since the 1960s. He started his own business at age 15, installing house intercom and stereo systems, and building Heathkit and Dynaco stereo amps for sale to friends and family (or anyone who wanted to buy one). He worked repairing stereo equipment for an audio store during his years in Pharmacy School. He has been a long time tube amplifier builder and restorer as well a contributor to *Glass Audio* and *audioXpress*. He restores tube amplifiers, as well as constructs custom amplifiers at his business Vintage Tube Audio (Vintagetubeaudio.com). He is happy to reply to questions at Tuninfork@aol.com.



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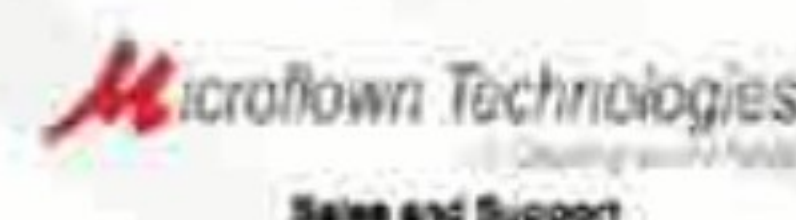
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Repairing the Oldies

After noting some of the antique radios waiting for repair around his home, our columnist decided it was time to provide information that might prove helpful to readers who want to take on some of their own antique repair projects.

Photo 1: This hand-wired under-chassis from an antique radio highlights some possible differences between the components found in older hollow-state gear and those in more modern equipment.

By
Richard Honeycutt
(United States)

Although my electronics time is mostly spent on audio projects, I do have experience in broadcast radio (including working as student engineer maintaining 50 W AM hollow-state carrier-current transmitters in college), and I hold an advanced-class ham license.

My high-frequency ham gear is hollow-state. Also, my lab is presently home to a 1940s Puritan table radio, my dining room hosts a late-1930s Philco floor-standing radio, and my living room contains both a 1924 Atwater-Kent radio that needs a new power switch and a 1939 Philco table radio that needs new filter caps.

The Puritan, the Atwater-Kent, and the floor-standing Philco are not functioning at present. Someday, I will fix all these treasures, as well as maybe the other antique radios in my attic and storage building.

I suspect that many of our readers have similar projects awaiting them, perhaps along with some old Western Electric, RCA, Stromberg-Carlson, or DuKane hollow-state audio gear, and maybe some hollow-state test equipment. But, there are some possible differences between the components hand-wired in antique hollow-state gear (see **Photo 1**) and those in equipment built since—say—1950. This article discusses some of these differences, providing

information that might prove helpful if you should indulge the urge to undertake some of those antique repair projects.

Obsolete Components

First among the possible differences are components that are no longer common. One of these is the selenium rectifier, used to supply B+ in some old hollow-state circuits (see **Photo 2**). In operation, selenium rectifiers had a forward voltage drop of 5 V to 10 V—well below the 10 V to 25 V drop of tube rectifiers. In replacing a selenium rectifier with a solid-state diode, you should add a series resistor that will drop about 10 V in normal operation, to limit the effect of turn-on surges. Selenium rectifiers age, with both reverse leakage current and forward voltage drop increasing through the years. Like modern solid-state rectifiers, selenium units had two failure modes: shorts and opens. It is a good idea to replace selenium rectifiers in any old equipment you repair, because if one of them shorts, it could catch fire.

Most resistors used in modern electronic equipment are either carbon-film, metal-film, or wirewound types. In contrast, most old hollow-state gear used primarily carbon composition resistors. These were composed of a mixture of powdered carbon and clay, then baked and painted (see **Photo 3**). Carbon-comp resistors tended to increase in resistance with age and heat. Exposed to a hot environment (e.g., inside the chassis of hollow-state equipment), the resistance increase can range from a few percent to thousands of percent. I once replaced a 100 k Ω plate resistor in an amplifier whose measured value had risen to 5 M Ω !

The Capacitors

Before plastic film capacitors, there were wax-paper capacitors. These did not fail very often, because even if a voltage surge should arc through the wax paper between the foil plates, the

Photo 2: The selenium rectifier is one component that is no longer common.



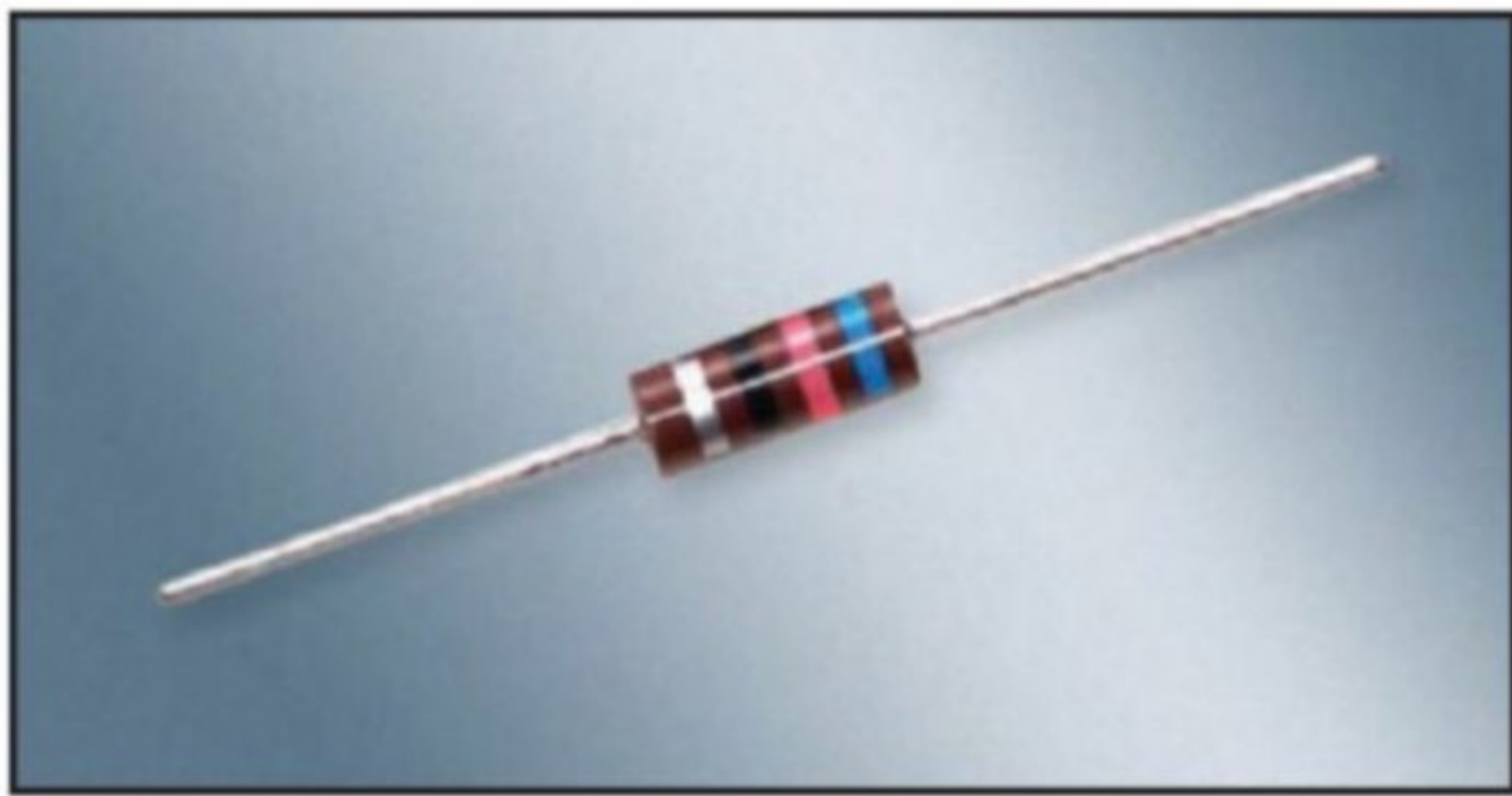


Photo 3: Carbon composition resistors were the primary type used in old hollow-state gear.



Photo 4: Here are four tubular molded paper-oil capacitors and at the far left of the image, a wax-paper capacitor.

wax would melt and fill the gap, “healing” the puncture. These caps were subject to drift in capacitance over time, though, as the wax oozed from between the foil plates, decreasing spacing and thus increasing capacitance and somewhat decreasing breakdown voltage. Thus, many servicemen routinely replaced bulged wax-paper caps as a preventive measure, although they might be functioning perfectly.

Similar to wax-paper capacitors were paper-oil capacitors. These were always packaged in sealed cases. **Photo 4** shows four tubular molded paper-oil capacitors and at the far left of the image, a wax-paper capacitor.

Photo 5 shows a metal-cased unit. Many antique hollow-state devices used caps similar to the one shown in Photo 5 as B+ filter caps. These had much longer lives than the electrolytic capacitors with which they were replaced, but the electrolytics were smaller and less expensive. Electrolytic capacitors and even film caps are subject to defects such as dielectric absorption that have been shown to affect signal integrity when audio signals are passed through the caps. Paper-oil capacitors are better in this respect. However, in addition to size and cost, paper-oil caps had the disadvantage that in years past, some used a toxic oil.

An electrolytic capacitor uses two plates—typically an aluminum outside can and inner aluminum foil—that are immersed in an electrolyte. In the earliest units, the electrolyte was salt water. “Paste-type” or “dry” electrolytics were patented by Samuel Ruben and Phillip Mallory in 1925, but “wet” electrolytics continued to be used until World War II. When voltage was applied from

one plate to the other, a thin layer of insulating aluminum oxide formed on the positive plate. Because of the thinness of the oxide layer, the capacitance was much higher for a given size cap than one could attain using other technologies. The electrolyte served as an extension of the negative plate.

Photo 6 shows a disassembled wet electrolytic cap. At left in the photo is a cardboard spacer; in the center is the loosely coiled positive plate; and at right is the can that serves as the negative plate. You can see a perforated plastic insulator inside the can. Its purpose is to prevent the capacitor from shorting if the equipment is unused for a long time and the oxide dielectric de-forms. (Turning the equipment on and waiting for a few minutes to a half-hour would re-form the dielectric.) The terminal at the top of the center assembly passes through a rubber seal intended to keep the electrolyte in the capacitor. However, with age, the rubber could dry out and fail to seal, allowing the electrolyte to escape.

In the old days, service persons sometimes used a hypodermic needle to inject replacement electrolyte. Today, most of us just replace dead wet electrolytics with new dry ones. If an authentic original appearance is desired, the wet electrolytic can be opened and a dry one of the same value (usually physically smaller) can be inserted inside the can.



Photo 5: Many antique hollow-state devices used caps similar to the one shown here as B+ filter caps.



Photo 6: The parts of a disassembled wet electrolytic cap include: at left, a cardboard spacer; in the center, the loosely coiled positive plate; and at right, the can that serves as the negative plate.

The Tubes

In addition to the classic 4-pin and 5-pin-based tubes, there were some tube types made in the 1940s and earlier that resemble, but are not interchangeable with, some modern tubes. Two examples are the 6L6 (with no alphabetic suffixes), and the 6K6 (see **Photo 7**). The 6L6 is similar to the 6L6G, GA, GB, and GC, WGB, and WGC, of which only the 6L6GC remains in production. All these tubes are similar, but the plate voltage and current ratings and the plate dissipation vary among the different types. For example, the 6L6 is rated for a maximum plate voltage of 360 V, and maximum dissipation of 19 W, whereas the 6L6GC is rated at 500 V and 30 W, respectively. Although there are other slight differences, in general a 6L6GC can be used to replace a 6L6 but not vice-versa.

The 6K6 is a power pentode designed for use in single-ended output stages of radios and TVs. It is similar in appearance to

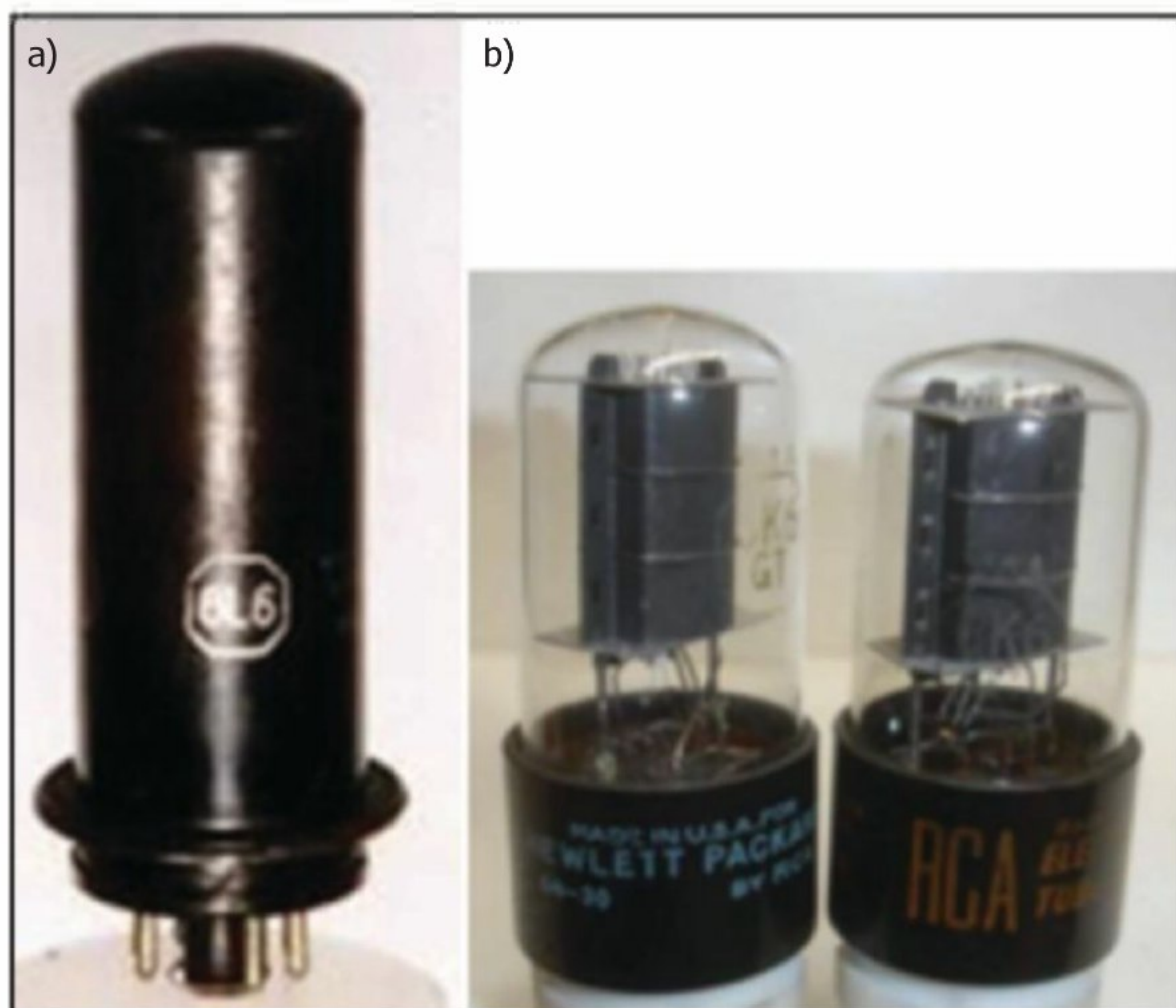


Photo 7: There were some tube types made in the 1940s and earlier that resemble, but are not interchangeable with, some modern tubes. Two examples are the 6L6 (a) and the 6K6 (b).



Photo 8: The field coil was often used as a filter choke in the B+ circuit.

Photo 9: A failure mode of vacuum rectifiers in the old days, but seldom seen today, was “gassing,” or air having leaked into the tube.



the 6V6, but should not be used as a 6V6 replacement, although the reverse is OK. For more information see the discussion on the Antique Radios forum at: <https://antiqueradios.com/forums/viewtopic.php?t=100348>.

The Field-Coil Speaker

The last old-guard component I’ll mention is the field-coil speaker (see **Photo 8**). Before 1940, better performance could be obtained at lower cost by using an electromagnet rather than an Alnico permanent magnet for a speaker. While it is possible to replace a field-coil speaker with a modern permanent-magnet speaker, you must be aware that the field coil was often used as a filter choke in the B+ circuit. So in addition to replacing the speaker, you’ll need to add an appropriate filter choke: typically 5 henries (H) to 10 H or more.

An alternative is to use a smaller choke and increase the filter caps (usually one to ground on each side of the field coil) with a larger value. These are often 8 μF to 10 μF each. If you replace these with new caps having as much capacitance as possible, you may be able to replace the field coil with a power resistor having about the same resistance. This is often about 700 Ω . Choose a resistor with a power rating at least twice the product of the square of the B+ current times the value of the resistor. If you use a capacitor or a resistor that is too small, you will hear hum from the speaker.

A failure mode of vacuum rectifiers in the old days, but seldom seen today, was “gassing,” or air having leaked into the tube. Electrons would strike the air molecules, ionizing them and causing both a glow between the elements in the tube and popcorn noise in the speaker. A gassy rectifier tube is shown in **Photo 9**. Often power tubes having a high plate voltage (6L6GC, 6550, EL34, etc.) may show a blue glow that is not to be confused with gassy operation. This glow is generated on the glass envelope itself, due to fluorescence, and is normal.

In Conclusion

Even though it is common for old-timers to mourn the passing of the “good old days,” I must admit that I have no desire to return to leaky wet electrolytic capacitors and sticky wax-paper capacitors. However, it is good to know about the old components when we delve inside antique hollow-state electronic equipment!



Here are a few places where you might find a copy of *audioXpress* and possibly meet one of our authors and staff members.

January 2020

January 7-10

CES 2020

Las Vegas Convention and World Trade Center (LVCC) and 10 other locations in Las Vegas, NV
<https://www.ces.tech>

January 16-19

The NAMM Show 2020

Anaheim Convention Center
 Anaheim, California, USA
www.namm.org/thenammshow/2020/attend

February 2020

February 11-14

ISE 2020

Amsterdam RAI, The Netherlands
<https://www.iseurope.org>
www.avixa.org | www.cedia.net
www.iseurope.org

February 15-16

CanJam NYC

Marriott Marquis, New York, NY
<https://canjamglobal.com/shows/nyc2020>

February 24-27

GSMA Mobile World Congress

Fira Gran Via, Barcelona, Spain
www.mobileworldcongress.com
www.mwcbarcelona.com

March 2020

March 18-20

electronica China

International Trade Fair for Electronic Components, Systems, and Applications
 China Import & Export Fair Complex,
 Guangzhou, China
 Shanghai New International Expo Centre, China
www.electronica-china.com

March 21-22

CanJam Singapore

Pan Pacific Singapore, Singapore
<https://canjamglobal.com/shows/singapore2020>

March 27-29

Salon Audio Montréal/Montréal Audio Fest

Hotel Bonaventure, Montréal, Canada
www.montrealaudiofest.org

March 31-April 5

Prolight+Sound Frankfurt

April 1-4

Musikmesse

Frankfurt Messe Exhibition Center,
 Frankfurt, Germany
www.musikmesse.com
www.prolight-sound.com
www.messefrankfurt.com

April 2020

April 17-19

Audio Expo North America (AXPONA)

Renaissance Schaumburg Hotel &
 Convention Center, Schaumburg, IL
www.axpona.com

April 18-22

NAB Show 2020

Las Vegas Convention Center (LVCC),
 Las Vegas, NV
www.nabshow.com

May 2020

May 2-3

CanJam Shenzhen

InterContinental Shenzhen, Shenzhen, China
<https://canjamglobal.com/shows/shenzhen2020>

May 11-15

179th Meeting of the Acoustical Society of America (ASA)

Chicago, IL
acousticalsociety.org

May 14-17

HIGH END Show

MOC Munich-Germany
www.highendsociety.de

May 25-28

Audio Engineering Society (AES) Vienna 2020

148th Audio Engineering Society
 International Pro Audio Convention
 Austria Center, Vienna
www.aes.org/events/148

June 2020

June 5-7

Lone Star Audio Fest

The Embassy Suites Dallas Park Central,
 Dallas, TX
www.lonestaraudiofest.com

June 10-12

CES Asia

Shanghai, China
www.cesasia.cn

June 13-19

InfoComm 2020

Exhibits: June 17-19
 Las Vegas Convention Center, Las Vegas, NV
www.infocommshow.org

June 14-15

Audio and Loudspeaker Technologies Professionals ALTI Expo 2020

South Point Hotel & Casino, Las Vegas, NV
<https://almaint.org>

June 20-21

CanJam SoCal

Irvine Marriott, Orange County, CA
<https://canjamglobal.com/shows/socal2020>

July 2020

July 9-11

Summer NAMM

Music City Center, Nashville, TN
www.namm.org/summer/2019

July 18-19

CANJAM London

Park Plaza Westminster Bridge, London, UK
www.canjamglobal.com/shows/london2020

July 23-25

2020 AES Audio Education Conference

Nashville, TN
www.aes.org/conferences/2020/education

July 24-26

California Audio Show

San Francisco, Silicon Valley, CA
www.caudioshow.com

August 2020

August 7-9

Hong Kong High End Audio Visual Show

Hong Kong Convention & Exhibition Centre (HKCEC)
www.audiotechnique.com
www.audiotechnique.com/av-show

August 17-19

3rd Audio Engineering Society (AES) International Conference on AVAR

DigiPen Institute of Technology,
 Redmond, WA
<http://www.aes.org/conferences/2020/avar>

August 24-27

SET EXPO

Exhibits: August 25-27
 International Technology Fair on
 Equipment and Services for Television
 Engineering, Broadcasting, and
 Telecommunications
 Expo Norte, São Paulo, Brazil
www.setexpo.com.br

September 2020

September 4-9

IFA Berlin (The International Funkausstellung)

www.ifa-berlin.com

September 6-8

PLASA Show London

London, Olympia, UK
www.plasashow.com

September 8-12

CEDIA Expo

Exhibits: September 10-12
 Colorado Convention Center, Denver, CO
www.cediaexpo.com

September 11-15

IBC Show

RAI Amsterdam, The Netherlands
www.ibc.org

September 17-19

Prolight+Sound NAMM Russia

Sokolniki Exhibition and Convention
 Centre, Moscow, Russia
www.namm-musikmesse.ru
www.prolight-namm.ru



Industry Calendar

September 24-26
Consumer Electronics China (CE China)
Guangdong, China, China
www.ifa-berlin.com
<http://www.cechina-ifa.com>

October 2020

October 2-4
Rocky Mountain Audio Fest (RMAF)
Gaylord Rockies Resort & Convention Center, Aurora, CO
www.audiofest.net

October 10-11
CanJam Chicago
Marriott Marquis Chicago, Chicago, IL
canjamglobal.com/shows/chicago2020

October 13-16
HKTDC Hong Kong Electronics Fair (Autumn Edition)
Hong Kong Convention & Exhibition Center, Hong Kong
www.hktdc.com/hkelectronicfairae

Dates TBD
The Loudspeaker Sourcing Show 2020
Nansha Grand Hotel in Guangzhou, China
www.loudspeakersourcingshow.com

October 21-24
AES New York
149th International Convention
Jacob K. Javits Convention Center, 655 W 34th St., New York, NY
Exhibition: October 21-23
www.aes.org/events/149

October 21-22
NAB Show New York
Jacob K. Javits Convention Center, 655 W 34th St., New York, NY
www.nabshowny.com

October 28-30
GSMA Mobile World Congress Americas
Los Angeles Convention Center (LACC), Los Angeles, CA
www.mwcamericas.com
www.gsma.com | www.ctia.org

November 2019
Dates TBD
Capital Audiofest
Hilton Hotel at Twinbrook Metro, 1750 Rockville Pike, Rockville, MD
www.capitalaudiofest.com

November 7-8
CanJam Shanghai
Shanghai Marriott Hotel City Centre, Shanghai, China
www.canjamglobal.com/shanghai2020

November 9-13
180th Meeting of the Acoustical Society of America (ASA)
Cancun, Mexico
acousticalsociety.org

Dates TBD
New York Audio Show
Park Lane Hotel, Central Park South, New York, NY
www.chestergroup.org/newyorkaudioshow

Dates TBD
Live Design International (LDI) Show
Las Vegas Convention Center, Las Vegas, NV
www.ldishow.com

Dates TBD
Inter BEE
Inter BEE 2019 | International Broadcast Equipment Exhibition
Makuhari Messe, 2-1, Nakase, Mihama-ku, Chiba City, Chiba Prefecture Japan
www.inter-bee.com

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A Constant-Regulated
Element Supply
By Christopher Paul

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Acoustics—Building
Michael Wenzel's
By Richard Wenzel

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Bob Cordell's Designing Audio
Power Amplifiers, 2nd Edition
By Dan Dinkley

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Bringing Old
to Life: Restored
Audio and Video
By Phil Saville
and Thomas Evers (Eds.)

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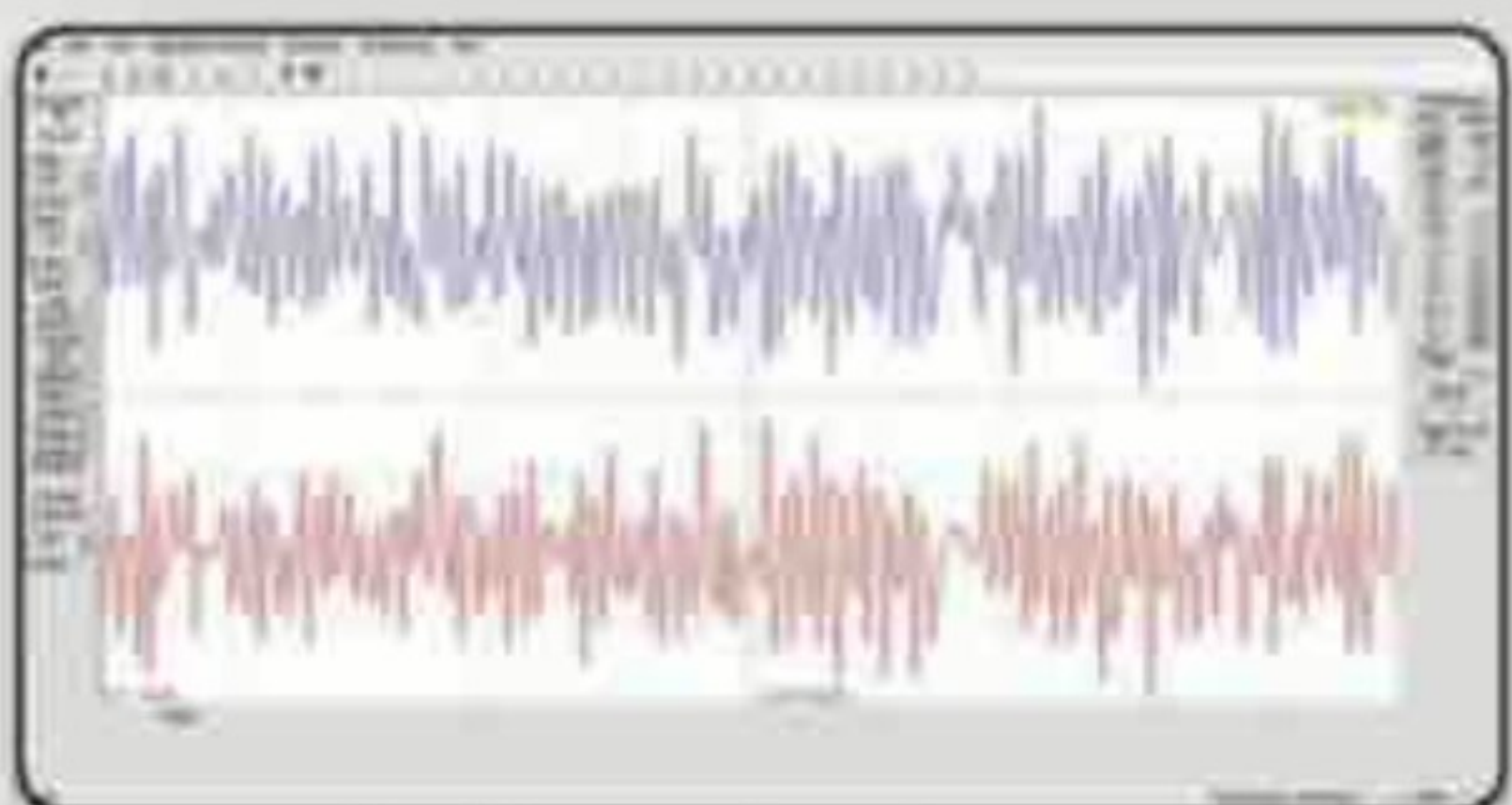
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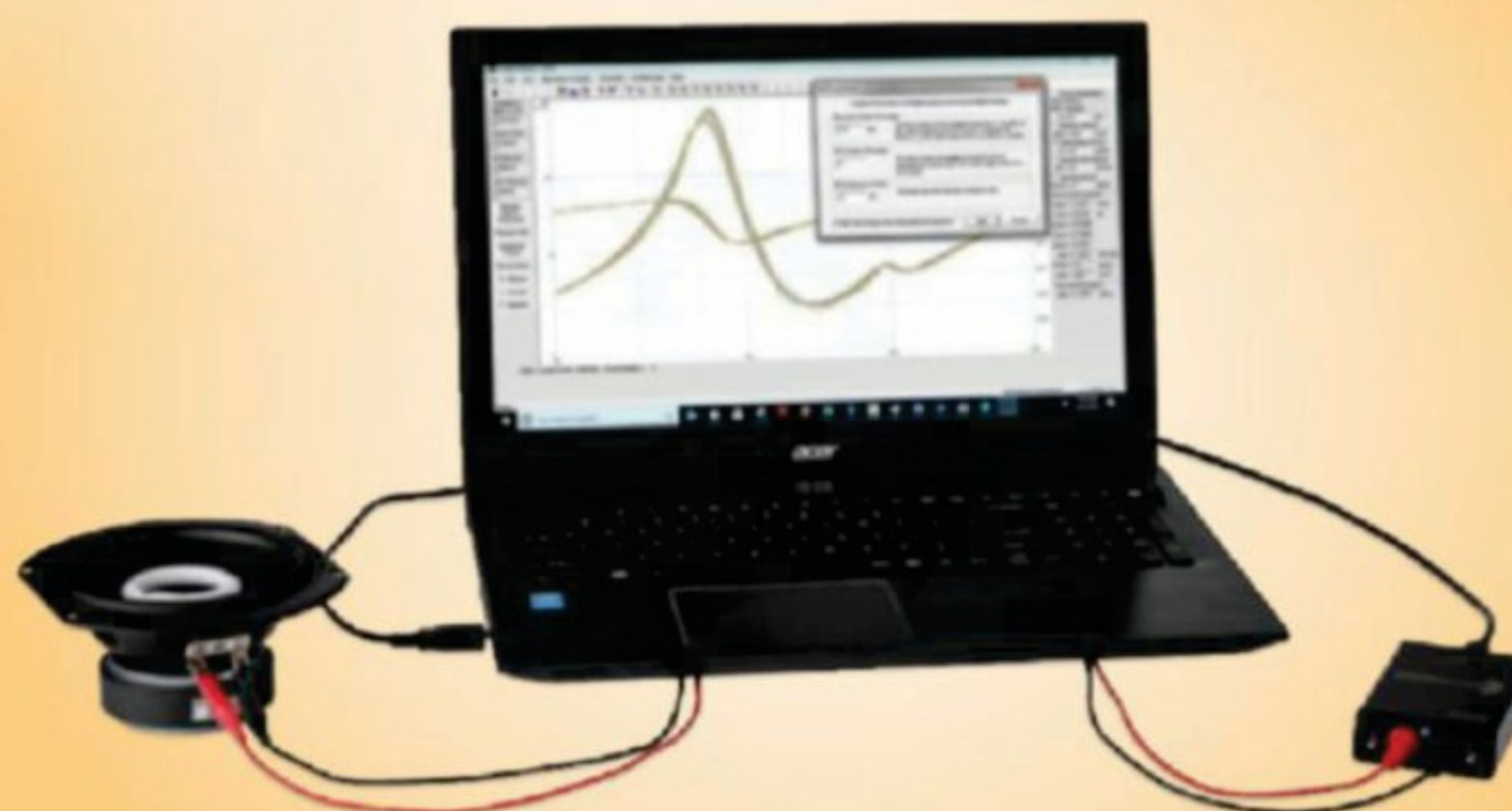


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