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ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

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Integrated, Networked, and Secure**

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Speaker Design Competition**

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Installation No Longer a Grey Area

When in 2013 I set upon the challenge of revitalizing *audioXpress* and expanding its editorial mission toward research, development, and innovation in the audio industry, I announced that we would not limit the scope of this magazine to any specific vertical market or application segment. Of course, installation for commercial, and residential areas was on our minds, not only because it was already one of the fastest growing segments in the audio industry, but more importantly because it was one of the most innovative. Yes, innovative.

It's changed quite a lot from the traditional—and in many ways stagnated—market of 70 V/100 V lines, boardrooms, and public address boxes we had until the mid 1990s. Digital audio introduced the first wave of radical change, but it wasn't until networking technologies took over and the concept of real systems integration matured that things started to expand exponentially.

Also exponential was the growth in trade shows from the installation and systems integration side—with Integrated Systems Europe (ISE) becoming larger than the InfoComm and CEDIA shows combined, since 2015—a direct reflex of systems integration companies globally taking over the audio segment. The system integrators did not take the business away from existing audio-focused companies. Instead, as the market significantly expanded, many audio companies have simply decided to focus on the specialty segments, such as entertainment systems—which have also got much bigger and become much more sophisticated—while other companies are more dedicated to the commercial, building, and public spaces applications, have done exactly what the market was telling them to do, and evolved to an integrated approach in systems. They have added network infrastructure, video, lighting, control and some even integrated security systems. This was a significant shift for the audio industry and one that has not yet been fully understood.

The real story in the installation segment for the audio industry was a transition from a product-focused approach to a systems approach. And that's where technologies such as audio-over-IP (AoIP) and networked-based systems quickly gained momentum, together with faster convergence with IT-based solutions and even fast-evolving consumer technologies (e.g., wireless protocols and data interfaces). The scale, sophistication, and level of reliability required audio signals to move to the network, where new concepts such as distributed amplification and signal processing quickly gained new meaning.

I still remember when audio manufacturers started toying around with the idea of software-based audio mixers for the live sound performance industry. Basically, they started to build those systems around powerful computers with more or less traditional control surfaces. The next step was leveraging the same platforms for digital signal processing, and the notion of network connectivity naturally followed. Curiously, many faced resistance in their intended markets, and it was the installation industry that quickly embraced the concept, introducing network amplifiers with native network transport control and monitoring capabilities. There was no going back.

On the more basic transducer level, audio continues to be originated in microphones and the signal is reproduced in loudspeakers. But while in other application segments (e.g., studio recording or live performances) there is still resistance to changing well-understood standards and the familiarity of professional audio tools, the installation industry is increasingly embracing the notion that the network interface and corresponding data protocols need to be pushed forward, directly integrated in the microphone with direct connectivity to the loudspeaker. Creating IP-addressable sources and end-points has introduced significant advantages and possibilities that are starting to be well understood.

The next step will see a convergence with consumer technologies (MEMS, sensors, AI) and the smart home domain (IoT, cloud), where technology is also evolving at a faster pace. But the bigger impact is the realization that the quality of audio has also increased. When visiting the audio demo rooms at professional shows such as ISE or InfoComm, the demonstrations are using the same audio material that we often see used in high end shows such as AXONA or the Rocky Mountain Audio Fest (RMAF). The resulting impression is that of listening to superb high fidelity, just in much larger sizes and at louder levels (ironically for the same or lower price...).

J. Martins  
Editor-in-Chief

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COLUMNISTS

**Vance Dickason** has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—which is 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.

**Mike Klasco** is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike specializes in materials and fabrication techniques to enhance speaker performance.

**Dr. Richard Honeycutt** fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject while 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.

**Steve Tatarunis** has been active in the loudspeaker industry since the late 1970s. His areas of interest include product development and test engineering. He is currently a support engineer at Listen, in Boston, MA, where he provides front-line technical support to the SoundCheck test system's global user base.

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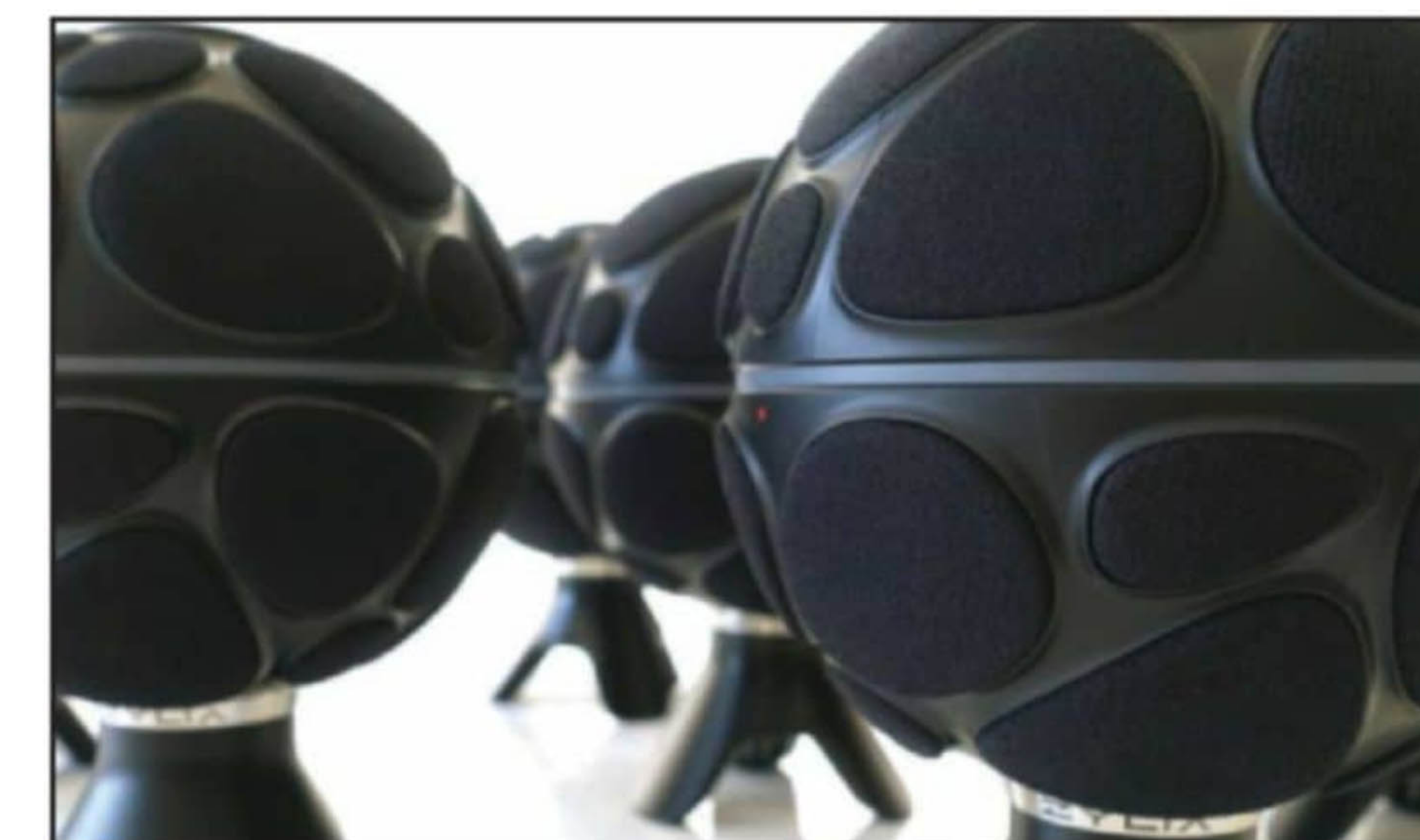
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It may be time to rethink your Class D audio design and measurement practices. Learn more about what multilevel Class D amplification can do for your products from expert Jens Tybo Jensen, who is head of Class D Audio Marketing and Applications at Infineon.

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## ● Market Update—Audio Installation

### 20 Integrated, Networked, and Secure

By J. Martins

In this market update *audioXpress* provides an overview of product and technology trends in the audio installation market. Given the size of the market and number of product categories, in this update we are focusing on the most cutting-edge technologies for commercial and corporate installations.

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## COVER ART

Photo credit: Cederquist installation with Genelec speakers in Stockholm, Sweden. (Photo courtesy of Genelec)





# 2019 Midwest Audiofest Speaker Design Competition

By  
**Thomas Perazella**

(United States)

Photos courtesy of Parts Express

Photo 1: Parts Express provided a new listening room for the Midwest Audiofest Design Competition.

One of the most important audio events every year is the Midwest Audiofest (MWAf) held in Springboro, OH, in July. Sponsored by a group of vendors who are listed on the website for MWAf 2019 and hosted by Parts Express, this event combines several activities including a huge tent sale at Parts Express. That sale is essentially a walkthrough of a special part of the company's warehouse plus tents full of hugely discounted merchandise. In addition, there are tables set up where participants can sell their "gently used" equipment, but Thomas Perazella says for him, the highlight of the event is the Speaker Design Competition. In the past, he has reported on the entire MWAf so this year he concentrates on the Speaker Design Competition.

I often get asked why people build their own speakers when there are so many very fine commercially available choices in all price ranges. This goes to the heart of the do it yourself (DIY) philosophy, where several factors are very important.

First is the learning experience where every time you build something new you learn new materials and techniques. Second is the ability to tailor the project to get the outcome you want, not what some product development person thought you should have. Third is the bragging rights you have when you show your new speaker to admiring friends. Fourth is the camaraderie that results from collaborating with other like-minded DIYers.

All of this combined talent and energy comes together every year at the MWAf. When you are there and listen to the interactions of both the new and also very experienced builders you realize the power of this event to generate enthusiasm at a very high level. This not only relates to good sound but often exceptional artistic and construction skills. I

have been auditioning very high quality commercial speakers and also building my own speakers for many years yet am really amazed by what I see and hear every year at the MWAf.

## What is the Speaker Design Competition?

The purpose of this event is to provide builders with a chance to show their speakers in a forum that is controlled and also tailored to allow for different levels of speakers. There are four categories that contain all the entries. Since complete details are available on the MWAf site, I will summarize.

Open Unlimited Category: This is a no-holds-barred category where any design is allowed. There is no limit on the number or types of drivers used, the housing design, or the cost, and it is the only category that allows active crossovers, any type of EQ, limiters, and time alignment. In theory, this should be the highest performing category sonically.

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Photo 2: The judges for the 2019 Midwest Audiofest were Tom Perazella, Vance Dickason, and Jerry McNutt.

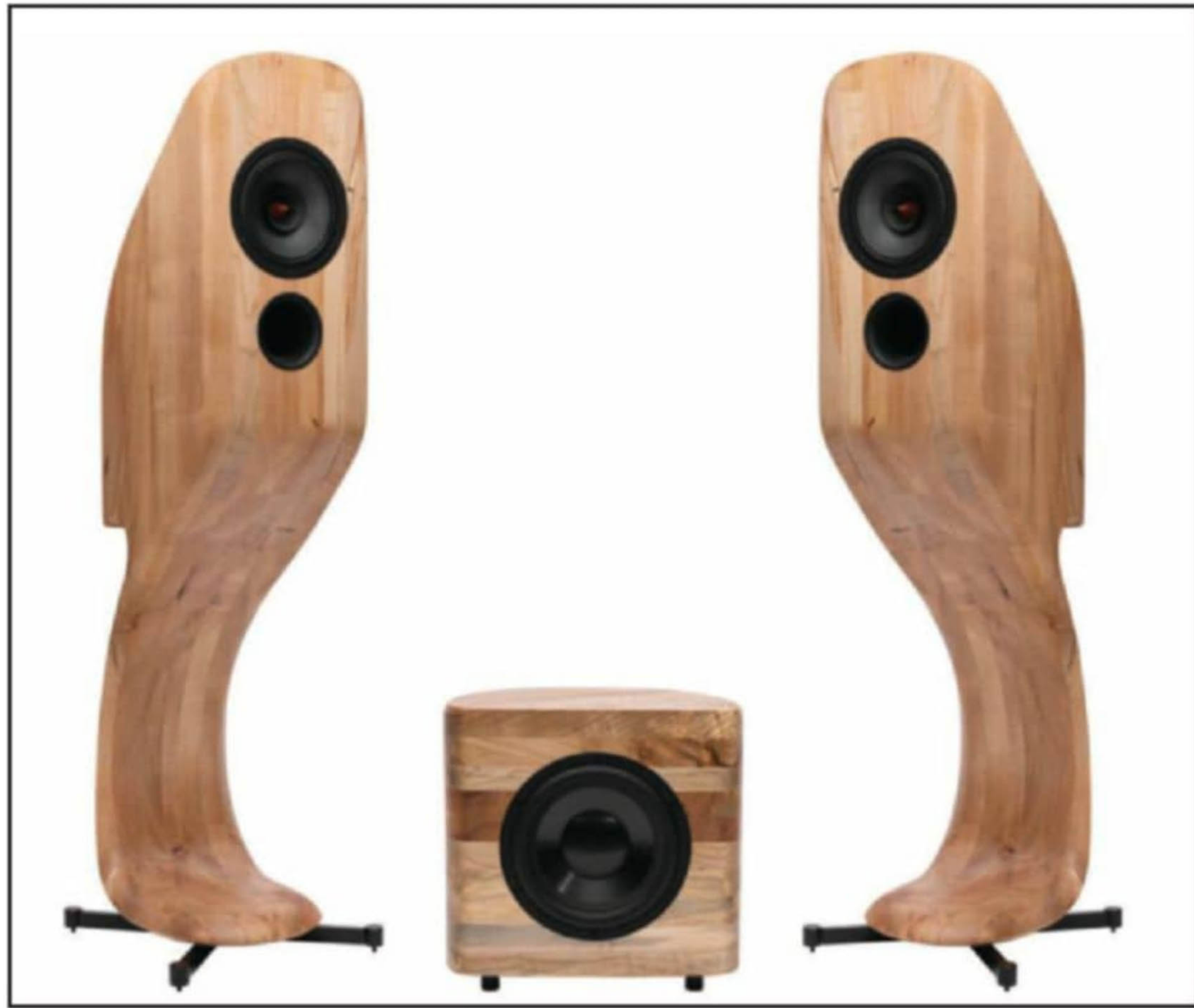


Photo 3: For the most striking form and finish I think the entry Troy—Maple Notes from Barrett Niekamp takes the honor.

## About the Author

Thomas Perazella is a retired Director of IT. He received a BS from the University of California, Berkeley campus. He is a Past President of the Rockville Chapter of the Izaak Walton League of America, one of the oldest national conservation organizations in the US and currently is the Treasurer. Audio has been his passion for more than 50 years and he is a member of the Audio Engineering Society, the Boston Audio Society, the Philadelphia Area Audio Group, the DC HiFi Group, and the DC Audio DIY Group. He has written for *audioXpress* magazine and prior to that for its predecessor, *Speaker Builder*. In addition to audio, his interests include photography, cooking and competition pistol shooting. He has authored several articles in professional audio journals and taught commercial lighting at the Winona School of Photography. Recently, he received a patent on a cost-effective high-efficiency LED lighting system for commercial and residential buildings.

**Dayton Audio Category:** This category allows any number or types of drivers, any cabinet design, any cost, but the crossovers if used must be passive with no EQ or other modifications allowed. In addition, the drivers must be of the Dayton Audio brand.

**Over \$200 Category:** Like the Dayton Audio category, the choice of drivers is unlimited but allows any brand. The only caveat is that the total cost of drivers is greater than \$200. Again, strictly passive crossovers are required. You might ask why this category forces the cost of drivers to be over \$200. The answer is to separate the entries from the last category which follows.

**Under \$200 Category:** I am always reminded of “The Little Engine That Could” when I think about this category. The overriding criteria in this section is that the cost of all drivers used for the stereo pair must retail for under \$200. Again, although any housing design can be used, the crossovers must be passive with no EQ or other electronic modifications.

When I first learned of the Under \$200 category, what immediately popped into my mind was “you must be kidding me.” On the surface it seemed like a nearly impossible task to produce a good sounding speaker with that restriction. As I auditioned some of the entries in this category while judging my first MWAf, I realized that there are some very talented DIY speaker designers out there. What is also critical are the tools now available to model the various choices to be made and the tremendous gains in driver performance as a result of new materials and designs. It is very possible to be successful with that price restriction.

## How the Competition Works

An entrant can submit speakers to as many categories as desired. The competition is divided up into four time slots, one for each category. This year the competition was held in a new venue that was much larger, which helped minimize early ceiling and wall reflections. It also allowed for more exhibitors, guests, and other interested people to listen critically (see **Photo 1**).

Ranking of the entries is done by a panel of three judges. This year the panel consisted of Jerry McNutt, Vance Dickason, and myself (see **Photo 2**). Background information on the judges is available on the MWAf site. Personally, being able to work with Jerry and Vance who both have huge amounts of experience and also are great guys is a definite plus for me. As part of the process, each judge provides three 1-minute music selections for each category to the event coordinator at Parts Express. From those, one selection from each judge is used

for each category during the judging. Therefore, each judge has one familiar selection for reference in each category. The list is made public before the event and I make sure I listen to all the final selections many times before the judging. The goal is to become familiar with the music so that you can listen for clues that match the selection criteria used. Those criteria are listed on the MWAf site.

The competition begins on a Friday evening right after a meet-and-greet period where the participants can rub elbows and talk about their projects. The first session for Open Unlimited is always run by itself because of the extra time that may be needed in the setups of the speakers often with their own amplification and processing.

The next morning starts the first of the three remaining categories. The speakers are all set up on the same spot in the room with locating tape strips on the floor for reference. The table with the three judges is centered on the speaker positions and about 12’ back. Spectators are seated in rows that begin about 6’ behind the judges. The entire listening area for the speakers, judges, and spectators is surrounded by heavy curtains to minimize visual and auditory distractions. Source equipment used is listed on the MWAf site.

## Designs of Note

As with any speaker designs, there are varying opinions on which look and sound the best. People are also impressed with uniqueness of design and types of finishes used. Before getting to the final ratings I would like to mention a few speakers I considered noteworthy.

For the most striking form and finish I think the entry Troy—Maple Notes from Barrett Niekamp takes the honor (see **Photo 3**). Made from recycled maple wood planks, the final result was uniquely shaped and finely finished. Unfortunately, it was entered into the wrong category so the separate subwoofer could not be used, adversely affecting the sound.

What at first appeared to be a sophisticated computer power supply turned out to be the entry Nebulous Snails from Clay Allison (see **Photo 4**). These are portable battery-operated speakers with internal amplification and DSP. The side panels are acrylic with artistic patterns and backlighting. Quite a visual impression.

The Defiants from Keith Etheredge are a conventional two-way in a rectangular housing. What sets them apart is the tapered front panel finished in faux blue snake skin. The tapered grille nicely accents the front panel (see **Photo 5**).

The Side Towers from Bill Schwefel take a standard rectangular housing and use it in a non-conventional way (see **Photo 6**). Instead of the longest dimension being vertical it is horizontal with the drivers mounted on the small end. This minimizes the size impact of the speaker when viewed from the front. The driver mounting surface is curved and absorptive material is mounted around the drivers to minimize diffraction.

A striking combination of shape, wood choice, and driver color was presented by The Safaris from Chris Biese (see **Photo 7**). Basically, a rectangle but with tapered sides, a Zebrawood veneer



Photo 4: What at first appeared to be a sophisticated computer power supply turned out to be the entry Nebulous Snails from Clay Allison.



Photo 5: The Defiants from Keith Etheredge are a conventional two-way in a rectangular housing.



Photo 6: The Side Towers from Bill Schwefel take a standard rectangular housing and use it in a non-conventional way.



front panel, and white driver cones and domes, the visual effect was very appealing.

The Cornettas from Paul Kittinger were medium-sized towers that took advantage of several shapes and woods to break the tower mold (see **Photo 8**).

Photo 7: A striking combination of shape, wood choice, and driver color was presented by The Safaris from Chris Biese.



Photo 9: These are the Moon Drops, a winning design from Nick Santorineos.

The main body was constructed of a glossy red wood with a rounded top that followed the lines of the drivers. Accenting the center section were two blonde wood panels with the final touch being a brass name plate on the front.

What appeared at first glance to be white columnar-shaped speakers with a flat front and tapered sides turned out to be a speaker with a PVC pipe section and wood mounting surfaces finished not in white but a metallic pearl type of automotive paint (see **Photo 9**). Lots of clear coat and sanding. They were called Moon Drops and were from Nick Santorineos.

Representing a totally different approach with a very high degree of originality and tongue in cheek humor was the Easter Eggs with Big Woofer



Photo 8: The Cornettas from Paul Kittinger were medium-sized towers that took advantage of several shapes and woods to break the tower mold.

from Meredith Cargill. The photo says it all (see **Photo 10**). This entry came complete with two trees and branches with nests holding the eggs with drivers. The tweeters were mounted on the left side in a chick and on the right in a duck. If you look carefully you can see the “time aligning unit” on the top of the woofer. The supplemental graphs for the left and right side were appropriately labeled not L and R but chick and duck. Meredith always supplies a much needed break in the middle of hours of testing. He received a round of applause for his detailed and very humorous presentation of the philosophy involved.

There are lots of other speakers that deserve mention but to get the entire picture you really have to be at the competition.

### And the Winners are . . .

In the Open Unlimited category, first prize was taken by Clay Allison with his entry Nebulous Snails. In second place was Cherry Bomb by Jack Putti. Third place was the Reference Mini by Brian Zheng (see **Photo 11**).

In the Dayton Audio category, Chris Biese took



Photo 11: Open Unlimited winners, L to R—Brian Zheng, Clay Allison, and Jack Putti



Photo 13: Over \$200 winners, L to R—Javad Shadzi, Dan Neubecker, and Adam Malito

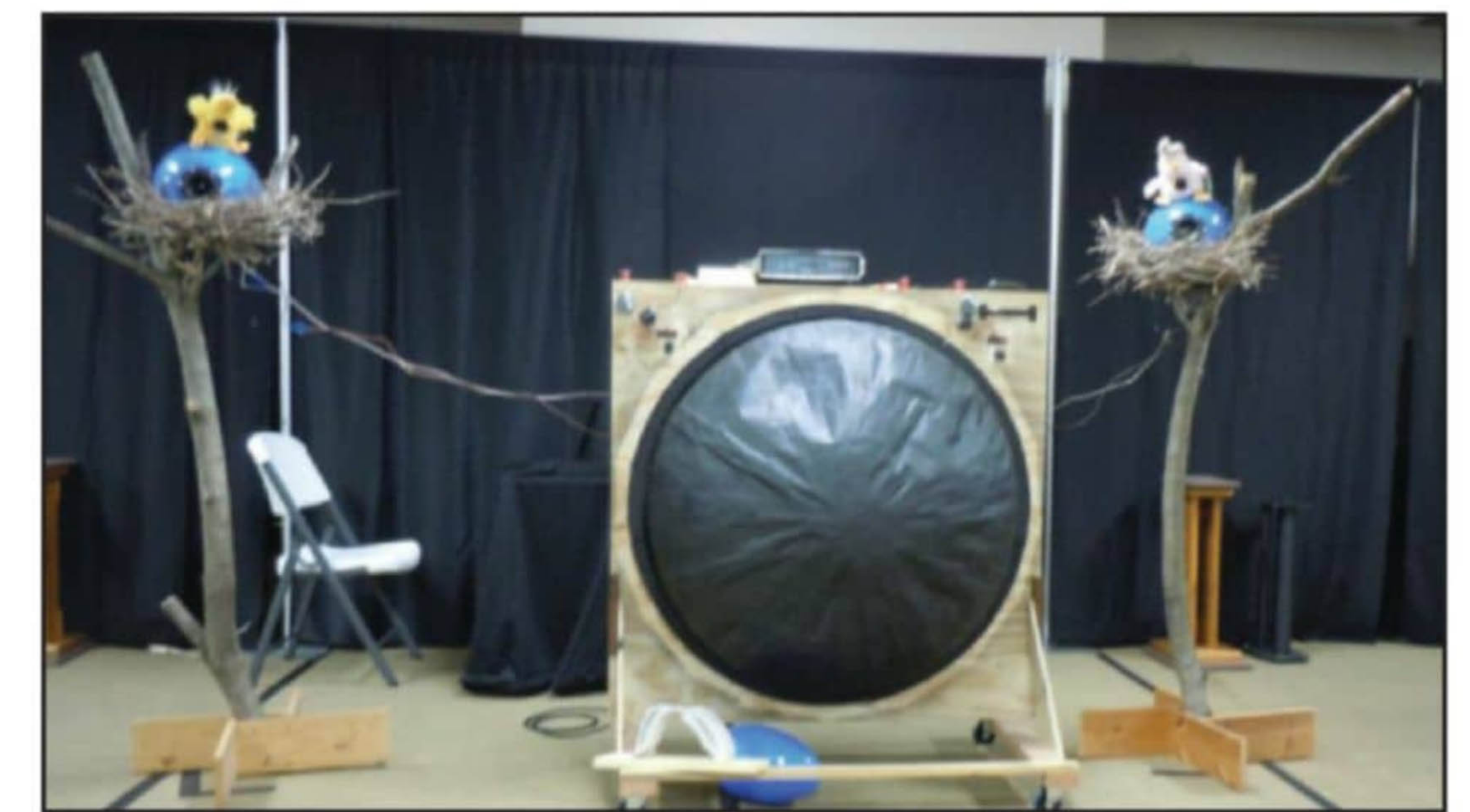


Photo 10: Check out the Easter Eggs with Big Woofer from Meredith Cargill. (Thomas Perazella photo)

first place with his entry RSX. John Hollander took second honors with C-Cake. Third was Troy—Maple Notes by Barrett Niekamp who as previously mentioned was hampered by being entered in the wrong category (see **Photo 12**).

The Over \$200 category has historically had some of the best performing entries and is also difficult to judge. The first three positions were very close together in total points. The Mystiques by Dan Neubecker took top awards. Second was Javad Shadzi



Photo 12: Dayton Audio winners, L to R—John Hollander, Chris Biese, and Barrett Niekamp



Photo 14: Under \$200 winners, L to R—Javad Shadzi, Nick Santorineos, and Bill Schewfel





Photo 15: Chris Biese took first place in the Dayton Audio category with his entry RSX.



Photo 16: The number two overall winner came from the Over \$200 category and was Mystiques by Dan Neubecker, with a total score of 166.

with the Enthraals. In third was Provare by Adam Malito (see **Photo 13**).

Again, the Under \$200 category not only had a large number of entrants, but also some of the best performers even at the low price. First place was Moon Drops by Nick Santorineos. In second place was Javad Shadzi with Rivalries. In third was The Side Towers by Bill Schewefel (see **Photo 14**).

### A Look at Some Numbers

Numbers will never tell you how a speaker will sound or if you will like it. They will give you an indication about how the judges felt about the entries and perhaps provide some surprises. So here goes. As far as number of entrants in each category, Under \$200 had 13, Over \$200 had 13, Dayton had 6, and Open Unlimited had 5, for a total of 37 that were evaluated. There were several more that originally registered but did not show. Let me tell you that evaluating 37 speakers in a little over a day with three pieces of music each is a very demanding task.

I decided to count the number of scores of 10 and 9 to give a sense of the performance of each category. Remember to look at the number of entries in each category when evaluating the high scores. The Over \$200 category had 29 10s and 78 9s for a total of 97 or divided by entries (which in reality was not evenly divided between the entries) yielded 7.5 high scores per entry. The Under \$200 category had 26

10s and 81 9s for a total of 106 or 8.2 high scores average per entry. The Dayton Audio category, where Chris Biese took first place with his entry RSX (see **Photo 15**), had 5 10s and 21 9s for a total of 26 or 4.3 high scores average per entry.

The Open Unlimited had 5 10s and 10 9s for a total of 15 or an average of 3 high scores per entry. Beware of averages because some entries had very high scores across all characteristics where some had the opposite. On average, it gives you an idea of the performance of the categories.

Just going by the total scores for each speaker regardless of the category produced the top three of the competition. Number one came from the Under \$200 category and was Moon Drops by Nick Santorineos with a total score of 167.5 out of a possible total of 180 points. Number two came from the Over \$200 category and was Mystiques (see **Photo 16**) by Dan Neubecker, with a total score of 166. In third, also from the Over \$200 category was Enthraals by Nick Santorineos with a score of 163.5. All were quite close and on a given day with different judges the results could change dramatically.

### Conclusions

The most important thing to know about this competition is that there are a lot of dedicated and skilled speaker builders out there who devote a ton of time conceptualizing, designing, modifying, and then building some outstanding speakers. In addition, the time, money, and sweat equity of Parts Express and the vendors to make this event possible is outstanding and a tribute to the industry. So huge thanks to the participants and vendors.

My overall take on all the speakers is that some of them have styles and finishes that would give some of the finest commercial speakers a run for their money. Some of these speakers get an unbelievable amount of performance for the size and money. As I have stated in past years what continues to be a limitation in most of the entries is the lack of adequate linear volume displacement to handle high levels of bass with low distortion and good transient response. Although box design and DSP can help in maximizing the amount of bass available from smaller drivers, at some point you run out of sufficient volume displacement. Once you exceed Xmax things go downhill very quickly. That said, some of the entries, if coupled with a good subwoofer, would be no-holds-barred systems. I could live with many of them forever just the way they are. Kudos to all for the outstanding work.

For more information visit: Midwest Audiofest ([www.midwestaudiofest.com](http://www.midwestaudiofest.com)) or Parts Express ([www.parts-express.com](http://www.parts-express.com)). 

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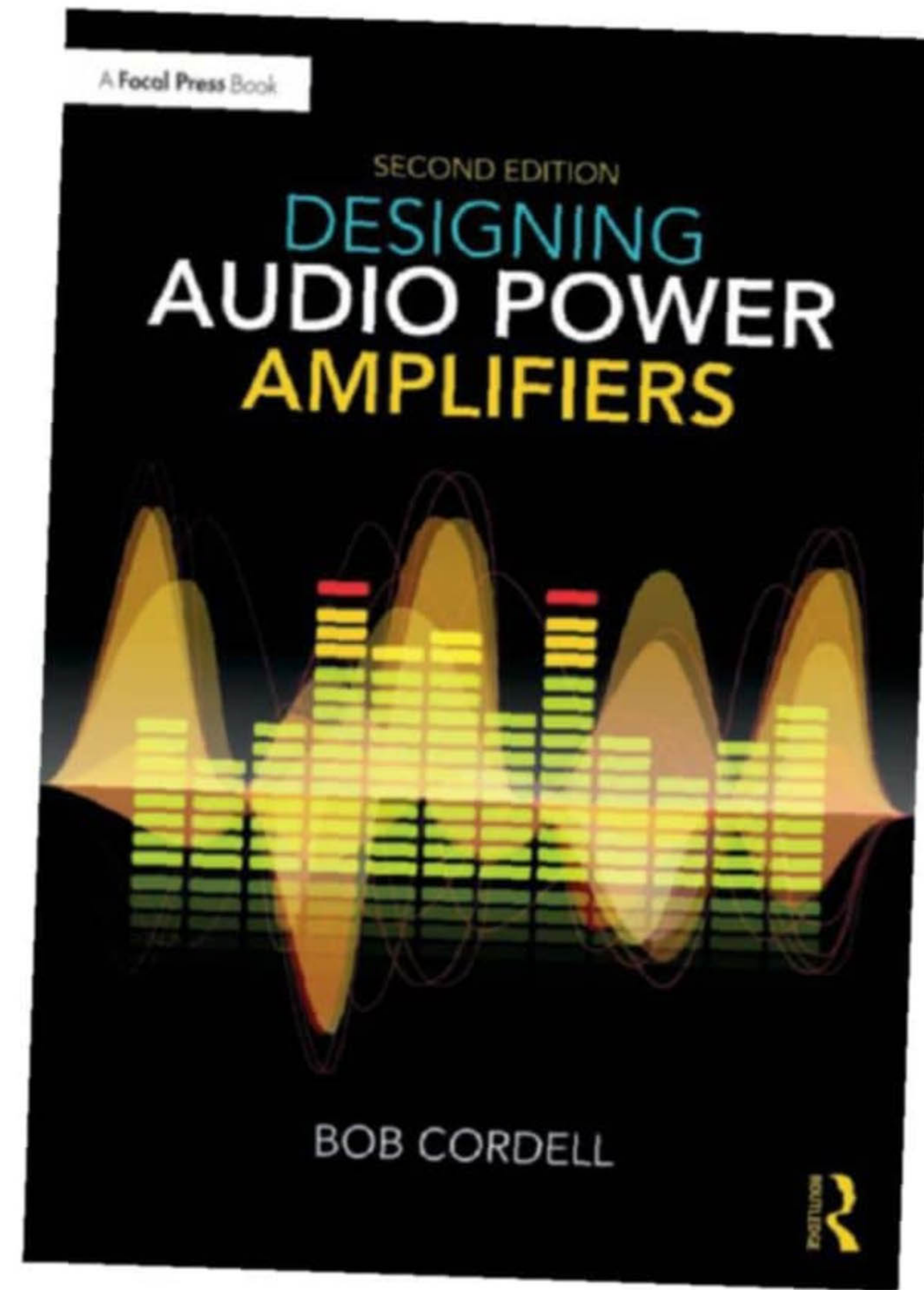


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# A Look Inside Bob Cordell's *Designing Audio Power Amplifiers*, 2nd Edition

By  
**Jan Didden**  
(Technical Editor, *audioXpress*)



When Bob Cordell published his long-awaited first edition of *Audio Power Amplifier Design*, it was hailed as a significant addition to the knowledge store on the subject. Reactions of readers, both serious hobbyists as well as design professionals, made it clear that Cordell hit his mark dead center. So the second edition has been awaited with high expectations. Would Cordell be able to trump his own accomplishments? Well, let's find out. The book is divided in six main parts. There is a clear distinction in the goal of each part.

## Part 1

"Audio Power Amplifier Basics" discusses power amplifier subjects and presents all the information needed to design a basic audio power amplifier. Topics for each chapter range from transistor types and circuits, amplifier stages, current sources, voltage references, and operational amplifiers (op-amps) to design analysis, which is a sort of pulling together of all the described factors for a complete basic amplifier design.

New in Part 1 in the second edition is a full chapter to do just that: "Building an Amplifier," which goes through all stages of a basic design, building it, testing, troubleshooting, and upgrades. This is something Cordell does in several places in the book: explaining circuits and design considerations, then pulling it all together showing how the parts fit to design a complete amplifier. This is a very effective strategy and will help the reader quickly get to the point where he/she can apply what has been discussed. Also new in Part

1 is a chapter on noise, both from a theory as well as a practical point of view.

## Part 2

"Advanced Power Amplifier Design Techniques" goes one level up, discussing advanced power amplifier design techniques. Here we find topics such as input and voltage amplifier (Vas) stages, DC servos, advanced compensation, output stages, and cross-over distortion and other sources of distortion. DC servos, you might say, what's there to say about such a mundane well-known circuit? Well, as it turns out, a lot! For all intents and purposes, the servo is in the signal path so it merits at least the same attention as the other signal-carrying parts of the amp. Servos can also influence amplifier stability and play a role in the protection system. A whole area, often only mentioned in passing, but very important if you aim for the best.

Chapter 11, "Advanced Forms of Feedback Compensation," is another chapter that you must absorb and understand if you want to design well-thought-out high-performance audio amplifiers. Feedback compensation is the critical factor for both high performance and stability. Designing a feedback amplifier presents the eternal compromise between high loop gain, allowing vanishing low distortion, high damping factor, and the other things we like, against increased phase shift around the loop that might endanger the amplifier stability. Learning about advanced compensation concepts in this chapter will help you find that optimum compromise in your own design. Here again, Cordell

shows himself a master of explaining advanced concepts in simple, easy-to-understand terms, which only comes from a complete and full understanding of the subject yourself.

Part 2 also dedicates a full chapter (Chapter 15) to an extensive discussion of error correction concepts. The chapter concludes with a complete detailed, proven design utilizing error correction to get to extremely low distortion over the audio band.

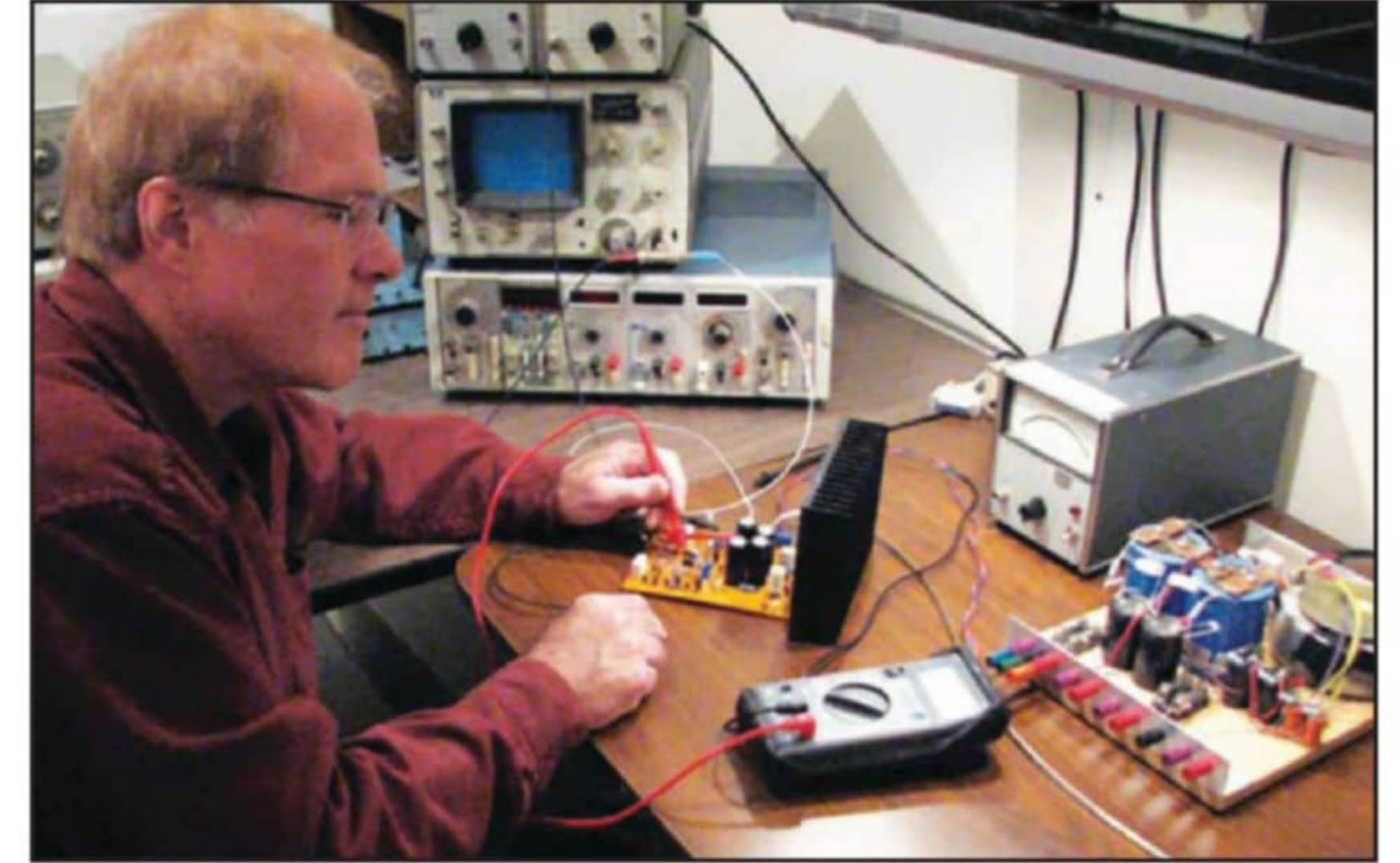
New in Part 2 is an additional chapter on output stages (Output Stages II). Here, Cordell addresses selected topics such as Complementary Feedback pairs, analyzes the interesting Bryston output stage, and discusses his own new development christened the DoubleCross output stage. This appears to be inspired by Douglas Self's Crossover Displacement concept (which is also discussed)—where the output stage is loaded by a DC current, displacing the crossover region away from the zero output voltage point.

The reasoning behind it is crossover distortion is less audibly damaging when the signal level is appreciable. Cordell's DoubleCross concept [I would not have chosen the term DoubleCross myself but am only too relieved he didn't choose DoubleDutch] extends this to a symmetrical displacement of the crossover region—it is like having two output stages, each displaced to either the positive or the negative direction, and their outputs joined. It is a very clever development but much more involved than can be discussed here.

## Part 3

In "Real World Design Considerations," Cordell starts off with a chapter (Chapter 17) dedicated to thermal design and thermal stability of the output stage. He clearly has done much investigation into so-called ThermalTrak devices—output power devices that contain an extra diode in the case, which can be used as a temperature sensor. The incorporation of such a diode in the thermal stability loop of the output stage is a major application, but since the diode provides an accurate information about the transistor junction temperature, they could also be used in a protection scheme.

Cordell provides detailed information and design guidelines on the use of these devices and shows that they can greatly increase the accuracy and reaction speed of thermal stabilization loops. My own experiences with similar devices [I have studied thermal track darlington's from Sanyo, while Cordell used single BJT devices from Motorola] mostly echo Cordell's, but I found that the thermal response speed is not fast enough to track low-frequency dissipation variations over a signal cycle or burst (see Resources).

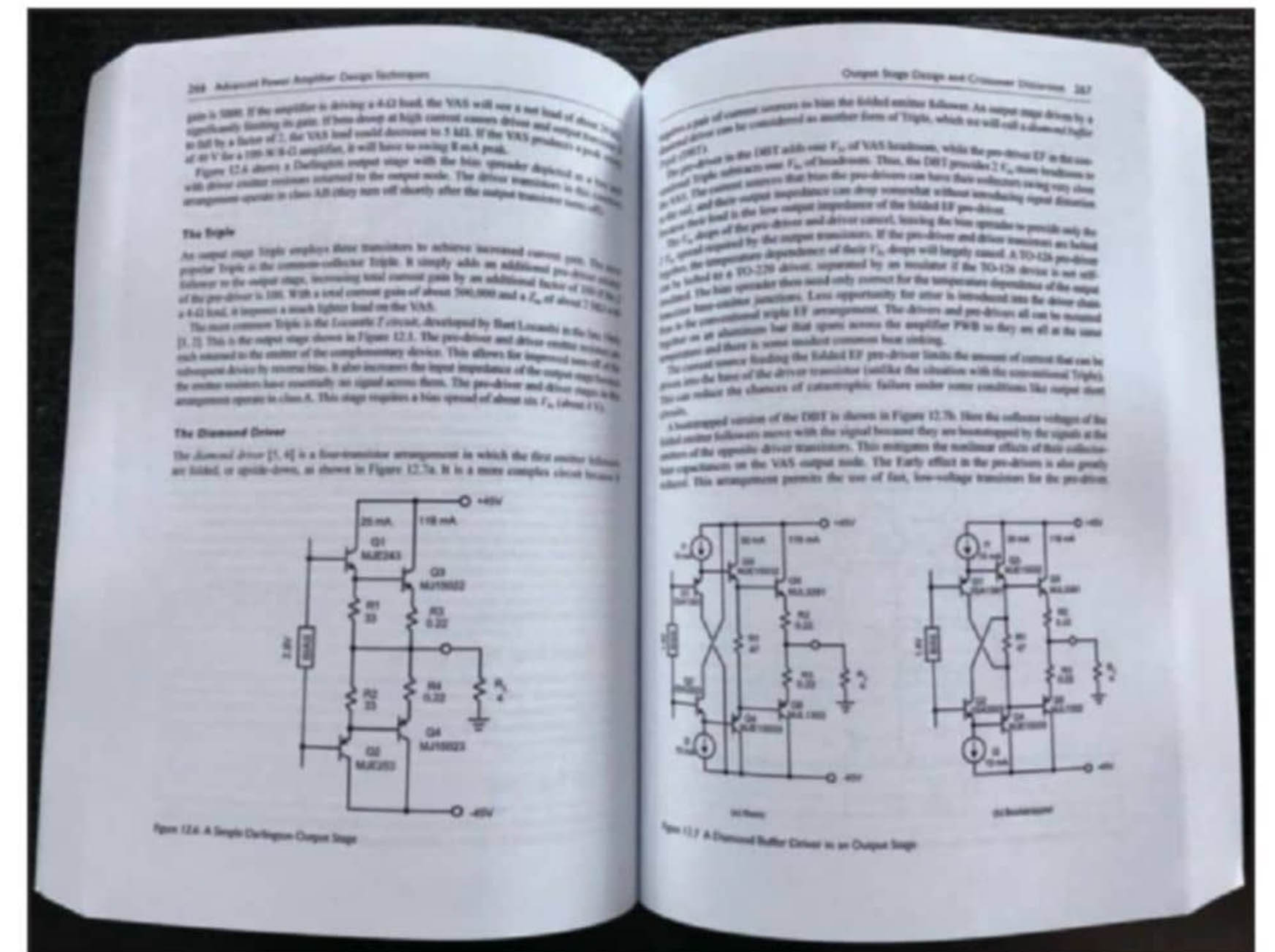


At the end of the chapter, a fully developed design is presented for a power amp taking advantage of these ThermalTrak devices.

Next is a chapter on "Safe Area and Short-Circuit Protection," going into much detail on how to design for reliability and disaster-survival, down to mitigating the possible negative effects of having a protection relay in the speaker line. (Hint: Use a very smart feedback arrangement to enclose the relay in the feedback loop yet insure that the loop is not broken when the relay opens.)

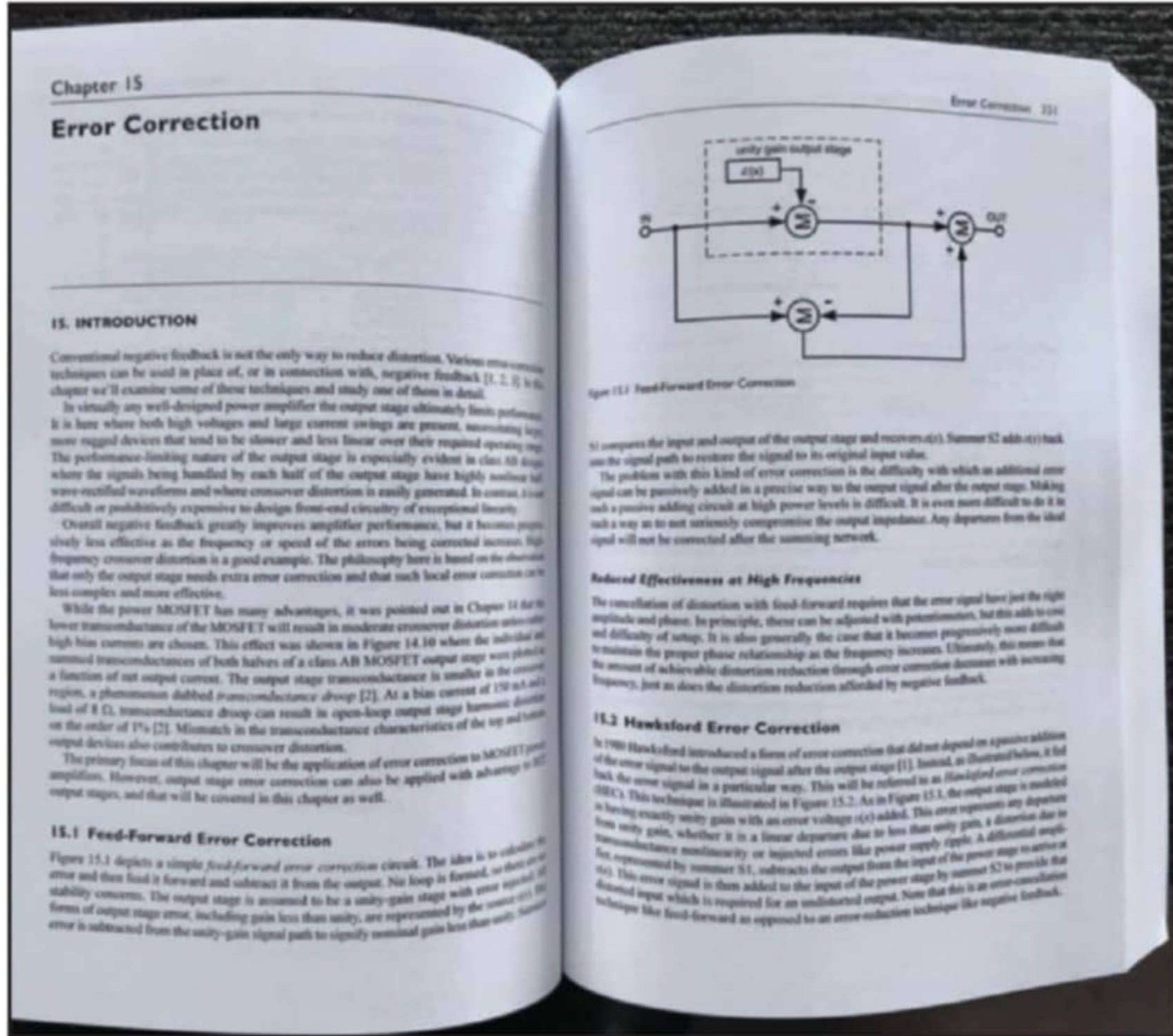
Power supplies and grounding are often neglected aspects of amplifier design. Great effort is expended

Bob Cordell is pictured in his shop, working on one of his many projects.



Schematics clearly illustrate Bob Cordell's designs as shown in this chapter on "Output Stage Design and Crossover Distortion."





Chapter 15 details “Error Correction,” beginning with the sentence: “Conventional negative feedback is not the only way to reduce distortion.”

to design the “World’s Best Amplifier,” only to slap on the usual transformer, rectifiers, and reservoir caps; connect all the grounds with heavy wiring, preferably in a star configuration; and assume all is well. Chapter 19 on “Power Supplies and Grounding” makes it clear that it’s more subtle than that. All aspects of transformer selection and sizing, rectifier diode selection aspects, capacitor sizing, bypassing, pre-stage voltage regulation, soft-start, grounding architectures, safety circuits, EMI aspects of power supply unit (PSU) design and safety grounds are dealt with. The chapter ends with a rather short paragraph on switched-mode power supplies, but—I should have known better.

New in Part 3 is a full chapter on “Switching Power Supplies” (Chapter 20). Almost 50 pages of everything from switched-mode power supply (SMPS) types; operational modes (e.g., continuous or discontinuous modes); regulated and open loop converters; half, full, and synchronous converters; EMI filtering and suppression; and power factor correction. After treating all these aspects with circuit examples and calculations, a separate section is dedicated to the actual use of these types of supplies in power amps and the intricacies of peak and average current demand and regulation. The only thing I missed in this

chapter was a discussion of synchronous rectification, which can help to further improve the power supply efficiency, especially in Class A amplifiers.

## Part 4

Part 4 is about “Simulation and Measurement,” a logical combination since simulation is just measurement with other means. It starts off with two chapters on simulation, focusing on the freely available LTspice simulator originally from Linear Technology, now Analog Devices. Not just about how to make the best of the simulator itself, but also about device models, how to verify them, how to tweak them, and how to test them. This is of utmost importance because without realistic part models, a simulator is worse than useless. Cordell’s website has many device models, which he himself has developed and tested.

The sections on audio measurement are simply complete, discussing any test anyone would ever want to do on audio power amplifiers, and presents available stand-alone and PC-based test equipment. Over the years, Cordell has developed several purpose-built pieces of audio test equipment and the designs of these are available on his website [www.cordellaudio.com](http://www.cordellaudio.com).

## Part 5

“Topics in Amplifier Design” starts off with a sensitive discussion titled “The Negative Feedback Controversy.” Here, Cordell does a very good job of dispassionately discussing the pros and cons of (global and local) negative feedback. He counters the usual arguments, that feedback “comes too late to correct what has already happened,” the “distortion goes around so grows every time” (which he calls “spectral growth”), and the other usual negative attributes with facts, figures, and reasoning. Not taking a stand, he just presents the facts and lets the reader draw the (often inescapable) conclusions.

One chapter that personally interested me is about “Amplifiers Without Negative Feedback.” Cordell is well aware of the sometimes heated controversy about this subject—some swearing by it, others eschewing it as it “kills the music.” Sensibly, he does not choose sides (although those who know him, know where his preferences sit), but clearly and succinctly explains the design trade-offs and challenges in attempting to design an amplifier with no feedback. Emitter degeneration and compensation around the Vas can also be considered (local) feedback and Cordell doesn’t use these types here either.

If you have never tried such an exercise, this is a real eye-opener! Somewhat oddly, after presenting a complete no-feedback amplifier, the chapter ends with

another complete design that does use feedback, but concentrates on a design sporting very large open-loop bandwidth, 40 kHz in this case.

New in Part 5 is a chapter on “Professional Power Amplifiers” as can be found in sound reinforcement systems for performances. The requirements for such amplifiers are different enough from home sound system amplifiers to warrant a separate discussion. In professional amplifiers, it is all about reliability, high power with small size, and high efficiency. They normally have balanced inputs for low noise with long cables, and often include some form of DSP capability for advanced protection or to configure them for the task at hand.

## Part 6

“Class D Amplifiers” is dedicated to the many aspects of Class D amplifiers and is a very complete overview of the technology and technical issues. Cordell starts with a discussion of basic audio Class D concepts, progressing to a chapter on design issues like filtering, sources of distortions, bus pumping and the specifics of closing a feedback loop around the output filter. Alternative Class D modulators and the requirements for measuring the performance and efficiency of Class D amplifiers round out the chapter, the part, and indeed the book.

## The Upshot

So there you have it. There is so much in this book that our review can only scratch the surface, but you can think of any subject in power amp design and be sure that somewhere within its pages Cordell explains it and tells you how to get the best out of it. Almost 800 pages is A Lot!

I should also mention each chapter concludes with an extensive list of reference articles and papers. These are invaluable, allowing one to dig deeper into the relevant topics. Many of these references are freely available on the Internet and on Cordell’s own website.

What remains now is to answer the question we posed at the beginning: does this second edition meet or exceed the very high standard set by the first edition back in 2011? I’d say it does. Cordell has clearly listened to the comments and suggestions of his readers. Existing sections are expanded, concepts clarified, new material added, to the point where it has become a one-stop shop for the serious audio power amp designer.

The book is not cheap, edging up to the \$100 mark, but really, with a book like this, the price is only of limited relevance. You just can’t get around it if you are serious about audio power amplifier design. You need it. Period.

## About the Author

Jan Didden has written for *audioXpress* since the 1970s. He is retired following a career with the Netherlands Air Force and NATO. He worked in logistics, air defense, and information technology. Retirement has provided him with the time to finish all the audio projects that have piled up for decades. He writes about them on his website [linearaudio.nl](http://linearaudio.nl). Jan was also the publisher and managing editor of the twice-yearly bookzine *Linear Audio*, and currently *audioXpress*’ technical editor.





# Integrated, Networked, and Secure

In this market update *audioXpress* provides an overview of product and technology trends in the audio installation market. Given the size of the market and number of product categories, in this update we are focusing on the most cutting-edge technologies for commercial and corporate installations.

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

Rixos Premium Hotel installation in Dubai  
(Photo courtesy of Powersoft)

Audio technologies and solutions for installed sound applications are not only one of the highest growth segments in the audio industry, they're also one of the more innovative and vibrant. Although even audio engineers tend to associate installation audio systems as a sort of grey area and not exactly the most exciting, the truth is, the installation market is currently at the forefront of exciting innovations in terms of networking, signal processing, and control, benefiting from the convergence with cutting-edge professional audio and consumer technologies and even leading in some areas.

As market research firm Futuresource Consulting recently characterized in its post-InfoComm show report, the installed leisure and installed commercial segments continue to flourish for professional audio, with new high-growth application areas (e.g., conference and collaboration rooms in corporate installations, as well as commercial sound, particularly for hospitality and retail). Sophisticated beam-forming meeting room microphones, growing at double digit rates in 2018, and pendant speakers are just some

of the most visible examples in installed sound, remarks Futuresource. This, while stadium, and event auditorium applications continue to generate high volumes, as the latest immersive sound trend is also introducing a new level of excitement.

## Notable Trends

As Futuresource reports, with double-digit year-on-year growth for professional loudspeakers in the installed commercial sector, microphones taking almost half of all value of installed applications, and DSP continuing to become a key selling point for audio systems, this market segment has certainly seen significant evolution in the last decade. Companies offering fully integrated systems are experiencing significant growth, especially in corporate applications, as the market also signals continued growth in key applications (e.g., houses of worship, conference centers, and hospitality venues). Also, many entertainment spaces are converting to permanent installations (evolving from the mandatory basic public address and evacuation systems) to

accommodate a growing range of events, including large corporate presentations and e-sports.

The 2019 *AV Industry Outlook and Trends Analysis (IOTA) Global Summary* produced by the Audiovisual and Integrated Experience Association (AVIXA), confirms these trends, revealing an upward revision that the "global professional audiovisual industry will grow to more than \$325 billion in 2024, from \$247 billion in 2019." "The growth in pro-AV revenue is actually outpacing expectations for global GDP growth," states Sean Wargo, Senior Director of Market Intelligence, AVIXA.

Even though the AVIXA/IHS Market IOTA report combines many different product segments in the overall audio-visual industry, the outlook reflects the trends specific to the audio product segments, signaling in particular the largest pro-AV revenue-producing region to be the Asia-Pacific (APAC) region overtaking the Americas in 2019. With the Americas projected to continue its growth, although at a slower pace than APAC, the corporate segment will be a major driver for the region, with the same trend confirmed for Europe, the Middle East, and Africa (EMEA), albeit with more moderate growth.

## Acquisitions and Mergers

With this much growth and promising expansion forecasts, there's no wonder that the installation segment has also been the focus of a strong merger and acquisition activity. A few recent announcements provide clear examples of the trend in the market.

Biamp Systems, a manufacturer of video and audio networked solutions for commercial applications based in Beaverton, OR, announced the acquisition of Audioprof Group International, the parent company of Community Loudspeakers based in Philadelphia, PA, and Apart Audio based in Antwerp, Belgium. Founded in 1968 by speaker design pioneer Bruce Howze, Community Loudspeakers is a North-American manufacturer, known for its broad range of premium speakers targeting demanding indoor and outdoor applications including commercial, leisure, stadiums, and other large venues.

Apart Audio, a Belgium company founded in 1992, is a strong brand in commercial and leisure audio solutions for small and mid-size venues including retail, food service, education, and small offices. The company offers a broad array of speakers as well as amplifiers, mixers, controllers, and paging stations, which enabled it to establish a strong position in the installation market in Europe. In 2015, with the support of 3d Investors, a family owned investment company, Apart Audio expanded internationally with the successful acquisition of Community in the United States. The acquisition lead to the creation



From consumer and residential to commercial installation. Klipsch Audio, a VOXX International Co., is one of the famous North American brands that made the profitable leap.

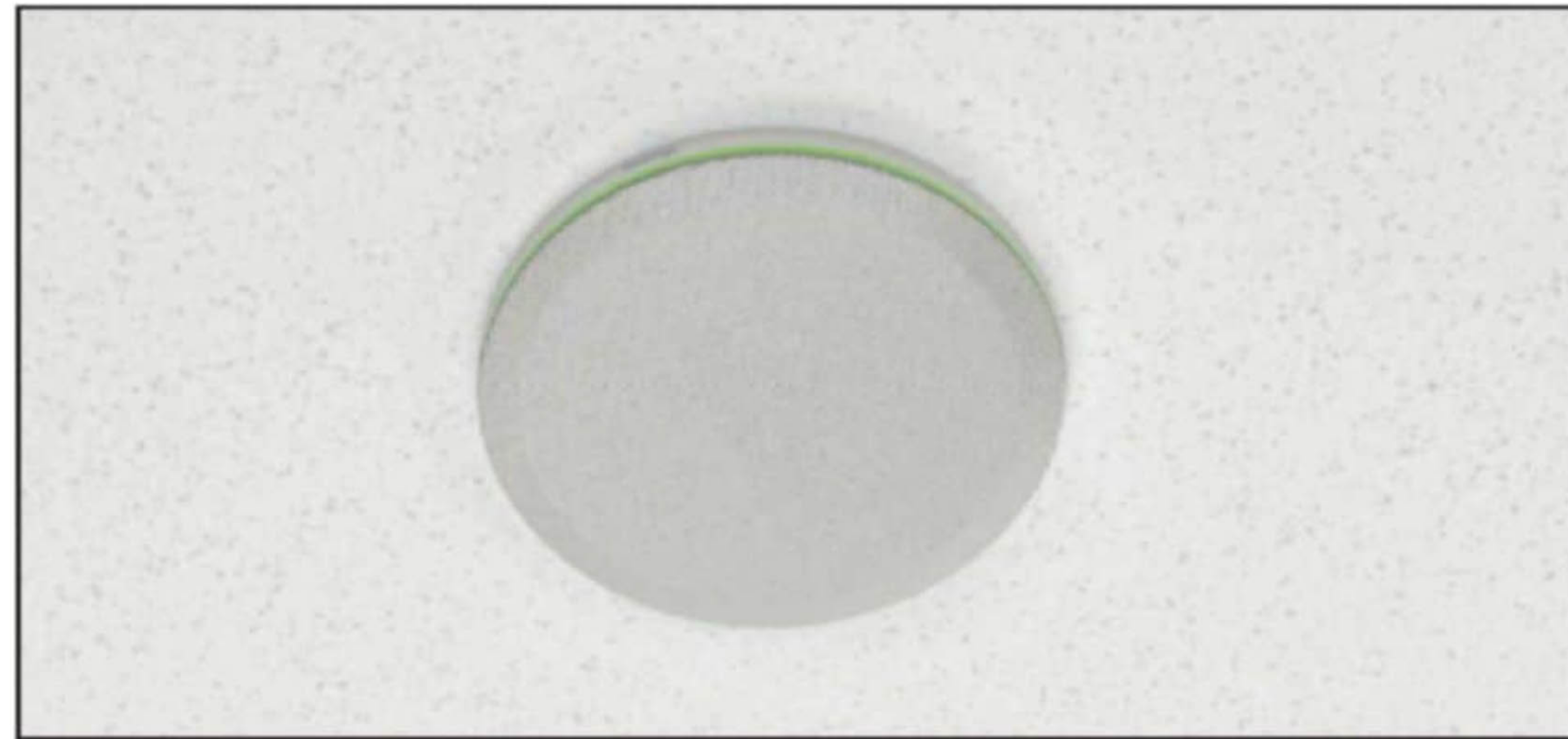
of Audioprof Group International, now acquired by Biamp. Founded in 1976, Biamp's recent history is also one of sweeping changes, particularly since in 2017 the company was acquired by Highlander Partners L.P., a private investment firm based in Dallas, TX.

In September of 2016, Highlander Partners added Rashid Skaf, the former CEO of AMX (acquired by Harman), as a Senior Advisor to identify technology company opportunities. Following Biamp's acquisition, Skaf became President, CEO, and co-Chairman of Biamp. After taking the helm at Biamp, Skaf declared



The latest L Series LVH-900 Beamforming Venue Horn from Community Professional, now a Biamp brand. The L Series LVH-900 loudspeaker system addresses the unique coverage and projection challenges of large venues and stadiums.





Biamp's latest Parlé family of beamtracking microphones includes ceiling and tabletop models, providing crystal-clear voice pick up by tracking and following meeting participants as they move around the room.

the company would rapidly grow within its existing markets and enter entirely new segments using a combination of product line expansion and company acquisitions. Since then, Biamp has introduced several innovative new products including beam tracking microphones, and in December 2018, Skaf expanded Biamp's scope with the acquisition of Cambridge Sound Management (CSM), headquartered in Waltham, MA, and a pioneer in the fast-growing sound masking market, with an extensive range of solutions and integration know-how that perfectly complements Biamp's activities. Already in 2019, Biamp appointed Mark Ureda, a former senior vice president at Harman Professional, as executive advisor to lead the company expansion and the integration of its new brands.

Another important story in this area was the acquisition of Polycom, confirmed in July 2018, by

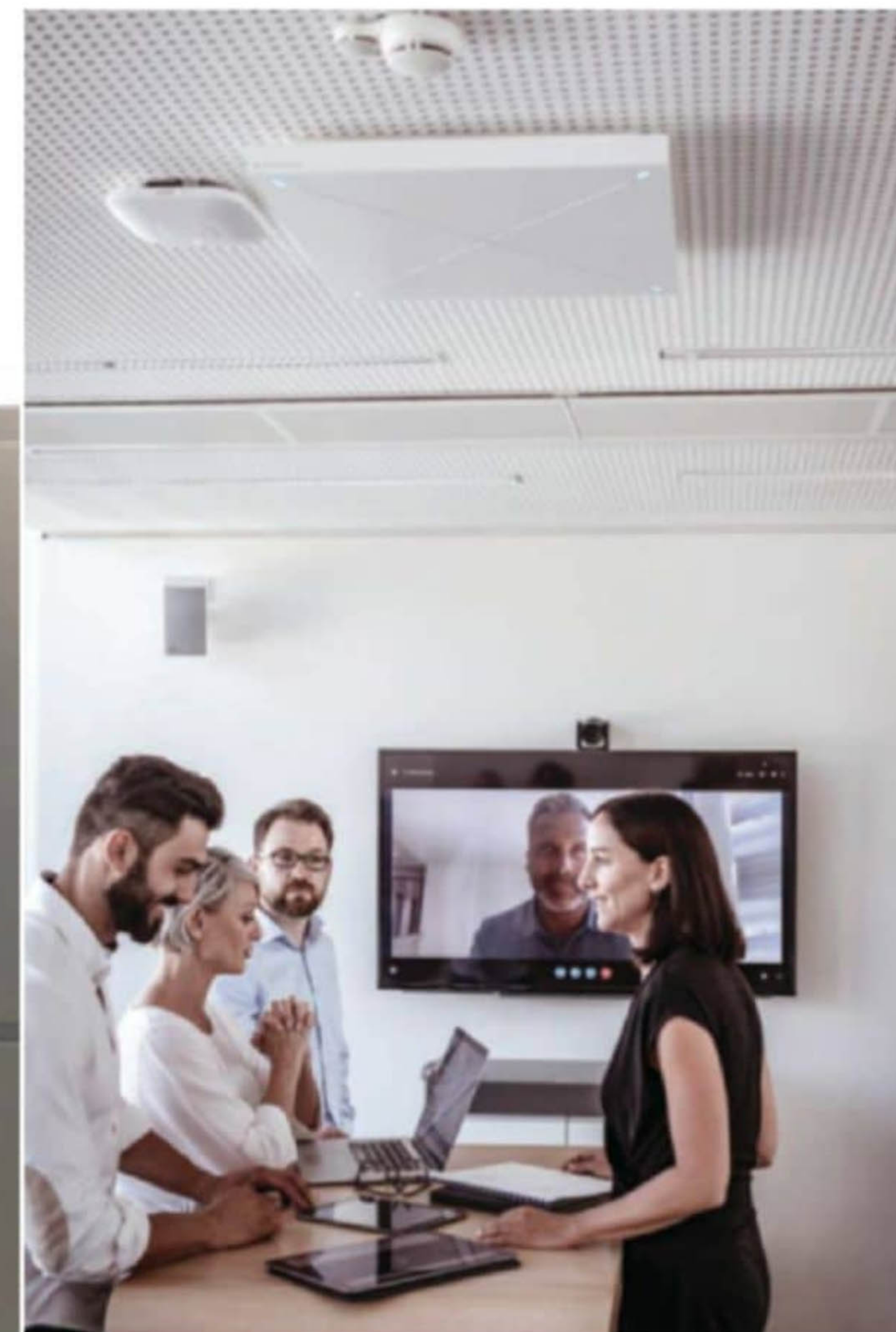
Sennheiser is now shipping its latest TeamConnect Ceiling 2 (TCC2) microphone array, which uses a patented automatic, dynamic beamforming technology. Unlike ceiling microphones with static beamforming technology that work with predefined speaking zones, TCC2 reliably captures any number of speakers no matter where they are in the room.



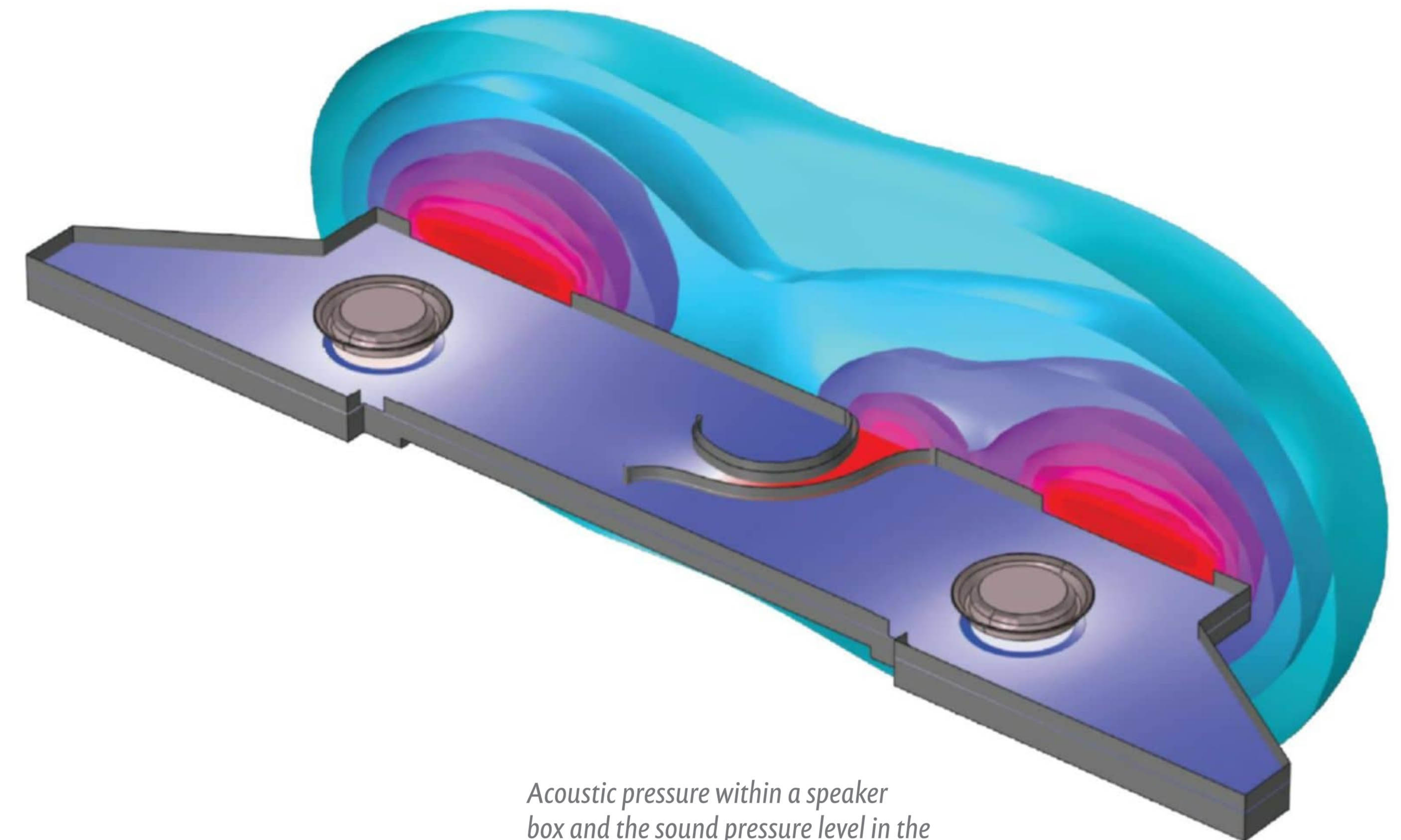
Plantronics. The two companies have already merged and have created Poly, a technology company "focused on the human experience of communications and collaboration technology and systems." Poly is now focusing on four distinct areas of innovation: starting with modern offices, an area pioneered by Plantronics in its Habitat Soundscaping solution, followed by collaboration solutions, mobile solutions for business, and cloud services.

The third and more recent example was when SnapAV, a manufacturer and supplier of A/V, surveillance, networking, and remote management products for professional integrators, announced the acquisition of Control4 Corp. Control4, which acquired Triad Speakers in 2017, is essentially focused on the residential and smart home segment, but its multi-room audio, intercom, lighting, climate control, security systems, and technologies are increasingly used in building applications.

In another more AV-focused acquisition in the past six months, Panduit Corp., a manufacturer of electrical and network infrastructure solutions, acquired AV manufacturer Atlona. Panduit is headquartered in Tinley Park, IL, and operates in 112 global locations focusing on building and corporate IP solutions, including its All-IP Meeting Space system, which simplifies meeting room designs by bringing the power of large, enterprise-sized, IP-based systems to individual meeting spaces without the need for



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complex switching systems. Atlona designs and develops products for a diverse range of commercial and residential AV and IT markets worldwide, including education, business, government, entertainment, and healthcare. Atlona will continue to operate from its Silicon Valley, CA-based global headquarters and from



Bose Professional has significantly expanded its portfolio of loudspeakers, amplifiers, DSPs, controllers, and software products for the installation market. With more than 50 new products announced at InfoComm 2019—including its DesignMax and ArenaMatch series—designers and installers can now benefit from a seamless system with components that have been designed to work together and simplify onsite use.



Since 2017, Kansas-based MSE Audio, the parent company of SoundTube Entertainment and other brands expanded significantly its range of IP-addressable, Dante-enabled and Power over Ethernet installation solutions. Pictured are the SoundTube pendant speakers, a Dante-enabled line of installed speakers using high-frequency dispersion technology for use in rooms with extra high ceilings.

its international headquarters in Zurich. And these are just some examples in the manufacturing and product side. The installation market as also seen a lot of merger and acquisition activity among system integration and installation companies, with large corporations expanding nationwide or internationally.

## Integrated Installation

Strong application segments in the installation market are currently conference rooms, huddle rooms, unified communications and collaboration, retail, digital signage, education, and entertainment. Even though we tend to think about entertainment applications as rental and touring activities for the audio industry, increasingly, installation audio is more and more about the number of fixed installations for that purpose, including installations in fashion retail outlets and even restaurants, not to mention the always vibrant bar and clubbing segment—which remains a big part of the installation market. And of course, installation has significantly expanded in the last decades as the number of projects for airports, transportation terminals, and large public spaces became increasingly more demanding with the convergence of safety and security systems, while building automation expanded to integrate all systems, with a shared infrastructure where audio (public address) is a vital part.

And finally, as the corporate market also evolved to much more sophisticated concepts, of which biophilia, soundscaping, noise masking are just a few of the examples (for those who haven't read it, I strongly recommend the "Curing the Open Office Noise Epidemic... With Nature?" article. See Resources), there is a renewed trend towards a new style of working spaces, promoting audio and video conference, remote collaboration, and the new concept of huddle rooms, a vital part of the open office concept, where workers sometimes need to retreat to make calls, and groups can quickly assemble in small conference areas, benefiting of the latest audio, video, and display technology to connect their own devices.

## Connectivity and IP

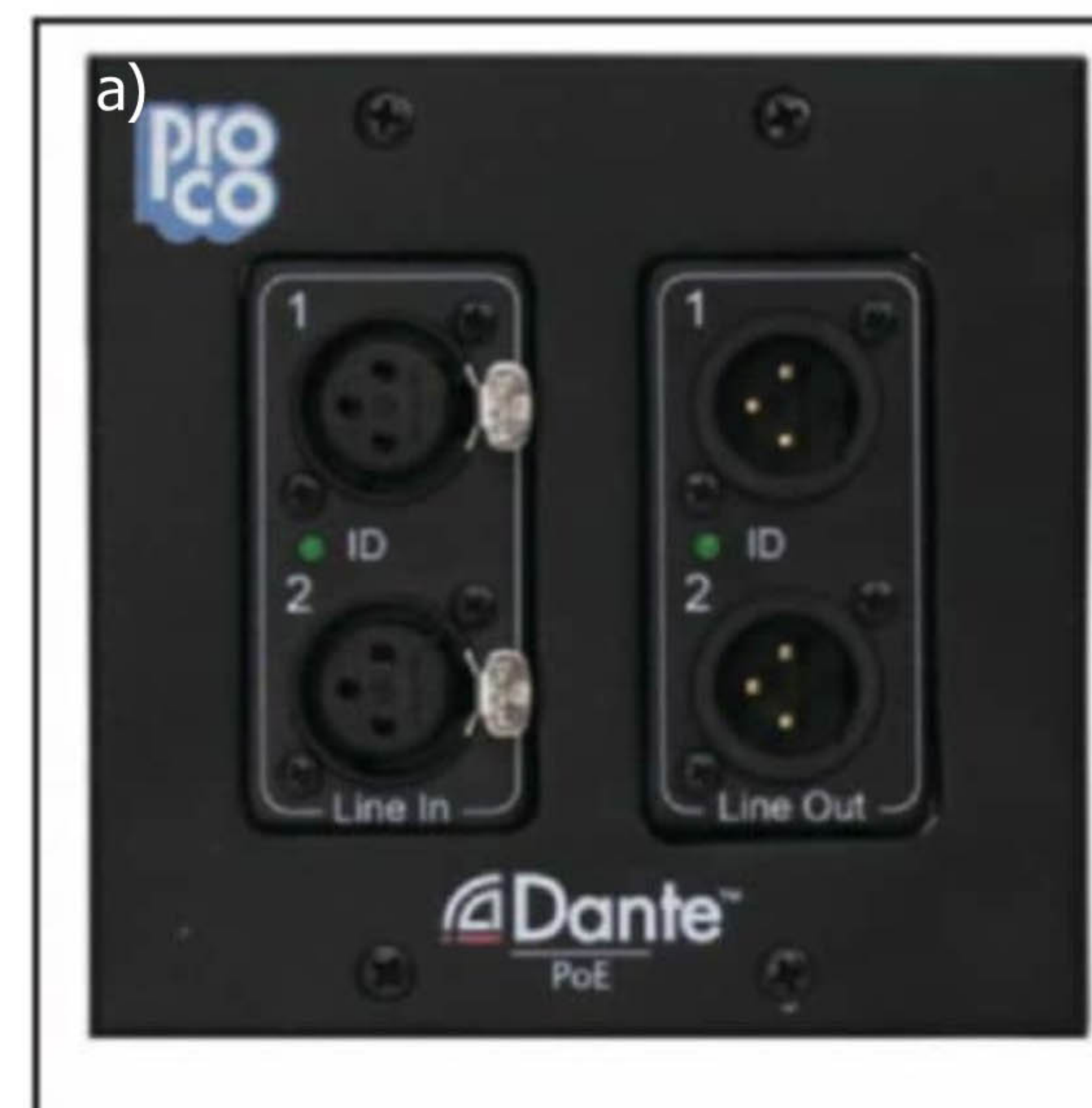
The world of installed systems is dominated by AV technologies, and any visitor to a trade show in the segment is going to feel overwhelmed by the unavoidable references to HDBaseT, HDMI, DisplayPort, KVM extenders, 4K over fiber, video wall controllers, and the like. When we start looking for an old familiar XLR audio connector, it is probably going to be an interface, designed to connect "legacy" systems. Even good old microphones have evolved to IP-connected systems—never mind the excellent quality of those analog microphone preamps still

available everywhere. In the installation world, the need for networked control and the convenience of self-calibrating and auto-adjusting AV systems is now a requirement. Another trend in the market is that of the giant vendors with huge product catalogs that can do it all (e.g., Crestron or Kramer, or the large brand conglomerates that deliver a completely integrated solution). Typically, those companies will start their pitch by selling their own network AV solution that handles audio, video, image processing, browser-based control, KVM routing, all over a single Gigabit Ethernet cable, with fiber LAN options for large campuses.

When we approach multiple small vendors instead, the technology message tends to get complicated, with many technologies referenced to complete a networked solution, making a contract with a systems integrator company almost always a necessity. On the audio side of things, references to Dante/AES67 audio networking are now almost a given, even if there are other networking protocols and technologies in other application segments (e.g., studio, live sound, or broadcast).

When a company, looking to create a new area of meeting rooms or conferencing facilities, looks at the description of the system, it will find the design typically built around a Dante/AES67-networked infrastructure. Audinate can rightfully claim that its Dante technology is now the de facto industry standard on commercial installation. With more than 2,000 manufacturers licensing Dante from Audinate, the availability of simple one-channel interfaces and distributed interfaces from two or four Dante inputs, to solutions routing hundreds of channels, the market offers everything needed for any type of system.

One of the reasons why Dante networking has gained this industry predominance is Audinate's continuous investment in the technology as an evolving standard. Its early adoption was propelled by the availability of practical and affordable OEM development solutions, both at chip and board level,



ProCo Sound, an ACT/RHC brand, introduced a range of Dante AoD I/O Modules in multiple form factors, including wall plates (a), making Dante audio connectivity completely seamless. Power is supplied through a standard Ethernet cable (b) via PoE capable network switch, or from a separate PoE injector circuit (c).



An ever-evolving networked ecosystem, Audinate's Dante technology bridges with any audio-over-IP solution by supporting the AES67 interchange standard.



Dante now supports popular ARM embedded platforms and is available as a software-based implementation for PC and Mac applications allowing for even broader adoption by the audio industry in multiple application segments.



Attero Tech is another success story of a company that pioneered practical networked installation solutions and currently offers an expanded range of Dante/AES67 products, including the Synapse D32Mi high channel networked audio interface for those installations requiring high numbers of analog to Dante conversion.



allowing manufacturers to quickly bring new solutions to market.

Now, Audinate is consolidating its position with the integration of video signal distribution into the standard, with its Dante AV solution. Audinate also recently announced a shift from purpose-built hardware to software-based solutions running on standardized hardware, with its latest Dante Embedded Platform, which enables manufacturers



Even Neutrik is expanding into audio networking. The new Neutrik NA2-IO-DLINE and NA2-IO-DPRO Dante interfaces are designed in direct response to current market needs, with solutions optimized for use in extreme environments and applications.



Genelec is significantly increasing its investment in the research and developments of solutions for the fixed install market. In 2019, the Finish company launched its Smart IP audio platform at ISE 2019, a unique open IP networking technology aimed specifically at the installation market, which delivers power, audio and scalable loudspeaker system configuration, supervision and calibration features via a single standard CAT5 cable.

to add full Dante functionality in software running on Linux for x86 and ARM processors, while on the other hand promoting the new Dante Application Library, which allows software developers to seamlessly integrate Dante directly into their PC & Mac applications. Complete reference designs for popular ARM SoC chipsets such as the Analog Devices ADSP-SC589 and NXP i.MX 8M Mini QuadLite are now available, and early confirmed adopters of the Dante Application Library include QSC, Zoom and Analog Devices. As part of its strategic technology partnership with Audinate, QSC announced that its Q-SYS Ecosystem—which runs over standard IT network infrastructures—will now feature scalable software licenses that will enable native Dante-networked audio integration without the need for additional hardware I/O or external configuration software.

## IP-Addressable Everything

On the network “peripherals” side, there will be hanging microphones, ceiling arrays, and table-top boundary or flush-mount steerable microphones that can be positioned in the rooms, eliminating the need for complex, distance-limited, and fault-prone analog cables, with digital audio signals traveling over Ethernet networks, offering the possibility of centralize control, monitoring, and logging of events in a control room with just software and a few screens (even an iPad can do it.)

Digital wireless microphones optimized for speech and lectures from Sennheiser, Shure, beyerdynamic, and Audio-Technica are another example of how the growth in the installation market inspired companies to create dedicated series of what is essentially proven-and-tested technology used in studio, live, and broadcast environments. Installation-focused microphones are now connected to a series of dedicated software control tools that focuses on remote or automatic management, multi-room, or campus-wide setup, replacing the need for technical support to little more than regular battery replacement. And, of course, those systems are available with Dante, and come in white, grey, or metallic color schemes to differentiate the “installation series.”

Even companies looking to equip a simple small meeting room and not wanting to compromise on professional specifications will find it easier to install USB-powered microphones (actually a sophisticated array of multiple microphones in a table-top unit). Analog XLR? A thing of the (remote) past. In larger multi-use rooms, the audio captured by microphones is optimized to the room acoustics, can cover multiple different zones, or even “track” the speaker if it is moving, with the signal processed with noise, eco, and reverberation cancelling, plus automatic gates,

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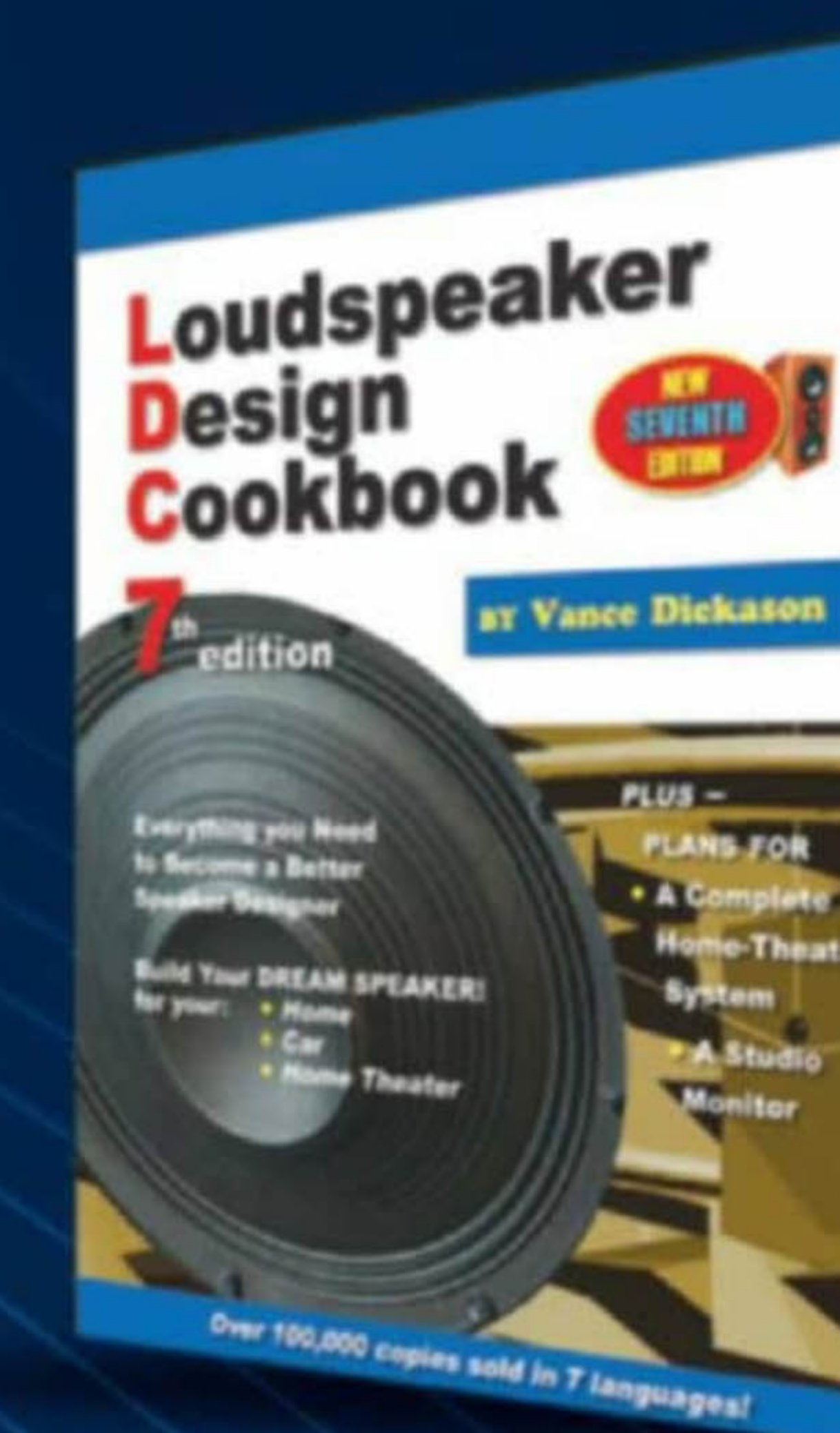
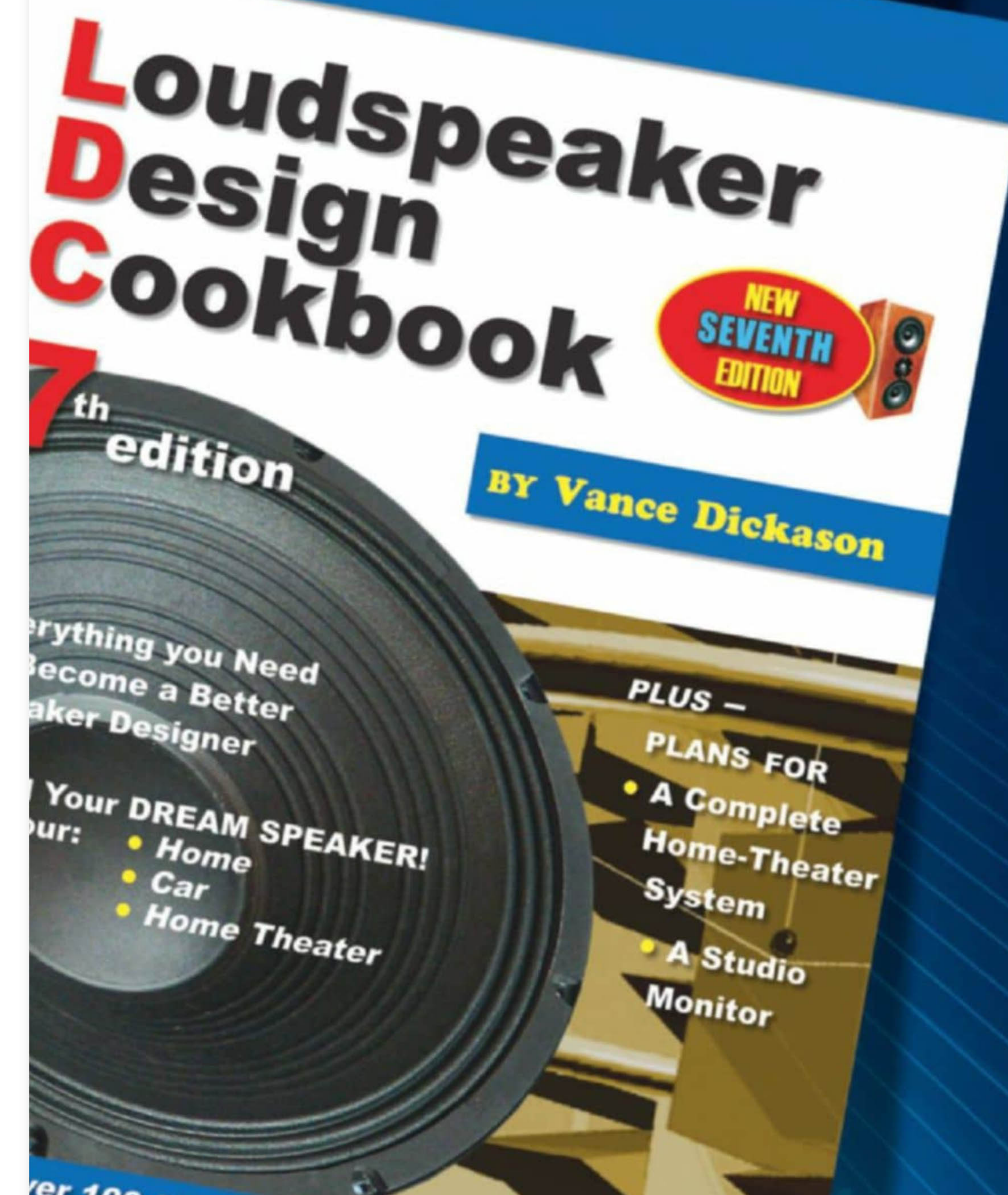
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An early pioneer in digitally steerable vertical array loudspeakers with its installation-oriented Iconyx series, California-based Renkus-Heinz has now significantly expanded its range with all-new Directivity Control Series, Iconyx Compact Series and ICLive X Series speakers, all of which can be wired networked or Wi-Fi connected.



Responding to the needs of versatile solutions for integrators, full-range compact loudspeakers are now in high-demand. Pictured is the latest Danley Sound Labs Go2-6CX compact coaxial design

levelers, and limiters, which make the need to have technical personal on hand, full time, unnecessary.

### Installation Speaker Trends

On the speaker side of things, options are also diverse, with the installation segment paving the way to a lot of new technologies and creating a market for different designs. Basically, system integrators and their customers have a wide choice of solutions, from simple line-level distributed speakers, all the way to high-power, high-quality PA solutions from the major brands like d&b audiotechnik, Yamaha, JBL, or Bose, which are able to support background music one day and accommodate the requirements of a live performance with a real band in another. When the needs are more critical and the acoustic environment is more challenging, the industry offers everything from extremely specialized beam forming speakers in vertical columns or in-ceiling formats, to large-sized, distributed array structures for large venues.

Yes, there are still those manufacturers who continue to offer traditional two-way 12" PA boxes with a rotatable horn, additional hanging or wall-mount options, maybe a removable connection panel and all-weather options, and call it an "installation series." But a walk through the aisles of the Integrated Systems Europe (ISE) show (much more than InfoComm...) reveals a diversity of "installation-focused" new speaker designs, with all sorts of colorful options and, more importantly, completely new electroacoustic designs—including many new pendant-type models—and different connectivity options, including AoIP.

Things are also diverse in terms of technologies. For situations where speakers need to remain invisible or there are critical all-weather requirements, companies such as Stealth Acoustics, pioneers in the invisible speaker approach, now offer efficient and practical designs for residential and commercial integration, with much improved flat panels that are able to generate the SPLs of a conventional PA and have impressive fidelity.

There's also Canadian company Revolution Acoustics with its SSP6 Multiducer, a unique transducer or actuator, which is able to transform nearly any panel structure into an audio speaker. Apart from enabling solutions with IP65 ingress protection for water and dust, the SSP6 Multiducer is able to convert drywall, wood, or glass windows, into a speaker that generates responses from 45 Hz to 20 kHz. In recent years, the company has been expanding its ecosystem in order to deliver specific solutions for different requirements and is now promoting a Pendant360 speaker, and the Ubiqui-T360 ceiling tile, complemented with a new dedicated series of installation amplifiers with Wi-Fi and Dante/AES67 support.

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Drawing from different industry segments, manufacturers have also been producing a wide variety of solutions for the fixed installation market, expanding on speaker arrays, from multiple beam-steering vertical columns to horizontal soundbar-type designs, and even wave field synthesis, like the very promising modules from German audio company HoLoplot, which enable a new approach in highly selective coverage and acoustically challenging environments. And,



A lighter, thinner, easier-to-mount solution that improves directivity and intelligibility, the new DML500 Flat Panel Speaker from Tectonic Audio Labs is a perfect example of the latest advances in speaker technology, using the company's next generation distributed mode loudspeaker featuring bending wave technology

there is the latest trend which is spatial audio, or immersive sound, which after taking live performance, corporate presentations, and even touring applications by storm, is now starting to also make inroads in fixed installations. All the manufacturers seem to have taken notice and are designing loudspeakers able to sustain the challenges of large distributed systems for immersive installations, from very compact full-range and frequently coaxial designs, all the way to new high-power vertical columns and installation-oriented line arrays.

Since immersive audio requires also the availability of purposely designed, object-based audio content, it will take some time until the concept expands significantly to installation applications. But the highly positive impact over standard mono or stereo sources, and the tremendous impression immersive audio causes on audiences, make it highly likely that it will continue to expand.

As demonstrated in corporate events, immersive audio installations enable a presenter to move freely with source-tracking of the microphone, enabling the audience to know exactly where that person is speaking from and creating a much more engaging experience. Highly distributed sound-objects and precise localization, together with the enormous potential of virtual (variable) acoustics, generate the lasting impression that easily convinces a facility manager to embrace the concept (and the additional investment.) The immersive audio concept is being led by the largest companies in professional audio (e.g., L-Acoustics, d&b audiotechnik, and Meyer Sound), but now even audio-processing pioneers such as Astro Spatial Audio, which typically have focused on live performance and experiential events, are reporting

increased interest in new permanent installations for theme-parks and museums.

## New-Generation Installation Amplifiers

But if the speaker product segment is evolving and diversifying for installation applications, no other product area reflects the technology trends and tremendous growth for these applications more than audio amplifiers. Companies such as QSC Audio, which have pioneered the concept of networked distributed amplification with integrated processing, are now going all-in on IT- and IP-based solutions. On the other hand, traditional amplification brands that always had dedicated amplifiers for installation, basically offering different mounting accessories and connector types, are now evolving toward the concept of networked and fully integrated digital signal processing solutions.

As an example, Powersoft recently unveiled Mezzo, its newest installation-oriented amplifier platform at InfoComm 2019. The new line of audio amplifiers consists of eight models, each offering different power and connectivity options, including Dante/AES67, perfectly suited for corporate, hospitality and educational environments. All Mezzo amplifiers are simple to set up and can be monitored remotely, making it attractive to integrators seeking reduced installation times and more efficient deployment in the field. Even the smaller 320 W models feature power sharing capabilities, which means they are able to deliver 80/160 W per channel into 2, 4, 8, or 16  $\Omega$  as well as 100, 70 or 25 V. This means one model can drive zones of varying sizes, and design changes can be accommodated "on the fly" without having to alter equipment choice.

Bose Professional also introduced new amplifier options for premium commercial applications with its PowerSpace+, P4150+ and P4300+ models, all available with built-in DSP, including SmartBass processing, routing, level control, delays, limiters, preset-based EQs, etc. All models feature an integrated webserver and a browser-based UI. These new Bose amplifiers have been complemented with new Bose ControlSpace processors optimized for conferencing rooms, all with Dante/AES67 connectivity, while the existing PowerShare amplifier range now includes patented technologies inherited from the existing PowerMatch line, allowing the amps to share power across all output channels.

And the new amplification brand in the market, LEA Professional, was the focus of attention at InfoComm 2019, by revealing the first working models of its new LEA Connect Series amplifier family, featuring internal DSP and Dante/AES67 connections. As promised, LEA is bringing the benefits of a connected world to the Professional A/V market, allowing



system designers and integrators to provide real-time system performance coverage to their existing service contacts. All the LEA amplifiers allow Internet or local area network remote control and monitoring of installed systems, creating a safe infrastructure based on the cloud (AWS).

While for many years the installation industry has been discussing IT convergence, the door now seems to be now for increasing integration with the Internet-of-Things, moving the focus also toward online and cloud-based services. An interesting transition.

Powersoft's new Mezzo line of audio amplifiers was designed for the corporate, hospitality, and educational install markets, offering different power and connectivity options—including Dante and AES67.

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# Auralization—Building Virtual Acoustic Models

By  
**Richard Honeycutt**  
(United States)

Photo 1: Wenseslaus Hollar drawing of Old St. Paul's Cathedral (Image courtesy of Sotheby's, London, England)

When most of us think of virtual reality, we conceive it in terms of computer games and similar types of entertainment. However, as early as the 1930s, acoustical virtual reality—"auralization"—was explored. Today, using computerized acoustical modeling tools, we can produce very convincing auralizations of speech and music sound sources. These auralizations can be and have been used to add experiential elements to historical research, as well as to facilitate the design of acoustical spaces and the sound systems installed in them.

The first experiments in auralization were performed by Spandöck and his research team in Munich, Germany, using 1:5 scale models. Accurate scale-model auralization is fraught with complexities, because much acoustical behavior depends upon the relationship between (1) the physical dimensions of sound sources and room

features and (2) the wavelength of the sound waves.

Today, auralization is usually performed using "fully computed" methods. These methods convolve the anechoically recorded source material with the Room Impulse Response (RIR). Convolution (more accurately, convolution) is a mathematical process similar to filtering in that both processes are analogous to multiplication. To produce an auralization, a digitized time-domain audio recording is convolved with the Binary RIR (BRIR). The result is a digital audio stream that can be heard through headphones as a binaural experience. If less accuracy can be tolerated, the reproduction can be delivered through stereo speakers with the use of crosstalk cancellation devices (see **Figure 1**).

Computerized acoustical modeling software such as EASE, CATT Acoustic, ODEON, Bose Modeler, or ULYSSES can be used to create the BRIR. None of these is perfect, because acoustical effects such as diffraction, refraction, and scattering are quite complex to model. Also, real sound sources have complex dependencies of directionality upon frequency, and non-planar surfaces make the common analytical methods—ray-tracing and

image-source modeling—approximate at best. It can, therefore, be safely said that no auralization is perfect. However, neither is any audio recording perfect, yet we regularly enjoy them. Presumably there is some level of accuracy at which "good enough" will serve!

## Examples of Acoustic Archaeology

Two spectacular examples of applying auralization are in the specialty sometimes dubbed "acoustic archaeology." The first is a virtual recreation of a sermon preached by John Donne on November 5, 1622, in St. Paul's Cathedral and the surrounding churchyard. The fact that the edifice in which Donne actually preached was destroyed by fire in 1666, adds historical value to the project: We now have a way to experience visually and aurally a historical landmark that no longer exists. The first phase of the project was completed early in the present decade.

The first step was to create a visualization of the cathedral based upon contemporaneous drawings of the building and its surroundings (see **Photo 1**). In addition to artists' drawings, an architectural drawing by Christopher Wren still exists, showing the proposed addition of a dome. Aside from making possible a visualization of the Cathedral (see **Photo 2**), the contemporaneous artwork showed surrounding buildings, people, and animals (see **Photo 3**), which not only added to the visual virtual reality, but also to the "soundscape" or audio virtual reality including the aural environment of Donne's congregation. Donne's probable style of speech—early modern London dialect—was rendered for the auralization by linguist David Crystal.

The auralization utilized custom computer software, and the acoustical model was built based upon the visual model (see **Figure 2**). The results can be experienced with still photographs of the visual model, from two different locations: the sermon house and the cross yard. The auralizations can be accessed at: <https://vpcp.chass.ncsu.edu/listen-the-sermon>. For the best experience, the auralizations should be heard on excellent headphones, since they are in a binaural format. Computer speakers are simply not adequate to allow you to appreciate the full experience.

## Preserving the Sound of a Venue for the Future

The second remarkable project using auralization in acoustic archaeology began in

2013. The project includes a 3D visualization (see **Photo 4**) and binaural auralization of vocal and instrumental music in Notre Dame Cathedral. Images from the 3D model are linked with the auralization at corresponding positions in the acoustical model using a feature of CATT Acoustics V.9 called a "walkaround" (<http://www.lam.jussieu.fr/Projets/index.php?page=GhostOrchestra>). This method provides



Photo 2: Paul's Cross from 50' away. (Image courtesy of the visual model constructed by Joshua Stephens, rendered by Jordan Gray)



Photo 3: Detail from John Gipkin, Painting of Paul's Cross (1616) (Image courtesy of the Bridgeman Art Library, New York, and the Society of Antiquaries, London)

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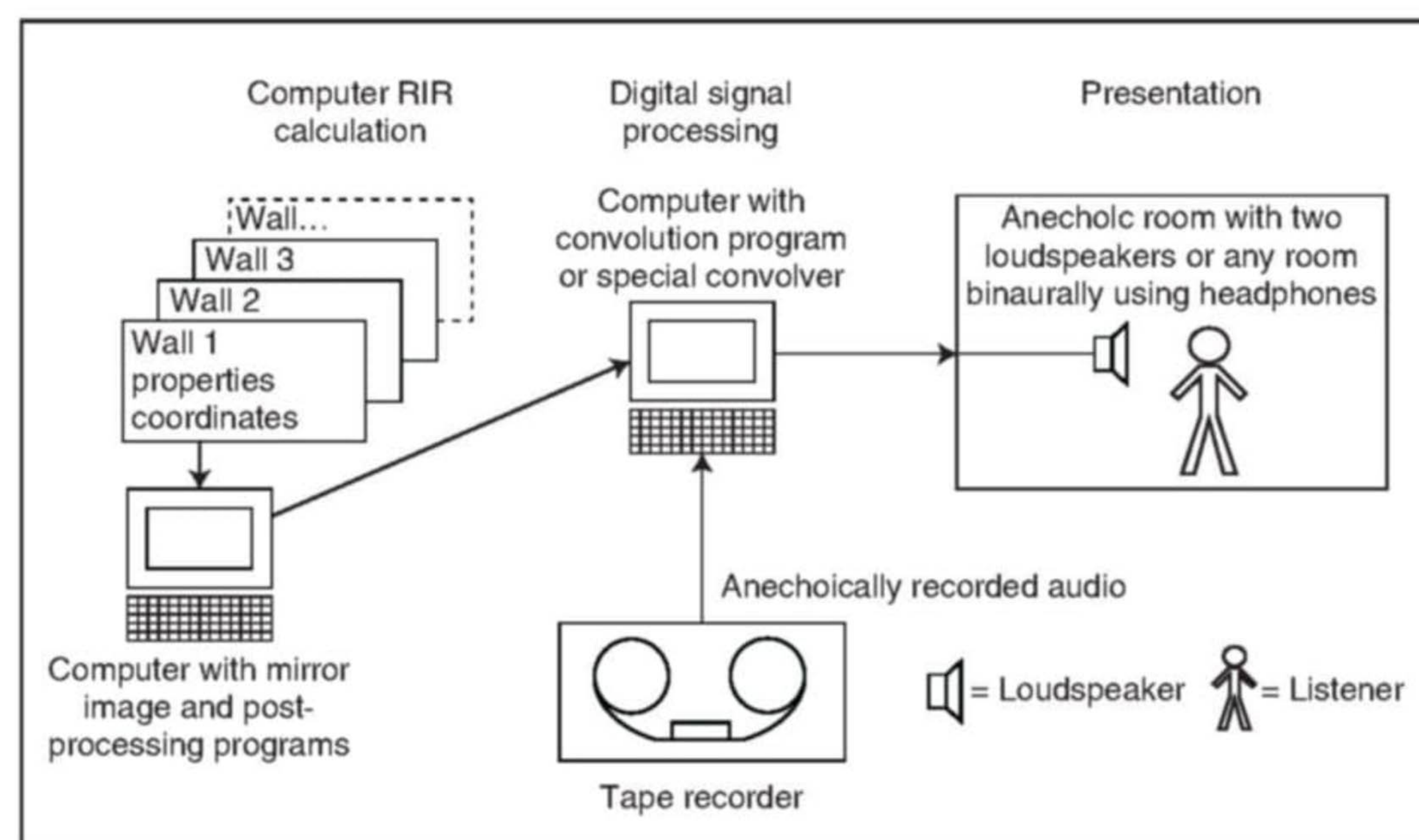
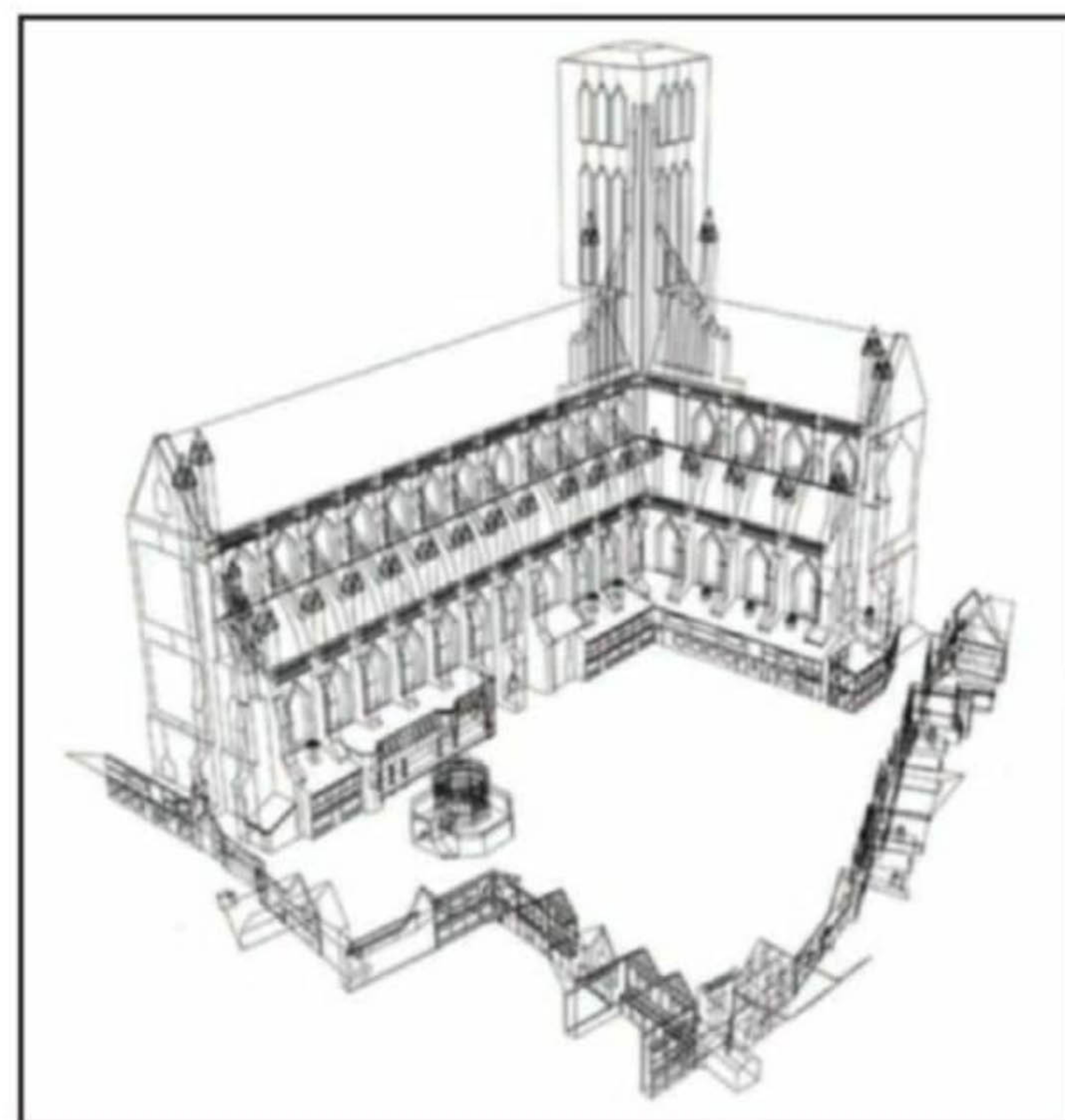


Figure 1: Basic principles of a system for fully computed auralization (Image courtesy of Kleiner, et al.)



Figure 2: Wire frame image of the acoustic model of Paul's Churchyard, the Cross Yard



the user with a virtual reality experience of actually walking around in the cathedral while listening to the music as the reverberation and other acoustical effects change. The 3D virtual concert can be experienced interactively with an Oculus Rift Head-Mounted Display (HMD). This real-time rendering was achieved with BlenderVR software.

The audio element utilized the close-miked audio tracks from the different musicians as sources for the auralization. This effort required the construction of a CATT Acoustics geometrical acoustic model, which was calibrated according to measurements also carried out in the cathedral. The 3D RIR is then numerically computed for each instrument and every potential listener position. These responses are then convolved with the corresponding audio track and appropriately

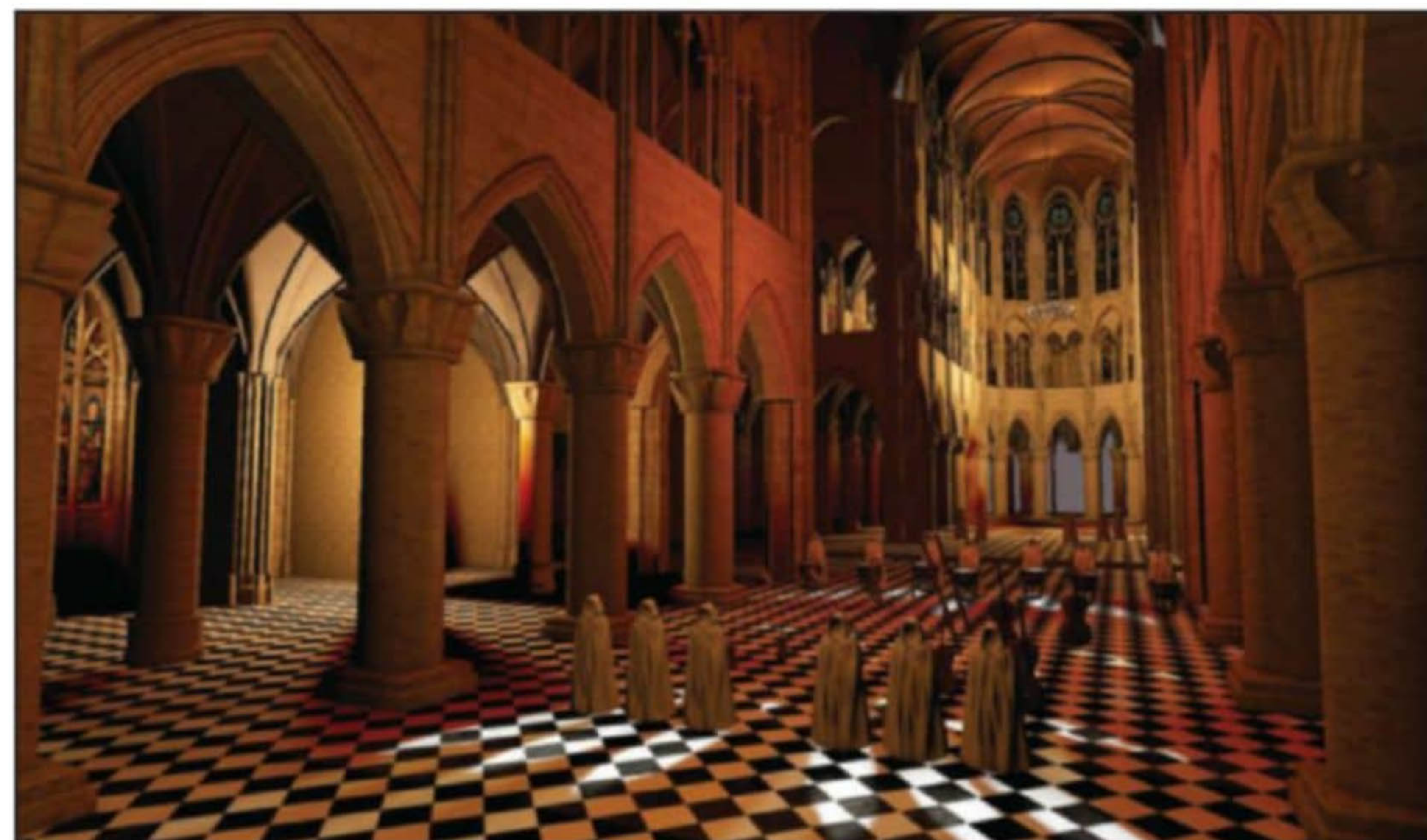


Photo 4: Image from 3D visualization of the Interior of Notre Dame Cathedral (Image courtesy of Lutheries Acoustique Musique)

combined to create the Ghost Orchestra performance.

The website referenced earlier in the article contains links to tracks with which the listener can compare (A-B) the auralized track for the conductor position with an actual mono recording made with the mic at the conductor position.

In the aftermath of the tragic fire that destroyed parts of Notre Dame Cathedral on April 15, 2019 (see **Photo 5**), the visualization and auralization take on particular historical value.

### More Practical, but Less Spectacular Uses for Auralization

When an architect is planning a new building—be it an auditorium, theater, church sanctuary, corporate board room, or whatever—the owner often wants to know how the completed building will sound. Sometimes the acoustical consultant makes recommendations that are met with objections from an architect or owner for aesthetic, budgetary, or other reasons. In such cases, auralizations showing the way the space will sound can be very helpful. If there are different options being considered, the acoustical effects of each option can be auralized. The different options may consist of details of architecture or sound system design. The EASE and CLF data files provided by most manufacturers of pro-sound speakers make such auralizations possible.

I personally include auralization as part of my acoustical and sound system design services for almost every new construction project and many renovation projects of public listening spaces. One example is a church fellowship hall for which I did the acoustical and sound system design. The speaker system I proposed was not among the best-known brands, and the church's technical staff wanted to know how they would sound in their new building. Since the building only existed in CAD drawings at that point in time, answering their request was not as simple as bringing in the speakers and setting up a demo. We used an auralization based on a CATT Acoustics model built from the architectural drawings of the hall. All acoustical design elements were incorporated, and the proposed speakers installed at the proposed locations and with the correct aiming. Thus, all doubts were allayed. The project was completed to the satisfaction of the owner, the architect, and the acoustical consultant.

A more critical application in which auralization proved its worth was a large conference room with a complex geometrical shape incorporating

many curved surfaces. The sound focusing created by some of these surfaces resulted in locations within the room where the speech intelligibility was poor. You may have been in a “whispering gallery,” in which a listener standing at one focal point of an ellipse can easily hear a person standing at the opposite focal point and whispering. The whisperer can be clearly heard even if other people are between the whisperer and the listener, and talking louder than the whisperer. In the case of this conference room, the effect manifested as areas of particularly high reverberation, although the average reverberation in the room was well controlled. The acoustical consultant was alerted to the problem by the small spots of high reverberation that showed up in the EASE AURA reverberation map of the room. The solution proposed by the acoustical consultant was rather expensive, and the owner and the architect were understandably hesitant to commit to the necessary expenditure. The consultant created an auralization of an anechoic speech track with the virtual listener located at one of the focal



Photo 5: The Spire of Notre Dame Cathedral on fire (Image courtesy of BBC News)

points. Upon hearing the poor speech intelligibility revealed by the auralization, the architect and the owner readily agreed to the consultant's proposed solution, and the problem was averted.

The computerized acoustical modeling tools that were introduced in the closing decades of the 20<sup>th</sup> century have provided us with invaluable abilities to create successful acoustical designs. Fully computed auralization, which is an outgrowth of these tools, is a great help in identifying and solving problems, and in communicating possible acoustical problems. **ax**

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# The DPA d:vice MMA-A Mobile Recording Interface

DPA has debuted an accessory called the d:vice MMA-A, which is meant to allow users to plug the d:vote mics into an iOS device, providing both an interface and phantom voltage. Stuart Yaniger tests it out and gives us his opinion.

By  
**Stuart Yaniger**  
(United States)

Not so long ago, serious field recording meant much lifting and carrying of heavy objects, lots of searching for power sources (or carrying even more weight), and in the case of video recording or television/radio production, the need for at least two people. Like everything else, field recording has been completely transformed. My current audio recording gear, which comprises a laptop, a stereo microphone, a USB interface, and associated cabling, fits into a medium-sized attaché case and might weight 4 or 5 kg. There's likely an audio equivalent to Moore's Law, and as one would expect, the next step involves the little audio and video recorders that everyone carries around with them these days. Like most people's, mine is an iPhone.

The improvements in the camera and optics of smartphones get the most attention, but most readers of *audioXpress* are also quite aware of the great sound capabilities of the current generation of these devices. But there are limitations, and those include the built-in microphones (generally very cheap electret or MEMS) and form factor of the phone, which determines the microphones' acoustic environment. So we sometimes (ok, often) see well-shot videos, but they have poor sound.

Some videographers and musicians will use a separate microphone that plugs into the smartphone—in the case of iPhones, this is a Lightning connector. There are a lot of choices here at the inexpensive end of the market, but not nearly as many high-quality professional-grade microphones.

I recently reviewed the DPA d:vote 4099 microphone (see Resources), which had the virtues of high directionality, low

distortion, rugged construction, and ultra-light weight. This would seem to be a perfect sort of device for use with a smartphone, and DPA has a wide range of high-quality mics optimized for different applications. But of course, one still needs a way to plug it into a Lightning or microUSB port (depending on one's use of Apple or Android) and control the recording.

## The Product

Recognizing this need, DPA has debuted an accessory called the d:vice MMA-A. The MMA-A is meant to allow the d:vote mics to plug into an iOS device, providing both an interface and phantom voltage. There are two input channels and no analog output. According to René Mørch, the product manager for DPA, "The MMA-A Digital Audio Interface was designed with broadcasters and mobile journalists in mind, but can also be used by other content makers for recording music, filming on location, and during both live and recorded broadcasts. With this interface, users can portably record and stream clear and professional audio from the connected microphones."

The MMA-A is about the size and shape of an Oreo cookie (see **Photo 1**), and has absolutely no controls on it, just two inputs and a microUSB port (see **Photo 2**). DPA's proprietary input interface is called MicroDot. The MicroDot system provides the DPA microphones with phantom voltage (strictly speaking, the interface is a constant current one, similar to the system used for measurement mics, see the article in Resources) and receives the analog signal from the mic (see **Photo 3**). The connector is a miniature screw-on coaxial, and DPA provides adapters for connecting a MicroDot microphone to conventional XLR balanced inputs, which is how I've been using the 4099 mic. With the MMA-A, the DPA mic just connects directly via its MicroDot plug—and that's the only connection option.

There's also a microUSB port for connection to the recording device, along with provided cables to connect either to a Lightning port or a USB port. Connection to my iPhone was painless (see **Photo 4**). This seems obvious, but at the same time that I had the MMA-A, I also had on hand another iOS-compatible interface, this one from CEntrance, the MixerFace 4, which (although more versatile than the MMA-A) was very picky about which USB-to-Lightning cables were used. For example, if I used the MMA-A's cable or any other third-party microUSB to Lightning cable, the iPhone would not recognize the MixerFace. I had no such problems with the MMA-A and my iPhone, regardless of cables and adapters- this could be



Photo 1: The DPA d:vice MMA-A microphone interface is compact and is equipped with no physical controls.



Photo 2: The MMA-A has two MicroDot inputs and connects to iOS devices via a microUSB port.



Photo 3: The MicroDot system provides the DPA microphones with phantom voltage and receives the analog signal from the mic.



critical in situations where a longer connecting cable might be needed.

Input level and some equalization options are all controlled by a free iOS app called, fittingly, “d:vice.” This app allows you to choose channel assignments, set levels, and apply an 80 Hz second-order low-cut filter (see **Figure 1**). Once your mic and interface are configured as you want it, the lock button freezes the selections, and you can begin by opening your recording software. One other thing to note is that if you are using a single mic, the MMA-A will put out a mono signal on both channels



Photo 4: Connection to my iPhone was painless.



Photo 5: DPA provides a rugged foam-lined case for the MMA-A's transport. (Photo courtesy of Cynthia Wenslow)

unless you manipulate the settings to limit it to a single channel. 99% of the time, their default will be preferable.

As befits an interface intended for professional field recording, the MMA-A is packaged in rugged fitted foam carrying case (see **Photo 5**), but it is small enough to be just dropped into a pocket if space and weight are at a premium—the rounded disk shape means that it won't poke holes in pockets, for which I was grateful in field use, and it's solid enough that the jostling around won't break it. The price for the MMA-A is about \$660, including the case and cables. This is not inexpensive, but it is intended to be used with DPA's top-grade mics, which are likewise priced (and designed) with professional use in mind.

### The d:vice MMA-A in Use

First and foremost, it is important to note three major application limitations for the MMA-A: It is designed for iOS only (no drivers for Android or PC), it can only be used with DPA's MicroDot

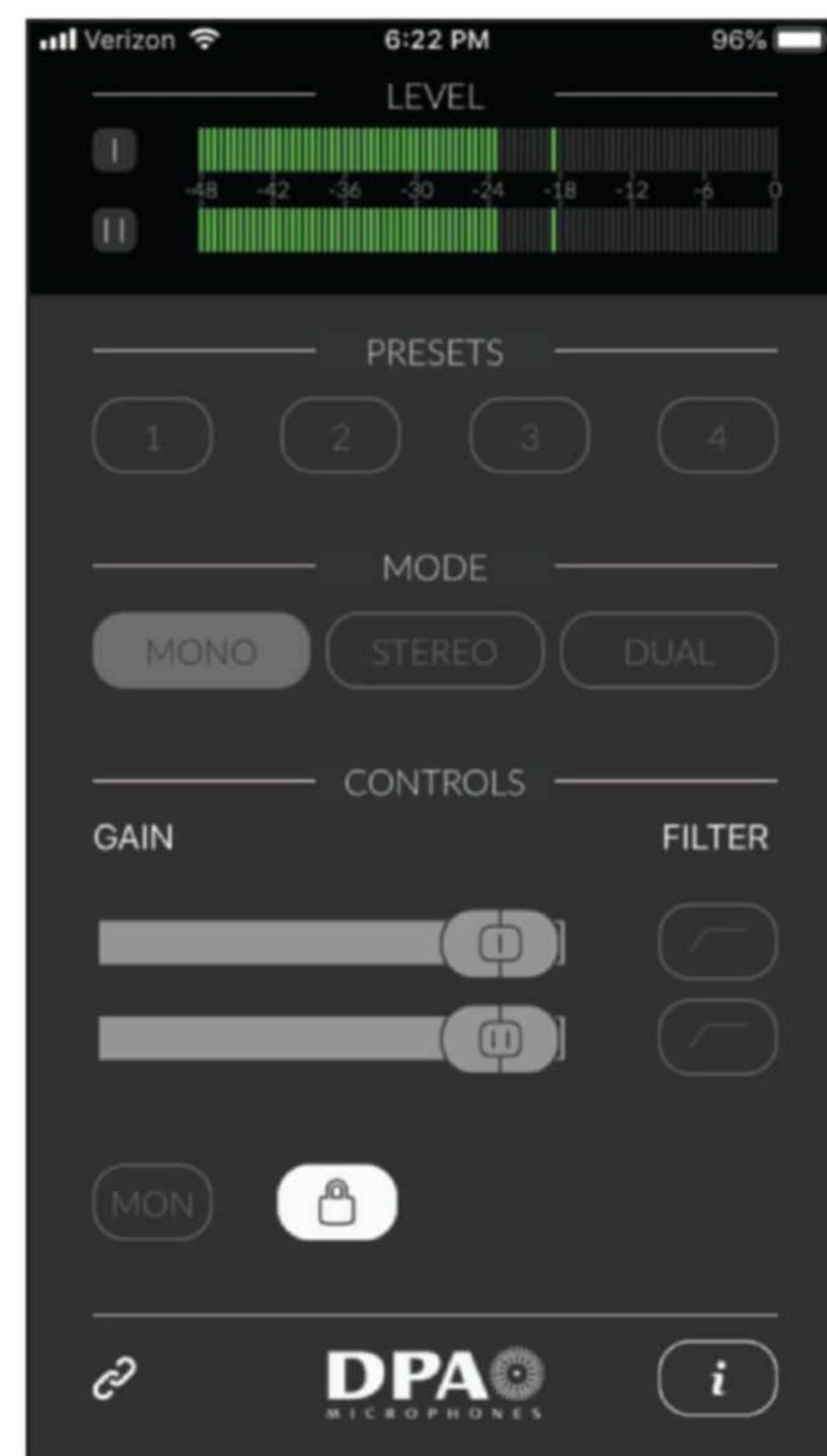


Figure 1: Operation mode and levels for the MMA-A are set by a free iOS app called d:vice.

mics (no other mics nor line inputs), and it has no output beyond the USB port. The first limitation should be qualified a bit—there is no PC or Android driver for the MMA-A, but a third-party ASIO driver could potentially be used. I tried this with ASIO4ALL and got rather mixed results. My usual recording software is Adobe Audition, but I could not get the MMA-A plugged into my PC to work at all with it. As I will show in the measurements section, ASIO4ALL and the MMA-A may not be a match made in heaven.

Turning to the actual intended use, I purchased a copy of the n-Track Studio app (one of the recording apps DPA recommends) for my iPhone 8, grabbed the d:vote 4099 mic I still had on hand after reviewing it, and plugged everything in (see **Photo 6**). And delightfully, it all worked right from the start.

I set levels with the d:vice app, locked them, then fired n-Track back up and made my first recording—in this case it was me brutalizing my guitar in an attempt to play Peter Mulvey's “Sad, Sad, Sad, and Far Away From Home.” Between the clarity of the 4099 mic and the seamlessness of the interface, the playback revealed my poor picking and clumsy fingering in full detail. More importantly, the hiss was below the ambient noise in my room, which was also faithfully captured, although the highly directional character of the 4099 mic certainly helped here.

Next, I created a short video with sound, in this case a cooking video (my cooking is better than my guitar playing). For this, I bought a copy of the MoviePro app. As before, the MMA-A allowed the mic to interface seamlessly with the iPhone and the video software, and I was able to get a recording that sounded far, far better than ones made with the built-in iPhone mics. By contrast, getting the MoviePro app to recognize and interface with a competing (though more flexible) interface took a bit of wrestling and searching through Settings menus.

Unfortunately, for my next attempt, the limitations of the MMA-A proved insuperable. The n-Track app allows musicians (and hacks like me) to create multitracked performances. My intention was to track two guitars, a bass, and a flute, which should be relatively simple. This, however, proved infeasible because of the lack of outputs on the MMA-A. For the last few generations of iPhones, there is no separate headphone jack, and all connections go through the Lightning port. Apple provides an adapter to allow standard wired headphones to be plugged in, but that requires the Lightning port to be available. So with the MMA-A plugged in, there's no way to connect wired headphones for monitoring.

There are a few possible solutions. First, one can use an older model of iPhone where the headphone jack still exists. However, if what you have



Photo 6: Connecting microphones to an iPhone through the MMA-A is simple and straightforward, creating a very portable and high quality recording system. (Photo courtesy of Cynthia Wenslow)

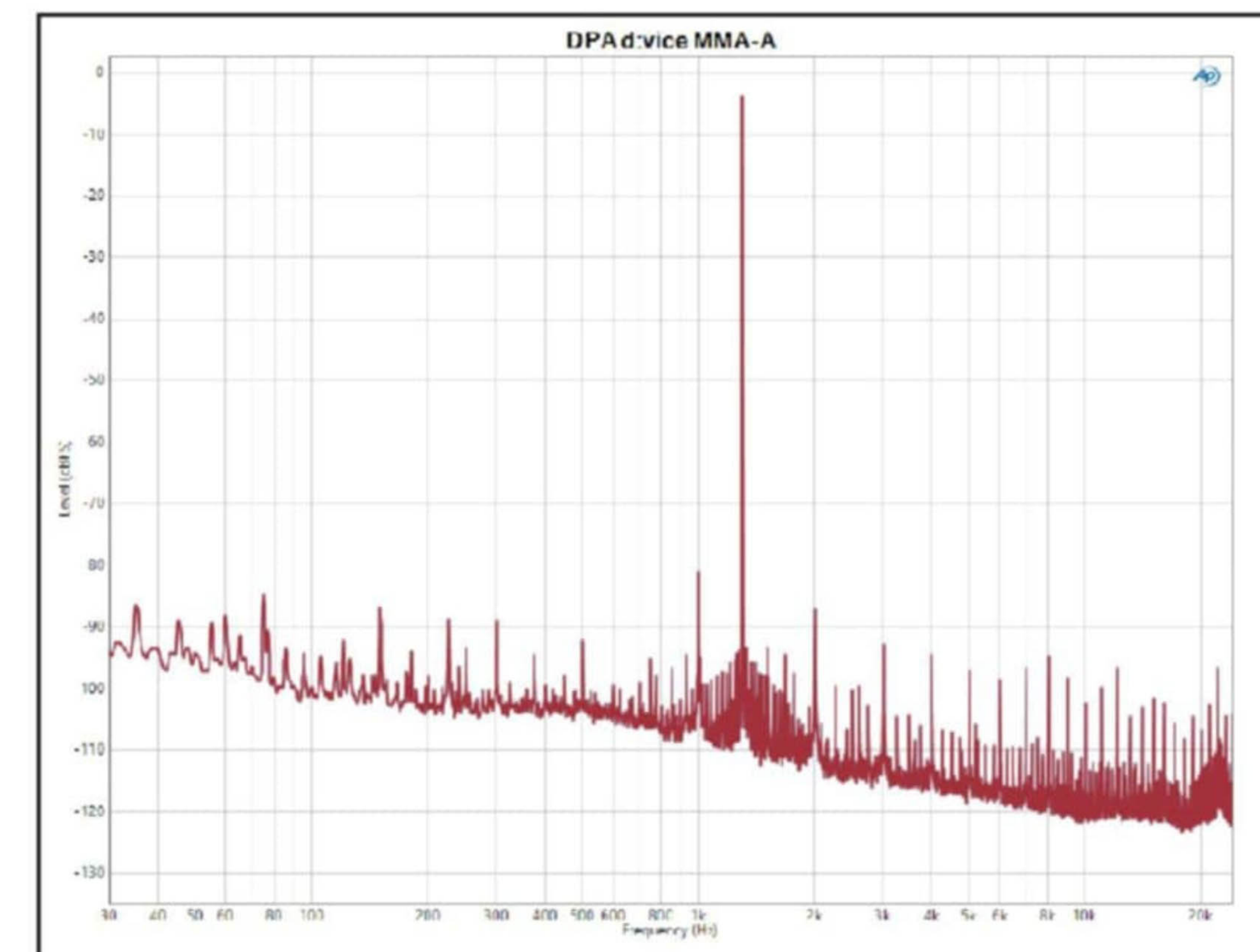


Figure 2: With the MMA-A connected to a PC through a USB port and a third-party ASIO driver, the MMA-A shows significant amounts of artifacts spaced at 1 kHz intervals.

### About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as a technical director for a large industrial company. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.



is an iPhone 8 or 10, it seems counterproductive to have to buy a separate phone for recording, not to mention the downgrade in the camera. As well, it is not exactly unknown for older iPhones to suddenly develop performance slowdowns as the operating systems are updated...

Another option is to use an Android device. This seemed appealing to me, given that, although I own an iPhone, I am very antipathetic to Apple's concept of an electronic ecosystem, and find simple tasks like file transfers to be overly clumsy and non-intuitive. Nonetheless, I tried using the MMA-A with a borrowed Android phone. Without an Android driver, it proved

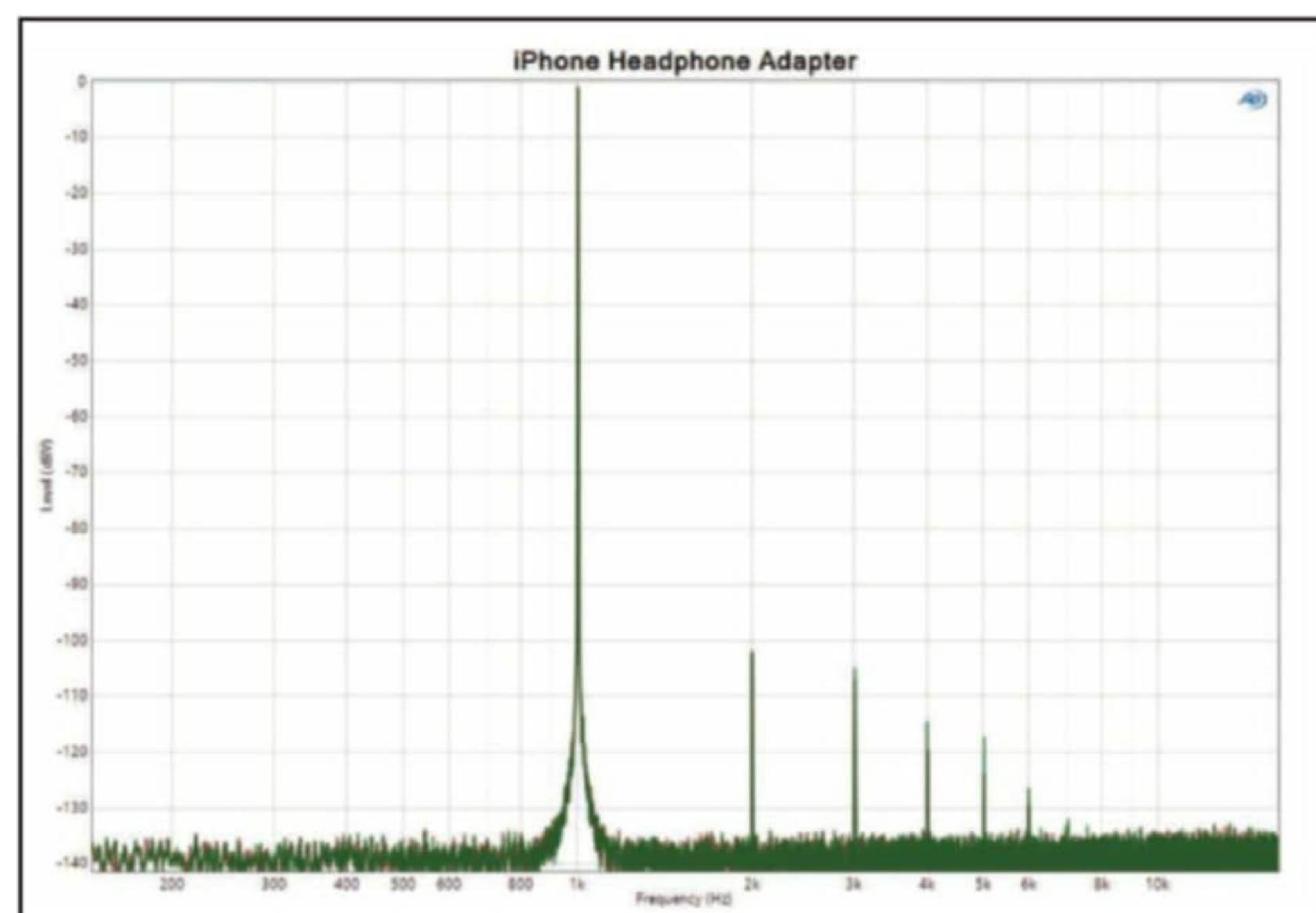


Figure 3: Playing back a WAV file of a generated sine wave through the iPhone's headphone adapter results in a very clean and low noise signal spectrum.

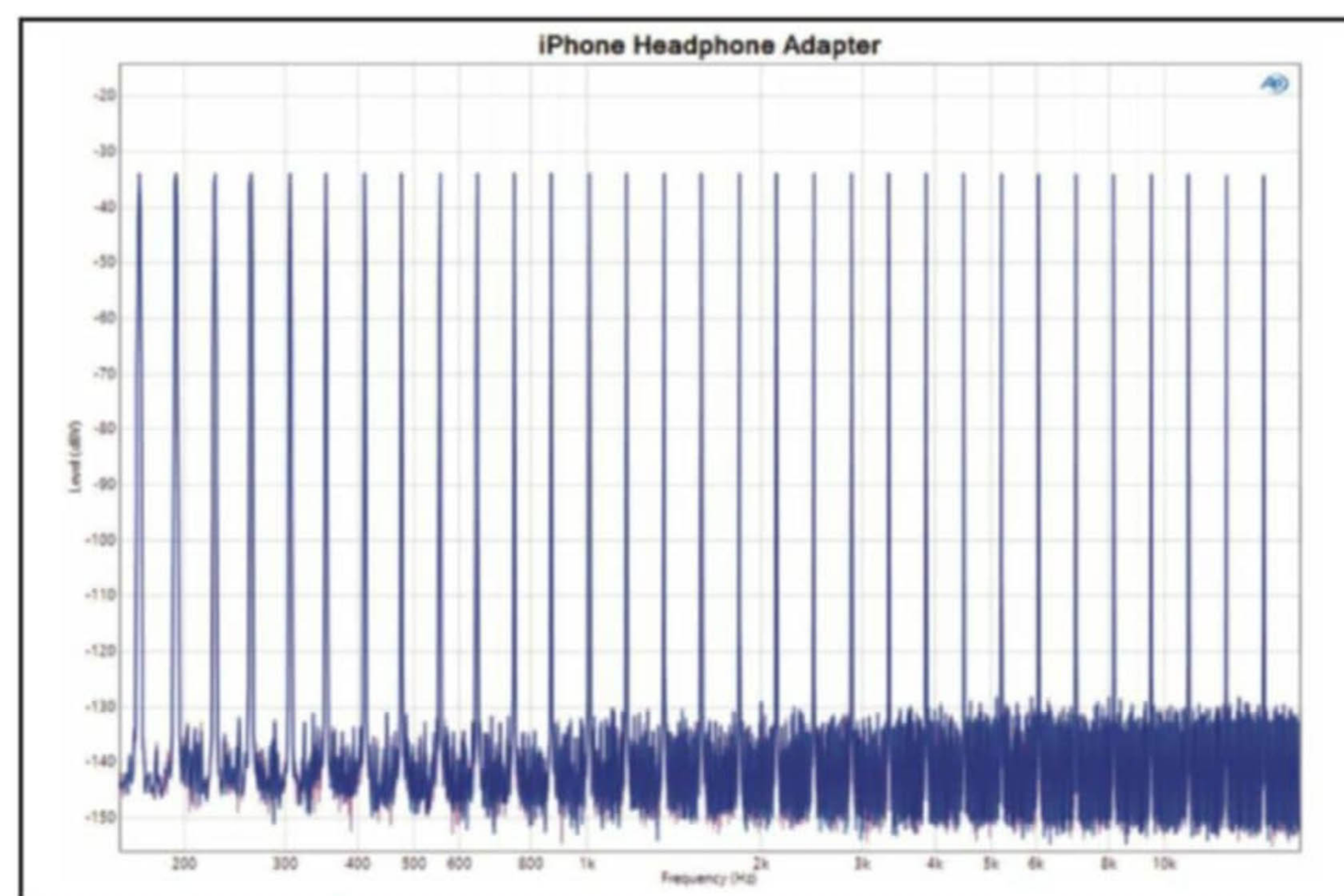


Figure 4: Using the same setup as shown in Figure 3, a generated multitoned WAV file playback shows an excellent spectrum.

to be a much more difficult task than, say, using a more conventional interface. It is not an experience I would recommend, and lacking the ability to use the d:vice app, nothing ran optimally.

The option recommended by DPA was to replace my conventional headphones with Bluetooth. I borrowed a pair of Bluetooth headphones and indeed, I could play back tracks while recording. Unfortunately, latency was an issue—although the timing and precision of my playing are not exactly state-of-the-art, the latency made doing overdubs of even mediocre quality nearly impossible. All of this would be trivially easy if the MMA-A had either a headphone jack or line outputs.

In my communications with Mørch, he elaborated on DPA's decision to omit monitoring capabilities. Mørch pointed out that there was a 100 mA limit to current drawn through the Lightning port, that the MMA already drew close to that, and that consequently there was no current budget left to drive headphones. This may be true (I don't know if the current to the ADC or Lightning interface can be reduced), but surely a line out option would not draw much current and would allow "studio on the move" capabilities to touring musicians and people integrating voice with video.

## Measurements

This section will be a bit less comprehensive than usual, due to the practical iOS-only limitations of the MMA-A. This became clear when I ran the USB output to the lab PC that I use for Audio Precision measurements, injecting an analog signal into the MicroDot ports and using the ASIO4ALL driver to look at the output. **Figure 2** shows the spectrum of a 1.3 kHz sine wave obtained this way—there's a series of artifacts at 1 kHz intervals at a moderately high level, which obscures the rather low harmonic distortion results and reduces the measured dynamic range. Clearly, the MMA and PC do not play well together.

As a workaround, I connected the MMA-A to my iPhone 8, ran the analog signals from an Audio Precision APx525 analyzer to the mic inputs of the MMA-A, then used the n-Track app to record the signals. Then I would unplug the MMA-A, plug in Apple's headphone adapter, and play back the signal to be captured by the AP analyzer.

It's a rather clumsy way of doing things, with much plugging and unplugging needed for each measurement, and there's certainly room for error. However, in a world where iOS and Windows are sworn mortal enemies, peacekeeping missions are not going to be easy.

To check the capability of this setup, I ran some baseline tests by playing back some WAV files directly.

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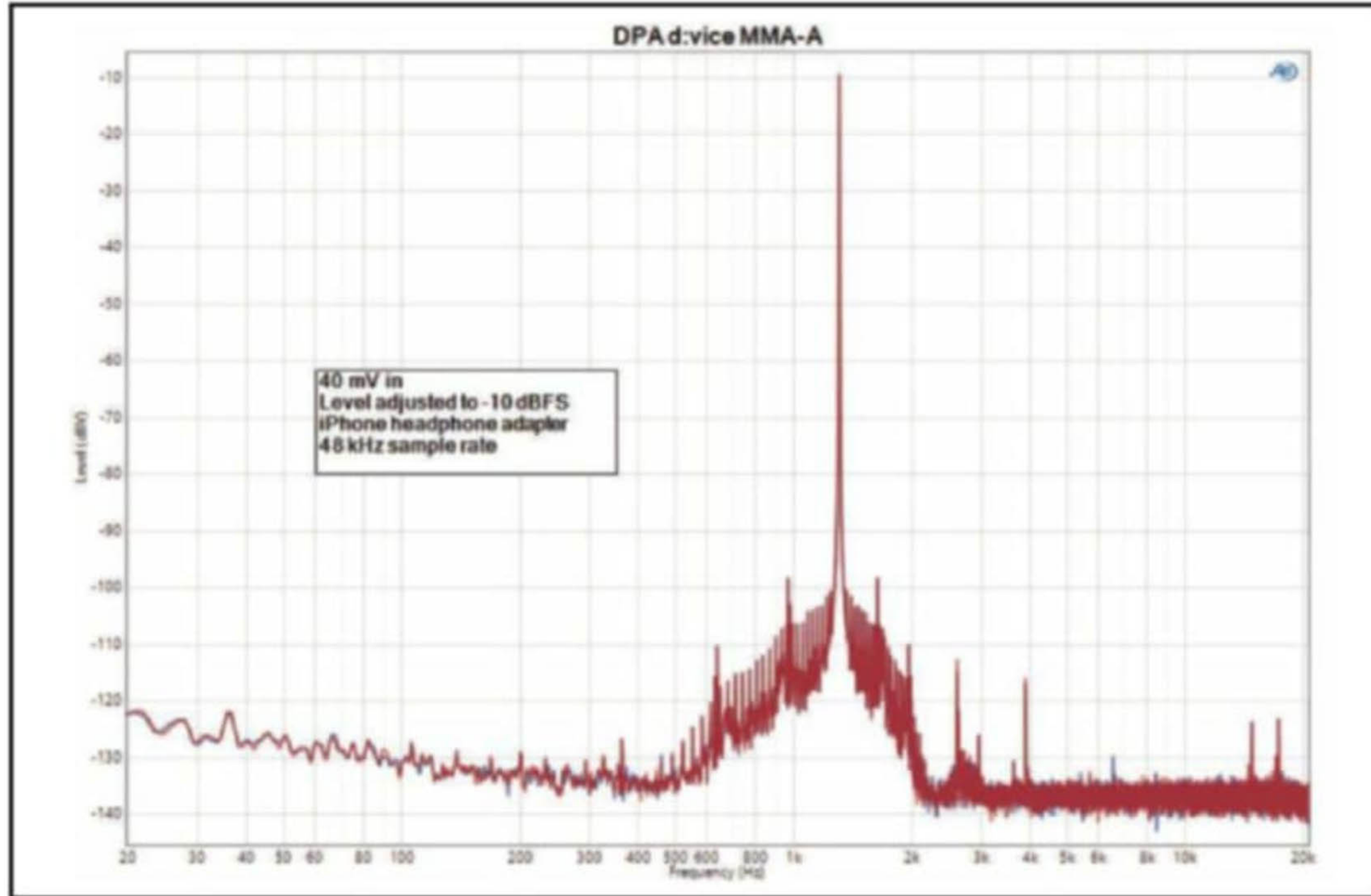


Figure 5: This is the spectrum of a 1.3 kHz sine wave recorded through the MMA-A and played back through the iPhone headphone adapter.

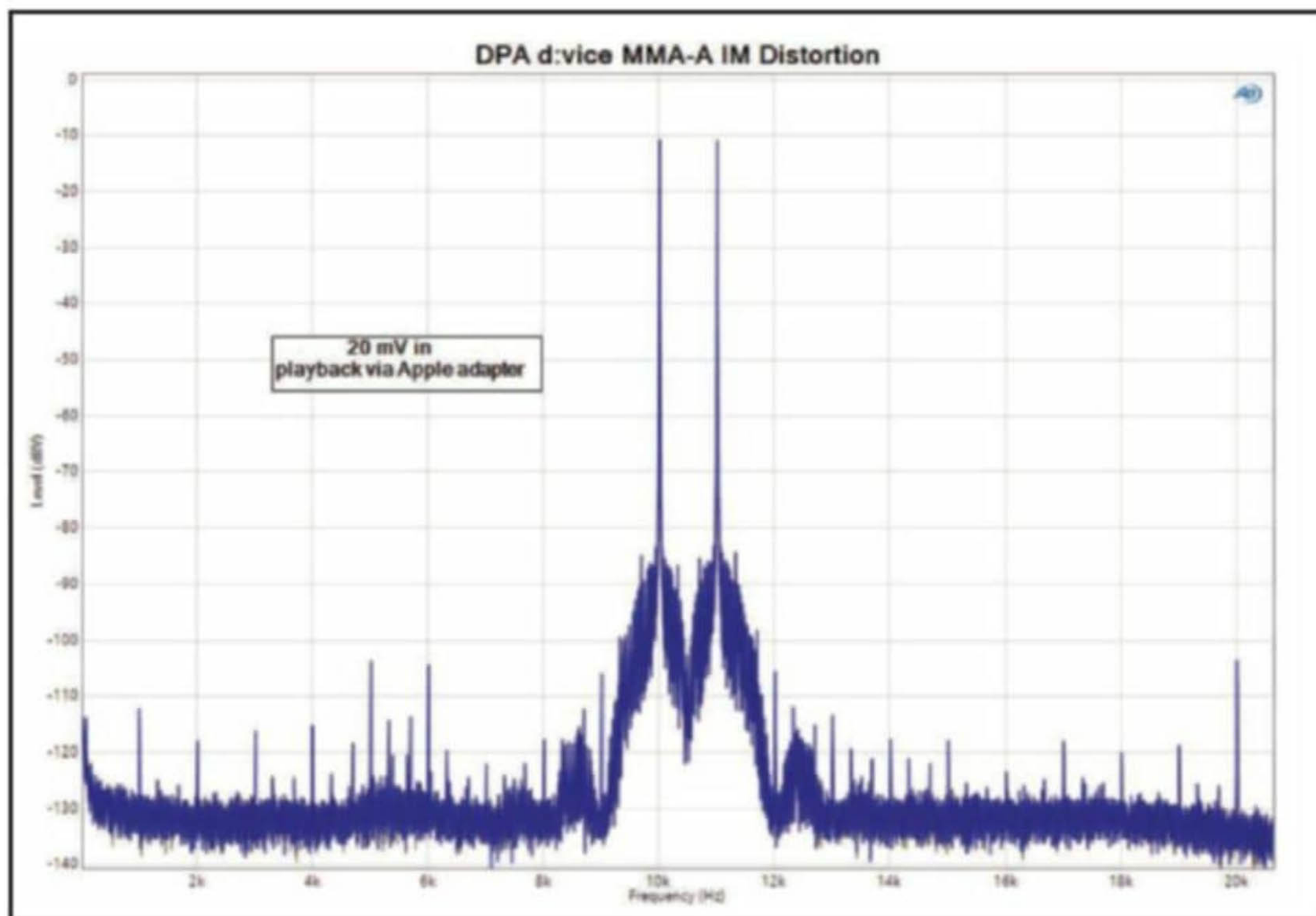


Figure 6: A 10/11 kHz intermodulation signal results in a low first-order intermodulation product, but higher (though still fairly small) levels of sidebands.

**Figure 3** and **Figure 4** show the spectra of a 1 kHz sine wave and a multitone signal, respectively. Although not the subject of this review, it's clear that the Lightning-to-headphone DAC and amplifier are remarkably good, especially considering the under \$10 price tag. This is making me rethink the need for anything fancier for music playback in my home audio system!

With the capabilities of this platform established, I moved on to the MMA-A measurements. First, a frequency response sweep confirmed that the MMA-A was flat above 40 Hz, with no signs of ripple at high frequencies. There was a very slight low frequency roll-off below that point, with 20 Hz being down by 0.5 dB. This suggests that the -3 dB point is below 10 Hz, a more than satisfactory result. I also checked the overload capabilities.

One of the advantages of a closed system such as the MMA-A with the MicroDot mics is that levels are well defined and will almost automatically match. With the MMA-A gain turned up to maximum, the overload point was at 40 mV in, which corresponds to about 110 dB SPL for the 4099 mic. With the gain turned down, the overload point rose proportionately, so the MMA will not be the determining factor for maximum SPL capability.

**Figure 5** shows the spectrum of a 1.3 kHz sine wave taken at -10 dBFS. The harmonic content is quite low, with second and third harmonics being well below -110 dBFS, but there are noticeable sidebands and a baseline spreading around the fundamental, which could indicate a slight jitter issue.

It should be noted that these artifacts are at roughly -100 dBFS and lower, so their audibility in most situations is questionable. Likewise, the spectrum of a two tone 10 kHz/11 kHz signal (see **Figure 6**) shows intermodulation products well below -100 dB, but the same spread and sidebands can be seen as before. Unlike the very versatile CEntrance MixerFace R4 I had for a short time, the ultrasonic range of the MMA is quite clean and free of significant artifacts.

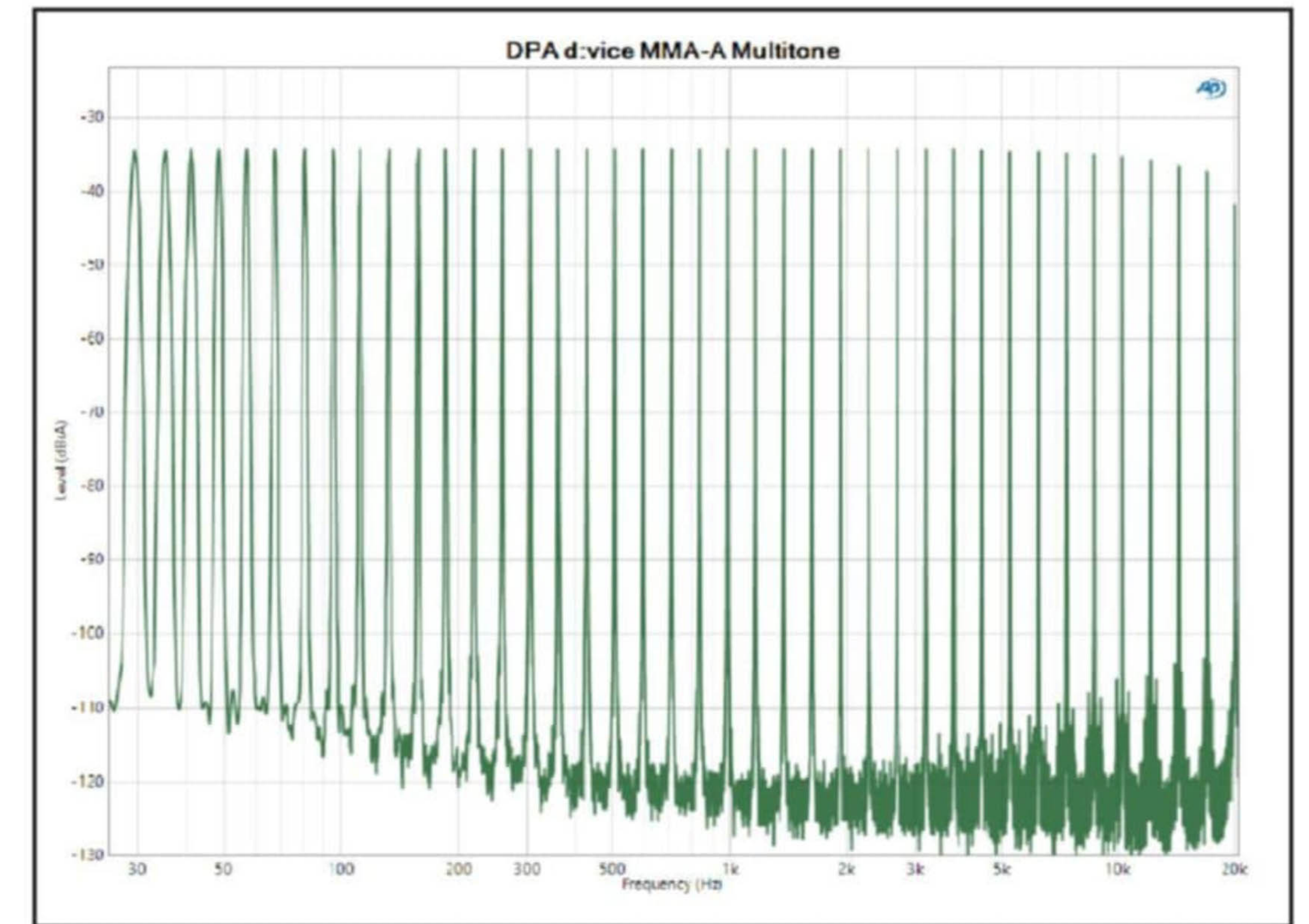
The spectrum of a multitone signal is shown in **Figure 7**; while not perfectly clean, the artifacts again are quite low in amplitude and would not be expected to cause audible issues. I should also mention that ultrasonic artifacts are similarly low, in contrast with the significant idle tones I've seen in other interfaces made for this application.

In summary, the measurements (with the caveat that there are some uncontrolled variables) are not perfect, but indicate performance more than good enough for the intended application.

## Wrap-Up

The MMA-A is a slick piece of engineering and industrial design. I always had the urge to slip it in my pocket! If you're doing live, single-take recording, nature sounds, or video journalism, and want top-level audio, the combination of a DPA mic and the MMA-A will offer portability, ruggedness, ease of use, and high performance. It is not inexpensive, but its price is in line with DPA's microphone offerings.

The limitations are likewise quite severe—the MMA-A is really only suitable for iOS, your choice of mics is strictly limited to DPA's line, there is no facility for monitoring, and for music production or other



overdubbing it's a coin flip whether you'll be able to overcome latency issues. If these are not critical to you, I can recommend this d:vice, but I suspect that the real market for this is rather limited. If I were DPA, I would be seriously considering development of Android and PC drivers and the addition of some sort of monitoring output, both of which would greatly broaden the market for this otherwise-fine device.

Figure 7: The spectrum of a multitone stimulus signal is reasonably clean but one can see low levels of sidebands and some spectral spreading increasing at higher frequencies.

## Resources

S. Yaniger, "Fresh From the Bench: DPA d:vote 4099 Instrument Microphone," *audioXpress*, February 2019.

S. Yaniger, "Innovative Measurement Microphones from PCB Piezotronics," *audioXpress*, December 2016.

## Sources

**DPA d:vice app**  
iTunes | <https://itunes.apple.com/gb/app/d-vice/id1182987675>

**MoviePro app**  
iTunes | <https://itunes.apple.com/us/app/moviepro-video-recorder/id547101144>

**n-Track Studio software**  
n-Track, S.r.l. | <https://ntrack.com/digital-audio-workstation.php>

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# Bringing 6DoF to Live Recorded Audio and Video

By  
**Piotr Szczechowiak**  
(COO, Zylia)  
and **Tomasz Żernicki**  
(CTO, Zylia)

Imagine a popular music festival you've always wanted to experience in person—seeing the performers on stage and being surrounded by the music, the people, and the energy of the show. For one reason or another, you can't make it to the festival. But what if you could teleport yourself there? What if you could plug into an immersive experience that allows you to move freely through the whole audio soundscape and visual landscape? This is the promise of Six Degrees of Freedom (6DoF).

True 6DoF experiences enable freedom of movement in 3D space, or along all three movement axes and all three rotational axes. Within this environment a person can move forward/backward (surge), up/down (heave), and left/right (sway), and also rotate—yaw (normal axis), pitch (transverse axis), and roll (longitudinal axis)—to change orientation.

With 6DoF technology in VR/AR, you can experience 3D audio/video content from an unlimited number of listening and vantage points. Wearing an appropriate VR headset and headphones, you could use 6DoF to go anywhere. You could walk along a tropical rainforest path, taking in the sights and sounds of animal life all around you (see **Figure 1**). You could amble among palm trees on a sunny beach, listening to the waves crashing on shore.

While some video games offer virtual worlds that effectively enable 6DoF movement, the real potential of 6DoF lies in exploring the real world and actual

places and events within it. The technology and techniques that enable development of real-world 6DoF do exist, and audio and video creatives are just starting to explore the possibilities.

## The Advance Toward 6DoF

Until very recently, immersive audio and video were limited to just Three Degrees of Freedom (3DoF). Examples include the 360° videos now available on platforms including YouTube, Facebook, and Vimeo. Sometimes also referred to as immersive videos or spherical videos, these 360°-video offerings incorporate video recordings shot simultaneously using either an omnidirectional camera or a collection of cameras.

Images are stitched together to build a panoramic scene, which can be viewed on a laptop or mobile device, on a head-mounted display, or within a purpose-built space equipped with projectors or displays in a spherical configuration. While people

Photo 1: Zylia, applying third-order Ambisonics ZM-1 microphone arrays with Infineon's digital MEMS microphones, has an innovative and unique approach to Six Degrees of Freedom (6DoF) technology in virtual and augmented reality (VR/AR).

viewing or listening to a 360° or 3DoF scene can rotate to look in different directions or to hear audio from different direction, they themselves remain static—they cannot freely translate their position in space.

The 360° audio is similarly recorded. Algorithms within associated software convert the signals from the recording device into the Ambisonics domain. Ambisonics is the full-sphere surround sound format that has become the standard for 360° filmmakers and video game developers, and it covers not only the horizontal plane, but also sources "above" and "below" the listener (see **Figure 2**).

Unlike other multichannel surround formats, the Ambisonics format carries speaker-independent representations of a sound field called B-format over its transmission channels. The sound field then can be decoded for playback on the target system or device. This approach gives creatives and engineers the flexibility to design sound in terms of source directions rather than loudspeaker positions.

**Photo 1** shows Zylia, applying third-order Ambisonics ZM-1 microphone arrays with Infineon's digital MEMS microphones, has an innovative and unique approach to Six Degrees of Freedom (6DoF) technology in virtual and augmented reality (VR/AR).

First-, second-, and third-order Ambisonics, respectively, offer higher spatial resolution. The higher the Ambisonics order, the better the sound source localization in the space. Research shows that it is essential to have at least third-order Ambisonics representation in order to fully experience the 3D audio that is critical to creating a truly immersive experience.

Higher-order Ambisonics allow mixing engineers to create 3D sound tailored to headphones so that the listener experiences panoramic audio. Although this technique—binaural reproduction or binaural recording—is not new, the capture of audio with an Ambisonics microphone and decoding specifically for headphones makes for a listening experience that rivals reality.

In a 360° production, the 3DoF approach to video and audio might allow the viewer to jump between different positions at a concert or other live event. 6DoF takes the experience to a whole new level by enabling the viewer to move freely throughout the scene.

Additionally, with the right software tools, typically plug-ins for digital audio workstations (DAWs), it is possible to process the data captured by the multitrack recording device and perform adaptive isolation of sound sources using a virtual microphone. The virtual microphone is, in fact, a spatial audio filtering technique based on

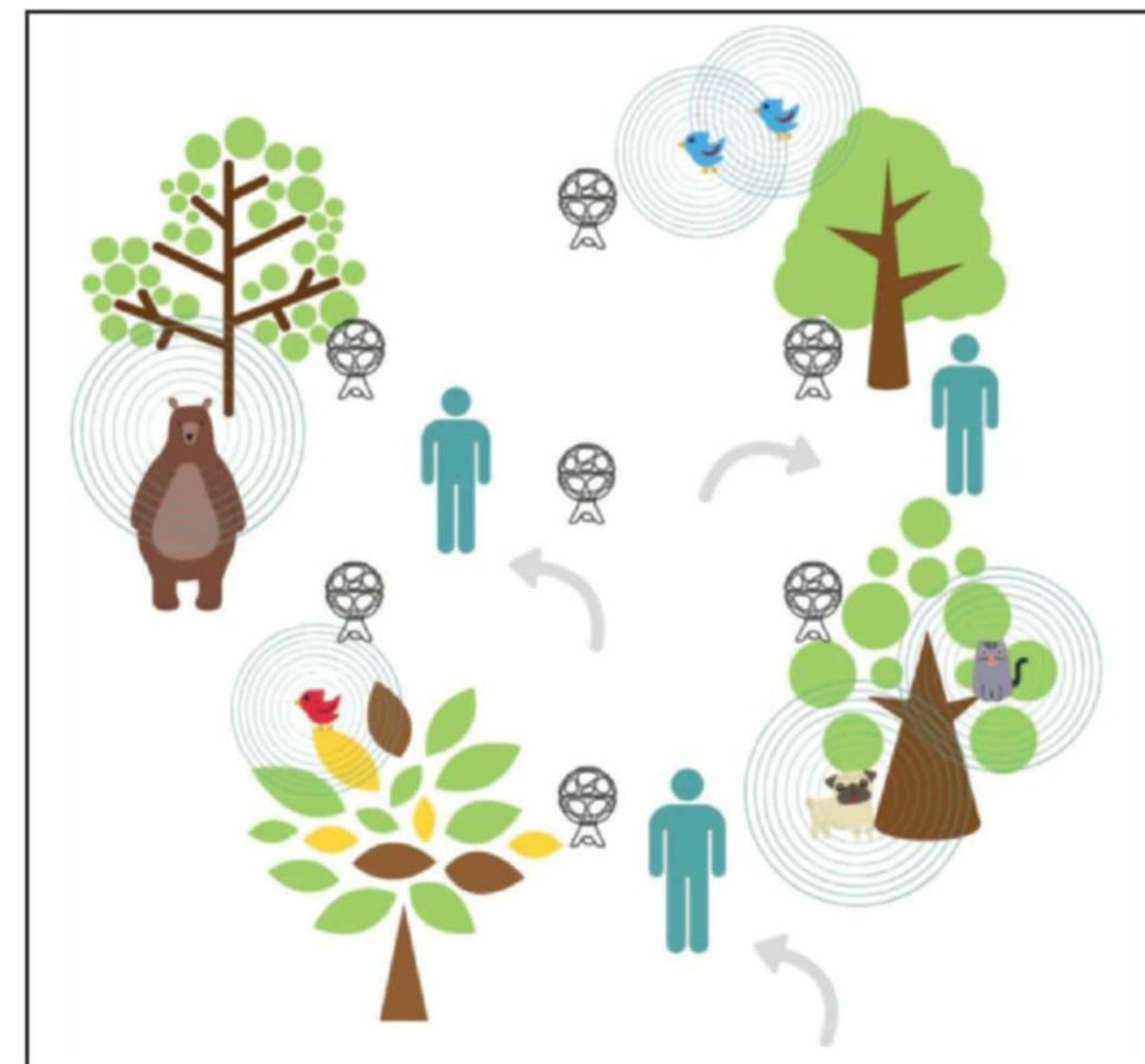


Figure 1: 6DoF technology in VR/AR enables the user to experience 3D audio/video content in the virtual space by introducing an unlimited number of listening and vantage points. Wearing an appropriate VR headset and headphones, you could walk along a tropical rainforest path, taking in the sights and sounds of animal life all around you.

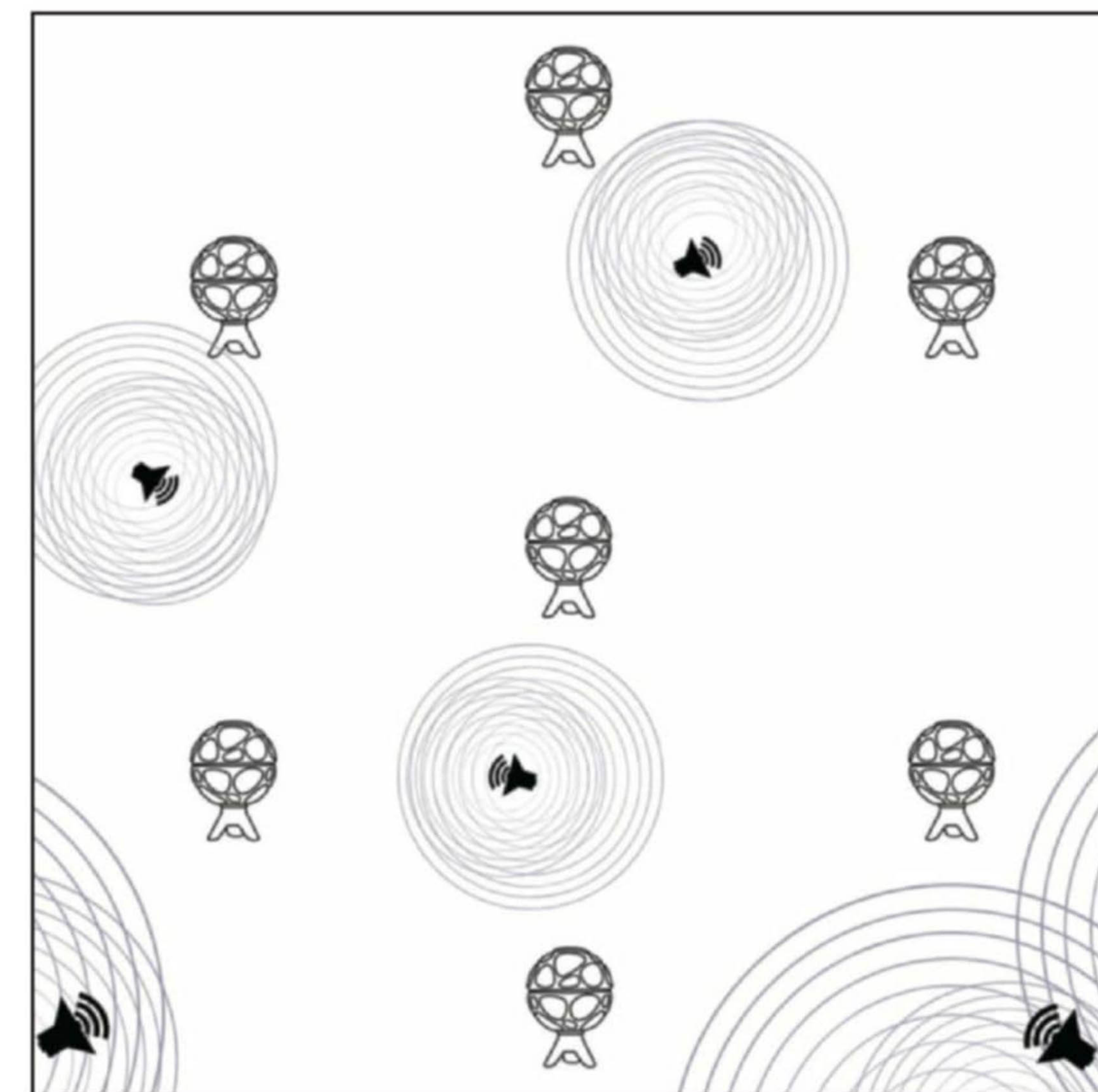


Figure 2: 6DoF technology with Ambisonics covers not only the horizontal plane, but also sources "above" and "below" the listener.



advanced beamforming and sound source separation algorithms, and it accounts for both phase and amplitude differences between signals captured by microphone arrays on the recording device. Each virtual microphone can be configured to with unique spatial characteristics, width, or shape, and each can be used as a single-spot microphone with super-narrow directivity response or with other virtual microphones in a multichannel surround recording set.

For the creative user or engineer, the process of isolating sound can be as simple as pointing the virtual microphone at a particular voice or instrument. Software algorithms interpolate any given position in the scene, allowing desired sounds to be emphasized and unwanted sounds suppressed. Adjustments such as balancing direct and reverberant sound can be fine-tuned during mixing. With the ability to change a spatial characteristic of virtual microphone in postproduction, musicians and audio engineers can avoid the need to repeat pre-recording and recording steps. The high spatial

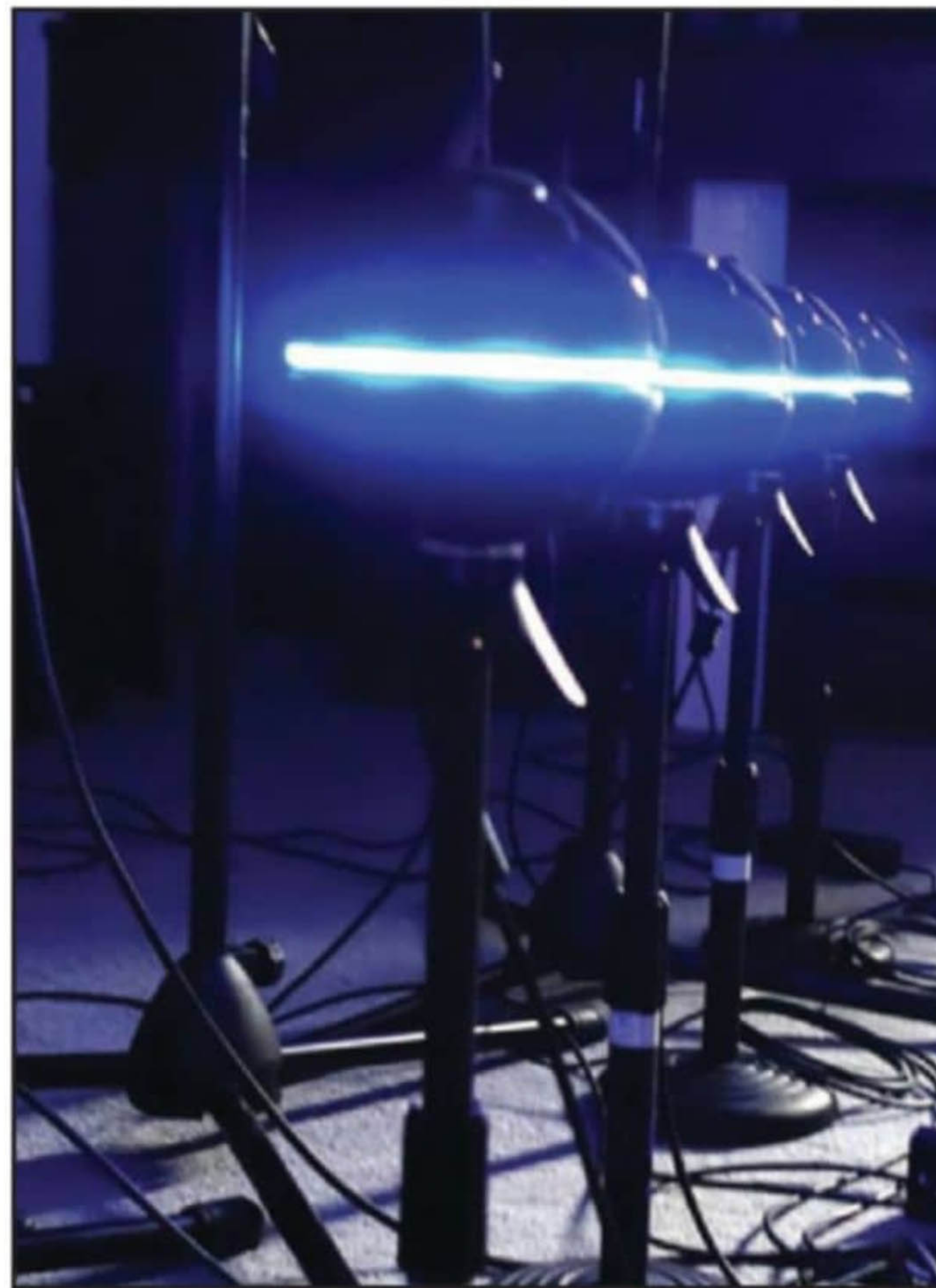


Photo 2: 6DoF audio environments deliver on the promise of lifelike, or live, immersive experiences with complete freedom of movement.

resolution of third-order Ambisonics audio ensures there are no gaps in sound and helps creatives and engineers bring a natural character to content.

### Effective 6DoF Workflow for Live Recordings

Implementing a simulated immersive audiovisual scene that supports 6DoF in 3D video games has become fairly common. The same cannot be said for cinematic audio-video or live recordings. Of the two current approaches to 6DoF production, only one opens the door to live recorded navigable 6DoF sound from an actual physical space (see **Photo 2**).

The first of today's techniques for 6DoF content is synthetic object-based 6DoF, which is much like the approach used to build audio in post editing for video games. Each sound source recorded is assigned to a channel, and then rendering engines and specialized DSP algorithms are used to mix thousands of tracks to try to mimic a real environment. Mono or stereo sound is artificially spatialized by software processing that attempts to replicate reverberation, diffusion, and other acoustic behaviors. This is a complicated and time-consuming process that can take months of work in postproduction.

The second approach to creating 6DoF audio environments is much simpler, and it delivers on the promise of lifelike, or live, immersive experiences with complete freedom of movement. Like the aforementioned 3DoF workflow, it leverages a recording solution equipped with a microphone array to capture the entire sound scene in a real environment, as well as software capable of delivering third-order Ambisonics signals.

However, while 3DoF production uses one Ambisonics microphone to recreate the spatial audio surrounding a particular recording device, the 6DoF approach uses many such microphones, distributed throughout the space, to deliver the extensive audio data needed to enable freedom of movement throughout the space. Software can process data in real time and calculate the sound experience for any given point within that space.

Because this model enables 6DoF audio production using sound captured in a natural environment, it features the actual acoustics within the recording venue. As a result, this approach also reduces the need for added processing and specialization in creating believable true-to-life sound. The higher quality of captured sound ensures a smooth listening experience as a person moves around the larger soundscape and gives them the natural feeling of actually being there.

### Practical Applications and Benefits

Of course, the feeling of full immersion in a virtual world can only be achieved if realistic sound follows high-quality images. 6DoF has been the subject of a great deal of work in the video realm, with leading use cases being in sports. In this area, 6DoF technology is enabling the creation of replays using a simulated camera—not entirely unlike the virtual microphone in the use of extensive interpolation from data acquired by high-resolution recording devices.

6DoF audio and video are both progressing, though at different rates. Eventually, they will meet at a point in their evolution where they together enable the fully immersive VR experience. In the meantime, demonstrations of 6DoF audio within an artificial visual environment simulated by a Unity game engine are offering glimpses at the future of immersive content.

For artists, the possibilities extended by 6DoF are incredible. The drop in revenues from traditional music sales makes it more important than ever to find ways to connect with audiences. Live shows are a tried-and-true way to engage with fans and to generate revenues. With 6DoF technology, musicians gain much greater flexibility in addressing fans' desires for unique and up-close experiences of the live shows. Musicians also can offer much broader access to their performances.

Fans anywhere can experience the show, with the ability to move throughout the soundscape and to listen from different perspectives. They enjoy control over the experience, moving at will toward the stage or further into the crowd. While it is exciting to get closer to the performer, it's also exciting to be able to watch and hear the show from any vantage point. That exceptional quality of 6DoF sound with today's sophisticated recording and processing solutions also benefits musicians and their fans.

With respect to production, an efficient 6DoF workflow for live recordings can significantly reduce the time, talent, and costs associated with building a 3D audio space for film or TV. In a conventional scenario, experienced sound engineers

would be hired to record different actors throughout a scene. Using multiple Ambisonics-capable recording systems with multiple microphone arrays, however, it becomes possible simply to place devices across the 3D space, trigger recording, and then use one or more virtual microphones in postproduction to isolate and manipulate sounds and speakers as desired.

### Future Work and Development

To support ongoing progress and, eventually, widespread adoption of immersive media, MPEG has been carrying out explorations on technologies that enable 6DoF. Zylia was the first company to offer practical demonstrations of 6DoF audio enabling reproduction of the listening point from any given location in the scene, and the company's recordings have been adopted by MPEG as reference material for its standards work. Current work includes describing the stream that will carry the elements of 6DoF content.

As standards development progresses, scientists, researchers, and artists are exploring real-world use cases for 6DoF—and they have the tools to experiment, thanks in part to Zylia's 6DoF VR/AR Development Kit. It is extremely challenging to record high-quality sound from many sources present in the sound scene at the same time. The new development makes it easy for users to experiment independently using a multiple-microphone array system in various spatial arrangements around the recorded sound scene. Users can experience not only the rotation of the sound scene (360° audio), but also freely change their position with respect to the sound objects.

The technology for 6DoF audio is not something buried deep in a research laboratory. It's a solution that's already on the market, with top companies and universities investing in exploring how 6DoF content can drive their future applications. Ultimately, the key to success will lie in being able to create immersive live recorded content using 6DoF and bring that experience into the home. The advance of 6DoF audio using multiple Ambisonics microphone systems is laying the foundation for this exciting future. 

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### About the Authors

Piotr Szczechowiak is the Co-founder and COO of Zylia. He manages the day-to-day operations. Piotr has a PhD in embedded systems and makes sure that ZYLIA hardware and software fits together.

Tomasz Zernicki is the Co-founder and CTO. He manages the technical operations. Tomasz received a PhD in digital audio processing and is an audio expert in the ISO/IEC MPEG standardization group.



# Updating the Rules on Amplifier Metrics

By  
**Jens Tybo Jensen**  
(Head of Class D Audio Marketing and  
Applications at Infineon)

Are your engineers using “hammers on screws” audio design methods to support outdated marketing claims? If so, it may be time to rethink your Class D audio design and measurement practices.

When I grew up, I was fascinated by fast cars. The faster, the better. In kindergarten, when I was playing Supercar Trump Cards with my friends and talking about which cars we’d drive when we grew up, like most boys that age, we focused heavily on metrics like “top speed” and “break horsepowers.” Peering into the window of a parked car that looked fast and fancy, the focus of my attention was on the speedometer dial. The higher the numbers on it, the greater my excitement would be. Not until later, I discovered that looking at the speedometer dial might not be the best way to gauge the qualities of a car.

As I grew older, I started to appreciate other metrics such as acceleration, torque, and gas mileage. Later in life, when I eventually had a driver’s license and started buying my own cars, I even realized some of the aspects that were actually most important to me like how well the car handles when you drive it, are fairly poorly judged by these metrics. Analogous to this, my view of audio metrics and performance claims has also changed over the years.

## The FTC “Amplifier Rule”

Following some pretty outrageous power claims made by Hi-Fi amplifier manufacturers back in the early 1970s, in an attempt to level the playing field prevent misleading consumers, the Federal Trade Commission (FTC) instated its “Amplifier Rule” in

1974. The Amplifier Rule says: for advertising and specifications of amplifiers sold in the US, continuous power measurements should be performed “using sine wave signals and at the impedance for which the amplifier is primarily designed, measured with all associated channels fully driven to 1/3 of rated per channel power.”

Rated power is usually determined by measuring the output voltage (RMS), with a sine wave signal, at the onset of clipping with a predetermined percentage of total harmonic distortion (THD), typically at 1% or even at 10% (at which point the output actually looks more like a square wave than a sinus curve) into a specified load resistance of 8  $\Omega$  or 4  $\Omega$ , depending upon amplifier ratings.

As such, although the test specification was perhaps not ideal, the “amplifier rule” served a purpose by somewhat leveling the playing field and enabled consumers to make better comparative product comparisons and purchasing decisions. In 1998, the amplifier rule was amended to also cover self-powered speakers (e.g., those now commonly used in home audio and portable applications).

However, as new audio technologies have emerged, and as consumers have since become more discerning, today they are more concerned with the actual listening and user experience of the audio product than how many watts of output power is claimed on the retail box.

Consumers still want audio products that sound great and play loud—but as I aim to illustrate with this article, the “Amplifier Rule” and other commonly used amplifier metrics, do not always pay justice to the best interests of consumers and/or manufacturers.

At the turn of the century, the advent of Class D audio amplifiers, which are a lot more power efficient than previous amplifier classes—especially at high power output levels, meant that audio manufacturers could now boast huge output power numbers. For that reason, the equivalent of the proverbial “top speed” cliché started to get a lot of marketing attention.

However, as shown in **Figure 1**, although their efficiency is a lot higher than with previous amplifier classes, conventional Class D amplifiers are still not particularly efficient at lower output power levels. The issue with this, is that for real-life input signals such as speech or music, where the input signal will be at or near idle most of the time (i.e., where the average output power is much lower than the peaks), it is the efficiency near idle that becomes dominant and determines the overall power efficiency of the amplifier.

Instead of converting all of the available energy into currents that can move speaker coils and produce sound, this means that most of the energy in this important output range is dissipated as heat in the amplifier and its surroundings. To make a meaningful assessment of an amplifier’s power efficiency, it needs to be benchmarked in a typical

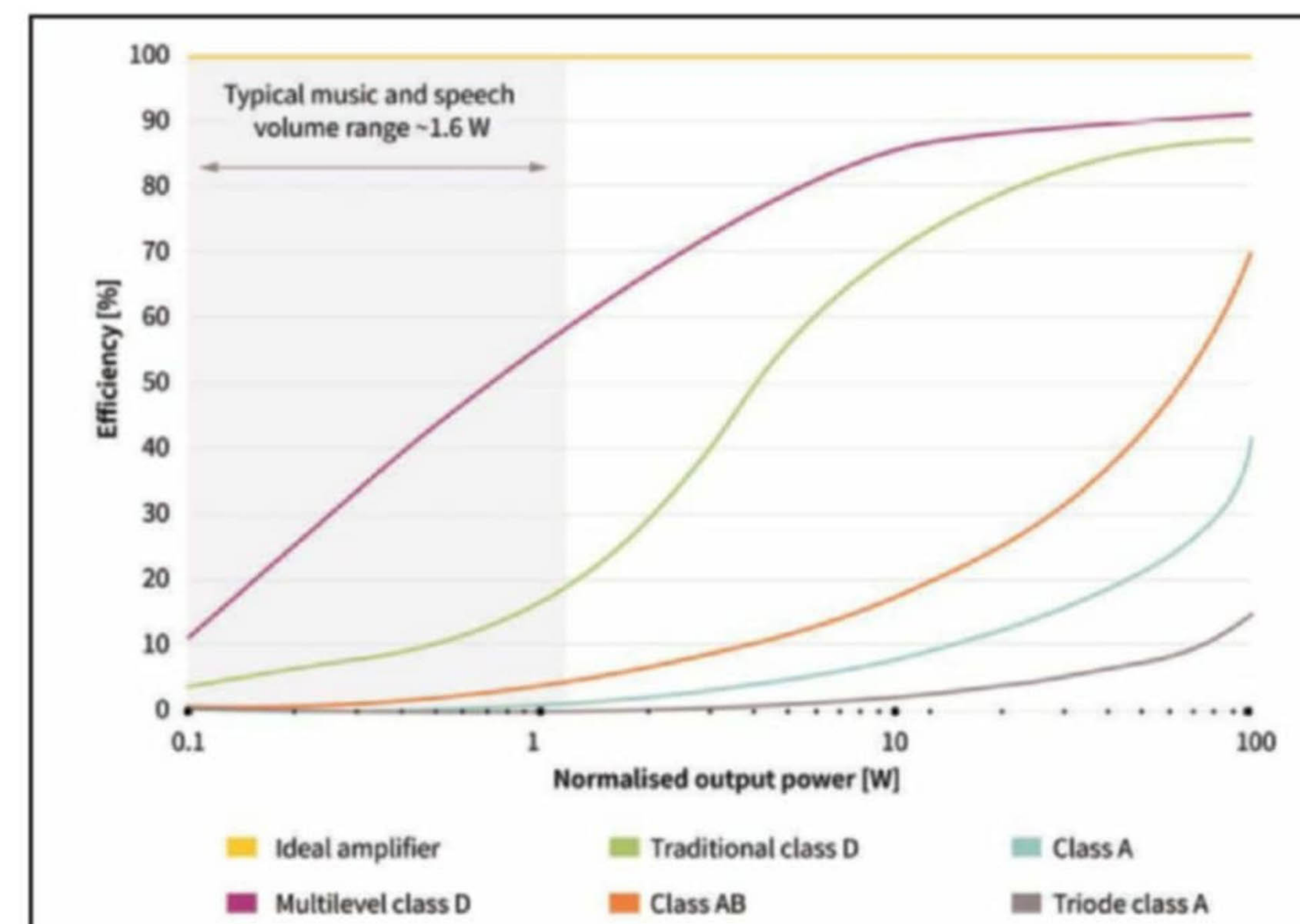


Figure 1: Conventional Class D amplifiers are still not particularly efficient at lower output power levels.

music and speech volume scenario—just like the gas mileage of a car should be measured under normal driving conditions and not at near maximum speed on an uncontrolled roll-stand.

With recent technological improvements such as multilevel Class D amplifiers, which are designed specifically for superior audio quality and power efficient amplification of music signals, failing to consider this means that the audio designers could miss out on opportunities to create better products for their customers.

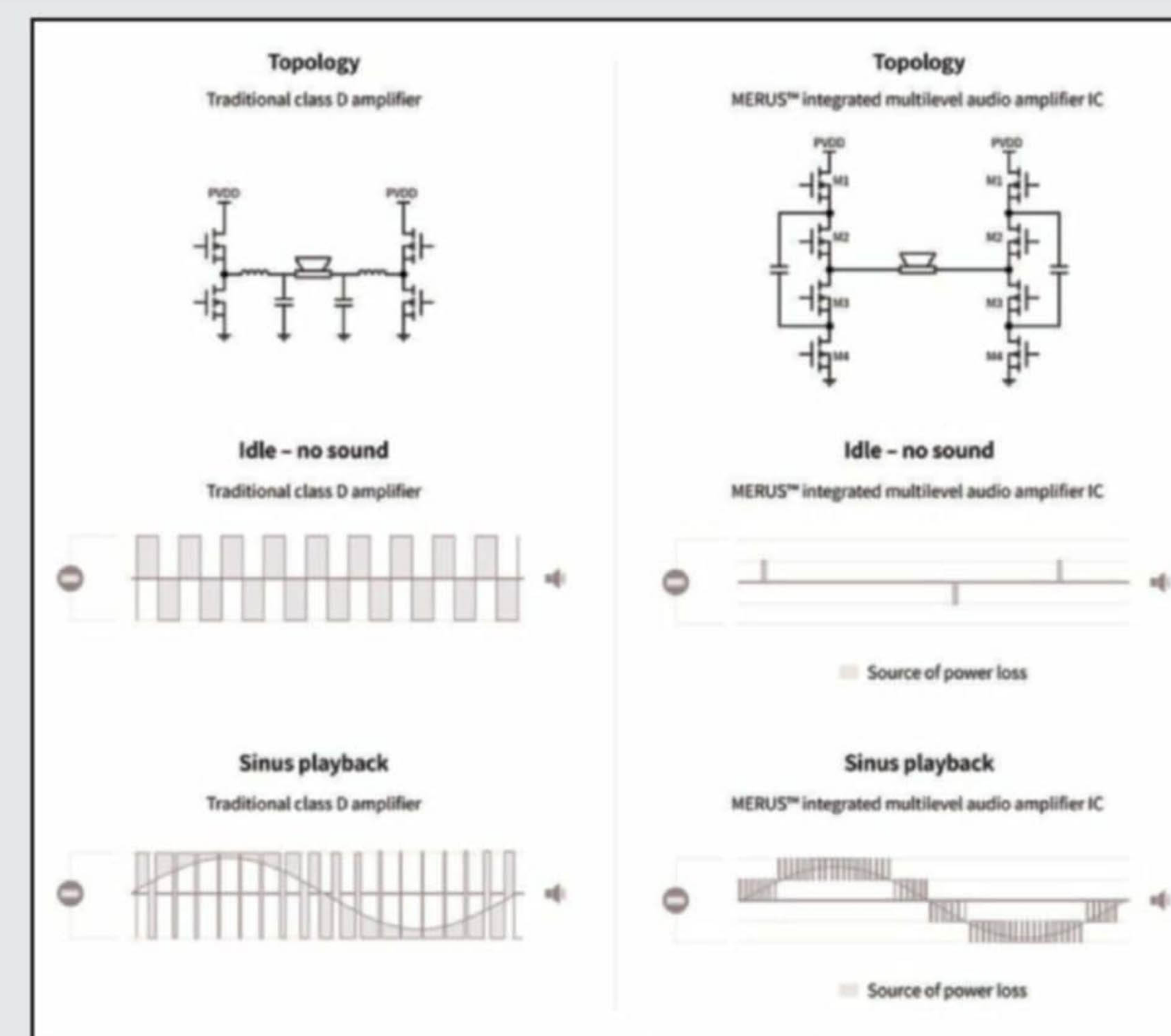
If power efficiency is only measured at rated output power with a 1 kHz sinusoidal input source (which is actually more suitable for stress testing than as a design target) and if we pretend that speaker

## What is a Multilevel Class D Amplifier?

In a conventional Class D amplifier, there is a lot of switching activity even in idle and near idle mode. Each half-bridge in the output stage changes the duty cycle to create the desired output level—but even if no output and no sound is coming out of the speakers, there is a lot of internal switching activity going on, which entails switching loss. You can see this in **Figure 1** as the shaded area in the curve.

In contrast, due to the proprietary and patented circuit architecture (shown in Figure 1), the multilevel amplifier 8 smaller MOSFETs can scale the output to a much higher granularity in terms of resulting switch frequency and output levels.

Essentially, this dramatically reduces power losses, provides much higher output signal granularity and yields many other additional benefits.





loads are purely resistive (which they are not), then important characteristics of these new amplifier types could be overlooked and this can result in sub-optimal designs—both in terms of audio performance and overall solution cost. Let me explain why.

## The Dynamics of Music

Consider a typical music input signal as shown **Figure 2**, as an example of a real-life audio signal. The peak value of the input signal is 1 V and the root mean square (RMS) value of the input signal is 0.125 V meaning that the Peak/RMS ratio is 8, corresponding to a crest factor of 18 dB. In contrast, a sinus input signal has a Peak to RMS ratio of  $\sqrt{2}$  corresponding to a 3 dB crest factor.

If we wish to amplify the music input signal with a Class D amplifier that has a rail voltage of 20 V in a 4  $\Omega$  nominal load (i.e., with a 5 A maximum peak current—meaning that it can deliver 100 W of instantaneous power) the resulting average value of the instantaneous output power waveform PAVG this amplifier will produce is only around 1.6 W.

To get to this number, we need to calculate the multiplication of the instantaneous voltage and current (which may not necessarily be in phase, if the speaker load is not purely resistive) averaged over time:

$$P_{AVG} = \frac{1}{T} \int_0^T v(t) \times i(t) dt$$

Also, it should be noted that the  $\sim 1.6$  W average output power is at the absolute highest volume setting (aka “delinquent teenager level”). For normal listening situations regardless of the music source which may have a slightly lower crest factor at decent harmonic distortion levels (i.e., unclipped signals) the resulting average output power is much lower.

Actual measurements of typical music listening levels with a reasonably efficient speaker, producing say 70 dB SPL (which is quite loud), the average

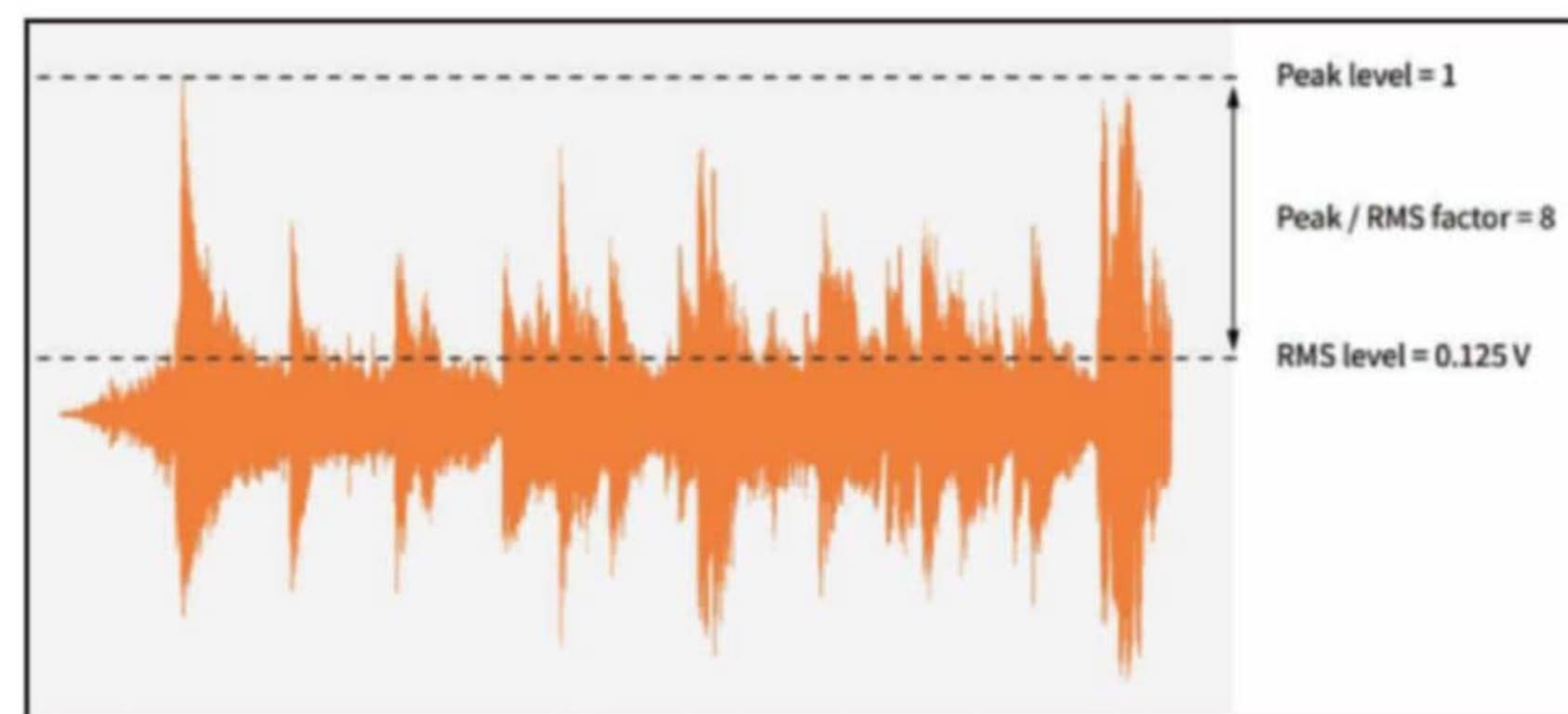


Figure 2: We used Norah Jones’ song “Come Away with Me” as an example of a real-life audio signal.

output power—even with a 100 W instantaneous power amplifier—is only around 0.25 W (i.e., 400 times smaller than the instantaneous power number).

If this is not considered and if power efficiency is measured with a sinusoidal input source and using purely resistive loads then the integral equation above can be reduced to:

$$P_{AVG} = \frac{V_{peak}^2}{2 \times R}$$

where R is the resistive load impedance.

However, by using the sinusoidal input as substitute for music signals or other higher crest factor input sources and by applying a purely resistive load instead of a speaker (assuming that currents and voltages are always in phase), we may falsely conclude that the average output power of the amplifier would be 50 W or half of the instantaneous power number—derived from the RMS of the 20 V peak sine wave voltage in a 4  $\Omega$  load:

$$P_{AVG} = \frac{V_{peak}^2}{2 \times R} = \frac{20^2}{2 \times 4} = 50W$$

This number is often referred to as “continuous RMS power” or “watts RMS,” which are also references in regulations presented by the Consumer Electronics Association (CEA—currently the Consumer Technology Association) in examples of how to mark electronic products. However, the term “RMS power” is in itself slightly misleading, as this is not actually the RMS value of the power output curve we are calculating (this would be a larger, but a somewhat meaningless number).

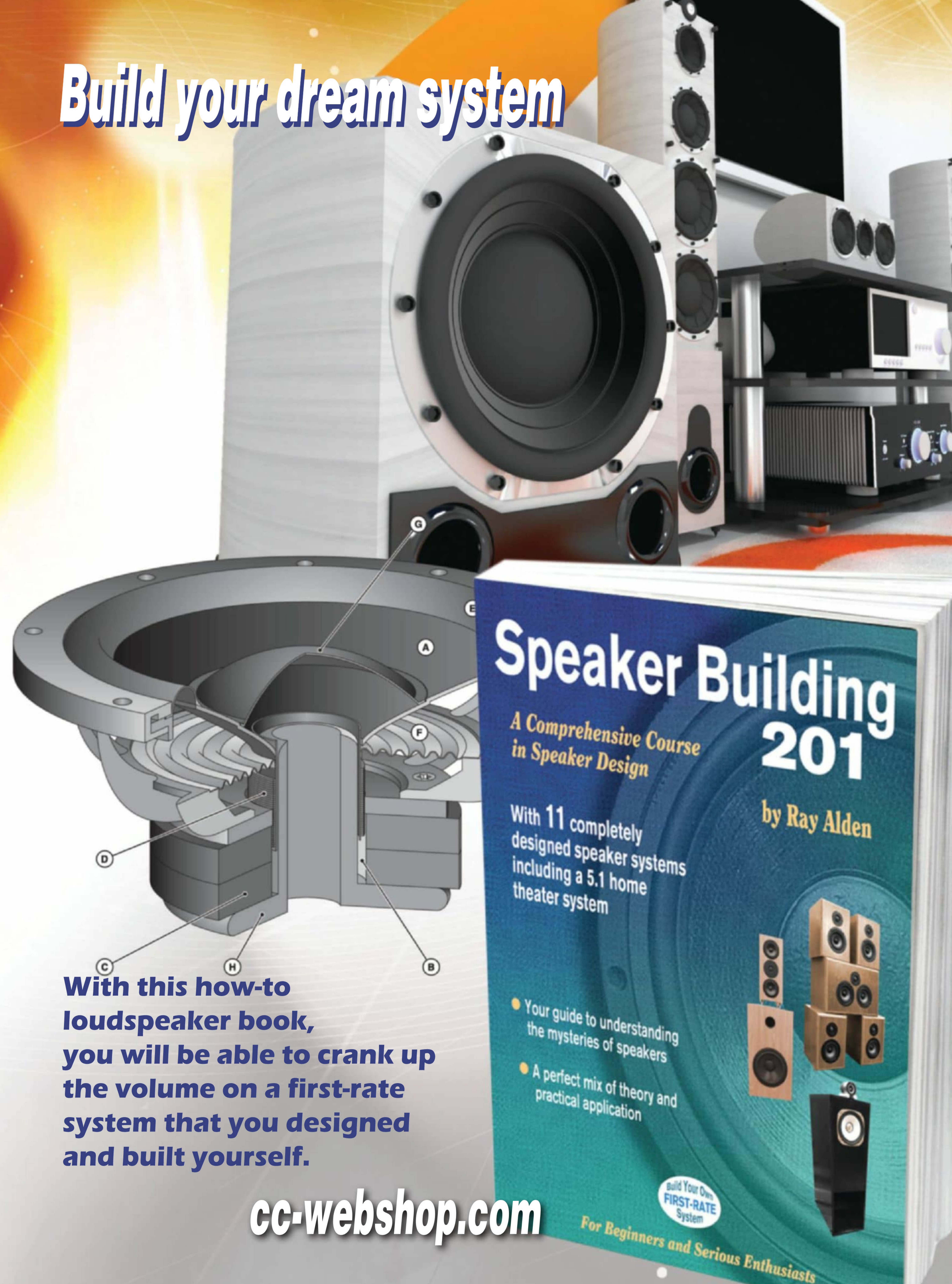
Continuous as opposed to “instantaneous” implies that the device can function continuously at this power level for long periods. Hence, it might be inferred that the amplifier in our example will have to produce a continuous output of 50 W in a resistive load from a sinusoidal input source at 1 kHz for extended time spans.

In contrast, the typical use-case average of playing real music is below 1.6 W so with the simplification the average output power is overestimated by a factor of 31.25 times!

Moreover, at the 1.6 W average output power level, the efficiency of a conventional integrated Class D amplifier IC is only around 20% meaning that 80% of the energy will go to waste as heat in the amplifier rather than used to move the speaker coil and produce sound. In contrast, the efficiency of a multilevel amplifier (e.g., the Infineon MERUS MA12070) at this output level is well above 50%.

Since an increasing number of audio applications are portable and battery operated, the difference in efficiency during operation is becoming increasingly

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## 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 subsequently acquired by Silvaco, Inc. a leading provider of EDA software tools and IP used for process and device development in analog/mixed-signal and power ICs. He also served as VP Sales & Marketing for Danish Hi-tech start-up, Exbit Technology A/S and 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.

significant. As an example of what is at stake, multilevel amplifiers can often double the effective battery playback time (or use half the battery capacity). However, if power efficiency is never measured with a realistic measurement setup (i.e., with music input or other high crest factor signals and real speaker loads or at least inductive loads to mimic a speaker impedance), this may not be apparent.

In addition, even if the targeted amplifier is reasonably power efficient at higher output power levels, the resulting heat from exaggerated continuous RMS power ratings must be removed at the same rate it is generated, without temperature building up to the point of amplifier shutdown. Consequently, this may lead to over design of the thermal relief in terms of more PCB layers than necessary or even the addition of bulky, expensive and unnecessary heatsinks.

Alternatively, if this is considered too expensive, it may be tempting to lower the amplifier rail voltage below what is ideal to lower the power rating and thereby solve the thermal issues. For instance, a rail voltage of 12 V (yielding a 36 W instantaneous peak capability in 4 Ohms) might be used instead of the original design intent of 20 V (which would yield a 100 W peak capability).

However, lowering the instantaneous output power capability of an amplifier can have detrimental

effects on the subjective listening experience of the audio system. Ensuring lots of headroom and good dynamic range by enabling a high enough rail voltage is perhaps the single most important sound quality aspect associated with the use of Class D amplifiers. The additional energy reserve can be “heard” and subjectively experienced as deeper and fuller sound even by non-audiophiles, much the same way that a car feels and drives differently, depending on its engine also below maximum speed.

Multilevel amplifiers are essentially efficient high performance “audio engines,” which are specifically designed for music playback. They produce high undistorted SPL output and excellent sound quality with heatsink-free operation. As input-levels shift up and down their power efficiency stays high and at the same time they have enough energy in store for those high output peaks that constitute real audio.

As such, the difference they make becomes apparent if the amplifiers are tested with real music signals and real speaker loads. On the other hand, if they are tested exclusively with a sinusoidal input source in a resistive load, as suggested by the FTC, then the advantages of the new technology may not be utilized and the analogy of using “hammers on screws” come to mind.

Sinusoidal signal sources are great for worst-case continuous output stress testing and other types of benchmarking—but to avoid overdesign the recommendation is to separate the design specification from the stress test requirement.

As a general rule of thumb, one-eighth of the maximum sine wave peak output power is sufficient for stress test purposes. In the example case, with a PVDD of 20 V, a nominal load impedance of 4  $\Omega$  and instantaneous peak output power of 100 W this would correspond to 6.25 W continuous power—which is quite feasible for a power efficient amplifier to handle—also without a heatsink. Infineon provides additional detailed recommendations on how to benchmark, measure, and stress test multilevel Class D amplifiers for real use cases.

Finally, in line with the car analogy: nothing beats a real test drive.

## Test It Yourself

If you want to find out what multilevel Class D amplification can do for your products, I would strongly recommend that you make a comparative trial under realistic usage situations using your ears as well as your Audio Precision measurement setup.

Ideally, the test should use the same music sources you expect your end customers to play. If the music is played to the same SPL level, I am convinced you will hear, see, and feel the difference. 

## Check the Power vs. Consumption Ratio

A good measure of how efficient a Class D amplifier performs in real-use cases is to look at the ratio between instantaneous peak power and idle power consumption. Generally speaking, the higher that number is, the better it is for the application. To learn more about the multilevel technology and its working principles, please read this article.

To check out Infineon’s audio offering and technical specifications, please visit [www.infineon.com/merus](http://www.infineon.com/merus).



 HAPPENS



# Building a Mini Speaker System

## The Bantams

As an avid speaker builder, Thomas Zarbo, Jr., decided to take on the challenge of building his own pair of small-size speakers. The result was a pair of full-range microspeakers that he dubbed the Bantams (see **Photo 1**).

Photo 1: A finished pair of Bantams—so named after the “small-but-powerful” division in the sport of boxing.

By  
**Thomas Zarbo, Jr.**  
(United States)

Ask anyone who has built speakers as a hobby for any length of time, and they’ll tell you it’s an endeavor that gets into your blood. My father jokingly (I think) calls it my “disease,” but probably the term “obsession” would be a better fit. At any rate, as with many speaker builders, our home is filled with various sizes and styles of speakers—from larger floor-standers, to more modest bookshelf speakers—often in various stages of completion.

However, few years back I realized that I had never designed my own pair of microspeakers. I suspect the oversight was possibly because it’s so difficult to fit everything inside a small enclosure, or maybe because small drivers traditionally don’t reproduce bass very well. Being a big guy with chubby hands who loves bass, I figure it was probably a combination of the two.

### Selecting the Woofer and Tweeter

These days however, getting nearly full-range sound out of very small speakers is no longer impossible thanks to advances in driver technology. For example, Parts Express sells a truly amazing driver—the Dayton Audio ND91-4—an aluminum-coned 3.5” driver that utilizes a neodymium magnet, and has an astounding amount of excursion for its size—over 4.5 mm one-way! Once I decided to take the plunge and design a full-range microspeaker, I modeled this driver using BassBox Pro in a small enclosure with a passive radiator and was truly impressed with what it would be able to do... realistic output into the mid-60 Hz range (more on that later), and distortion low enough to allow it to be listenable at fairly loud volumes. The ND91’s low distortion is due in part to the fact that it uses an

underhung voice coil, as well as a copper shorting ring to reduce distortion. I now had a great woofer to anchor my tiny speaker.

Most small speakers are logically mated to a budget-minded dome tweeter, but I wanted to pair this amazing little woofer with a tweeter that was a step up from the norm. I decided to use the Dayton AMTPOD-4 tweeter. Air motion transformer (AMT) tweeters use an accordion-shaped diaphragm that vibrates laterally to “squeeze” the sound waves forward. The AMTPOD-4 is packaged as an automotive-type tweeter, but since it is compact with a round body, I thought it would be a great way to implement an AMT tweeter in this design at a reasonable cost.

### Inside the Enclosure

The enclosure volume of this cabinet is 0.04 ft<sup>3</sup>, or around 1.13 ltr. For my original pair of Bantams I used a knock-down cabinet available from Parts Express. Although I used clamps because I have them, these precision-cut panels can easily be assembled in a half-hour with nothing more than some wood glue and masking tape (see **Photo 2**).

Since I wanted to apply veneer to the cabinets, and paint the baffle, I machined the driver openings on the baffles of the original Bantams before gluing them to the cabinets. **Photo 3** shows two additional pairs of Bantams built a year later that were glued up completely, having their various driver openings machined afterward.

Machining the openings in the baffles and cabinet takes a bit of precision. I mounted the tweeter in a 2” opening on the original Bantams, centered 1-5/8” down from the top of the baffle and centered left to right. I cut the tweeter hole with a forstner bit starting with a small 1/8” pilot hole and drilled the finished hole from both sides to eliminate tear-out. On subsequent pairs of Bantams I used a 1-7/8” forstner bit. The woofer opening is 3-1/8” and is centered 2-1/8” from the bottom of the baffle. The woofer opening and terminal relief were cut with a jig saw. I was careful with this procedure as there is not much of a flange on the woofer to cover the opening. I marked and drilled appropriate pilot holes for #6 screws to mount the woofer at this time.

The woofer is tiny, and even in 0.5” material there is not much breathing room through the driver frame. To remedy this, I used a drum sander to “open up” the areas where the driver “breathes” leaving the driver mounting hole areas untouched so the screws would have enough “meat” to bite into. I did this for the passive radiator as well, although it breathes a bit better with no magnet and more openings in its frame (see **Photo 4**).

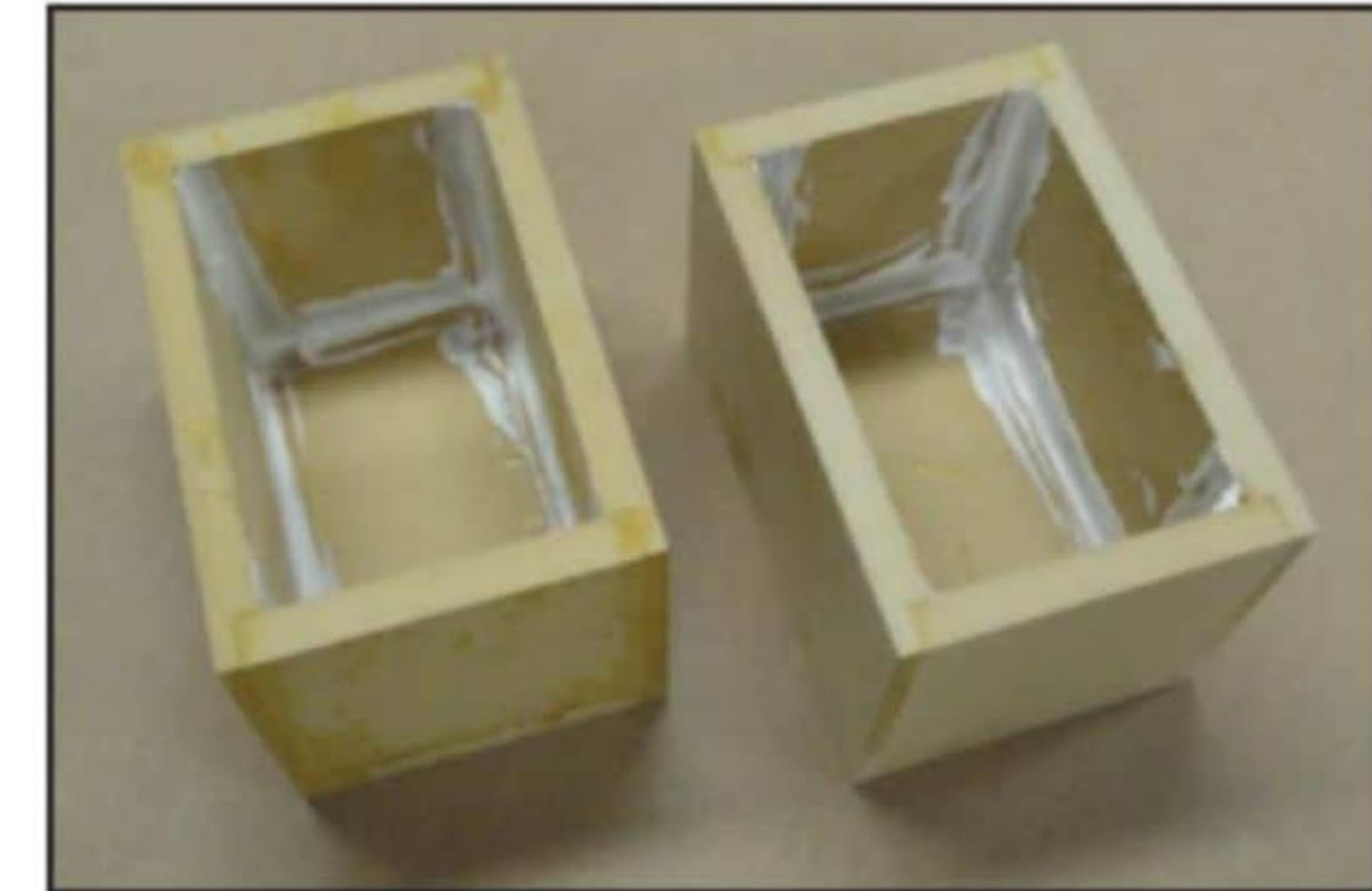


Photo 2: Assembled, and caulked 0.04 ft<sup>3</sup> cabinets are available from Parts Express. They are constructed of high-quality 0.5” MDF, and use dadoes to interlock the seams and provide maximum gluing surface.

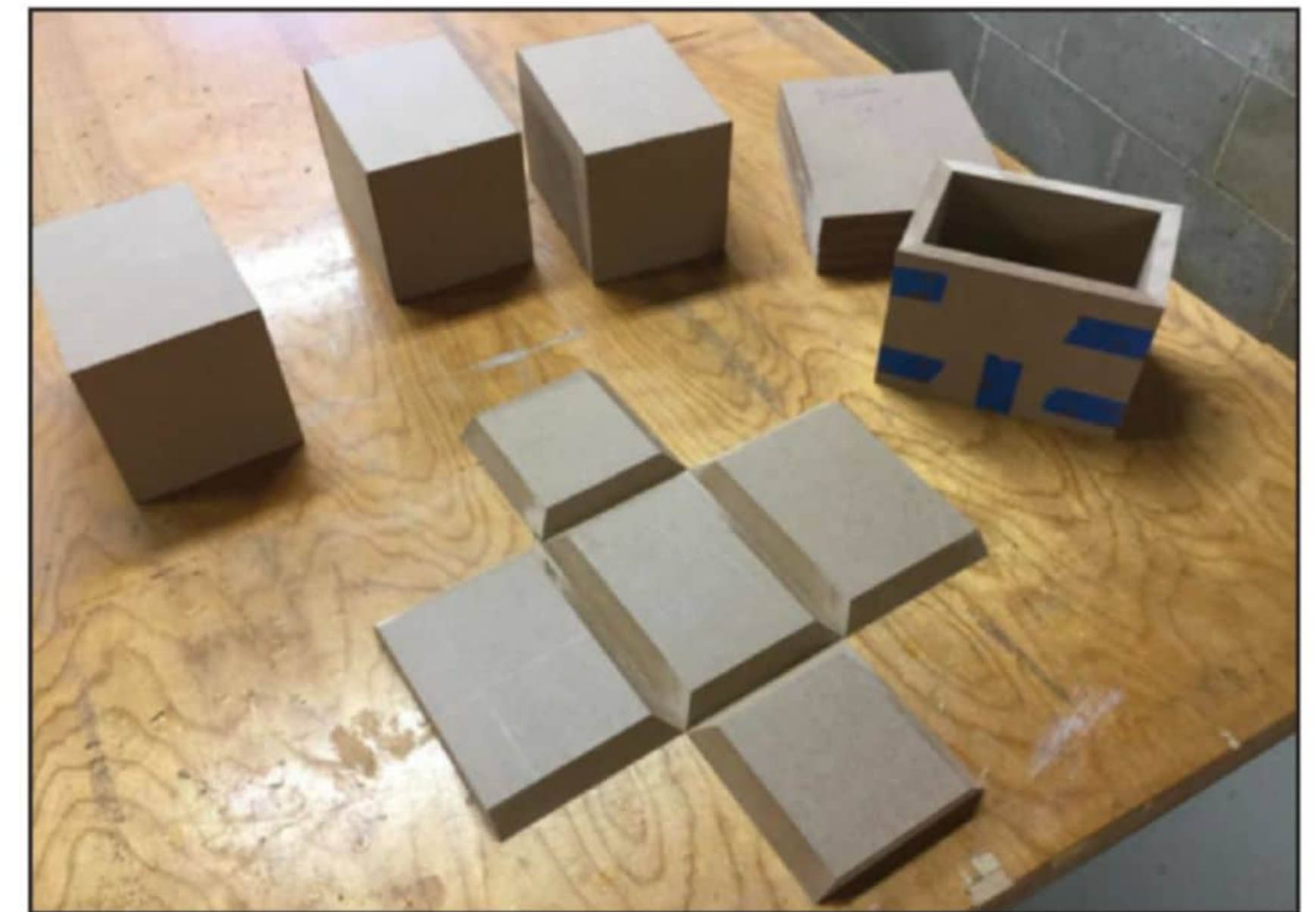


Photo 3: I made two additional pairs of Bantams constructed of 0.5” MDF with miter joints and bonded with wood glue. Accurate miter joints allow for panels to be “clamped” together with only blue painters tape.



Photo 4: Driver openings are machined. The woofer opening needs a notch to make clearance for the terminal. I created scallops on the rear of the baffle to allow for proper air flow around the rear of the driver.



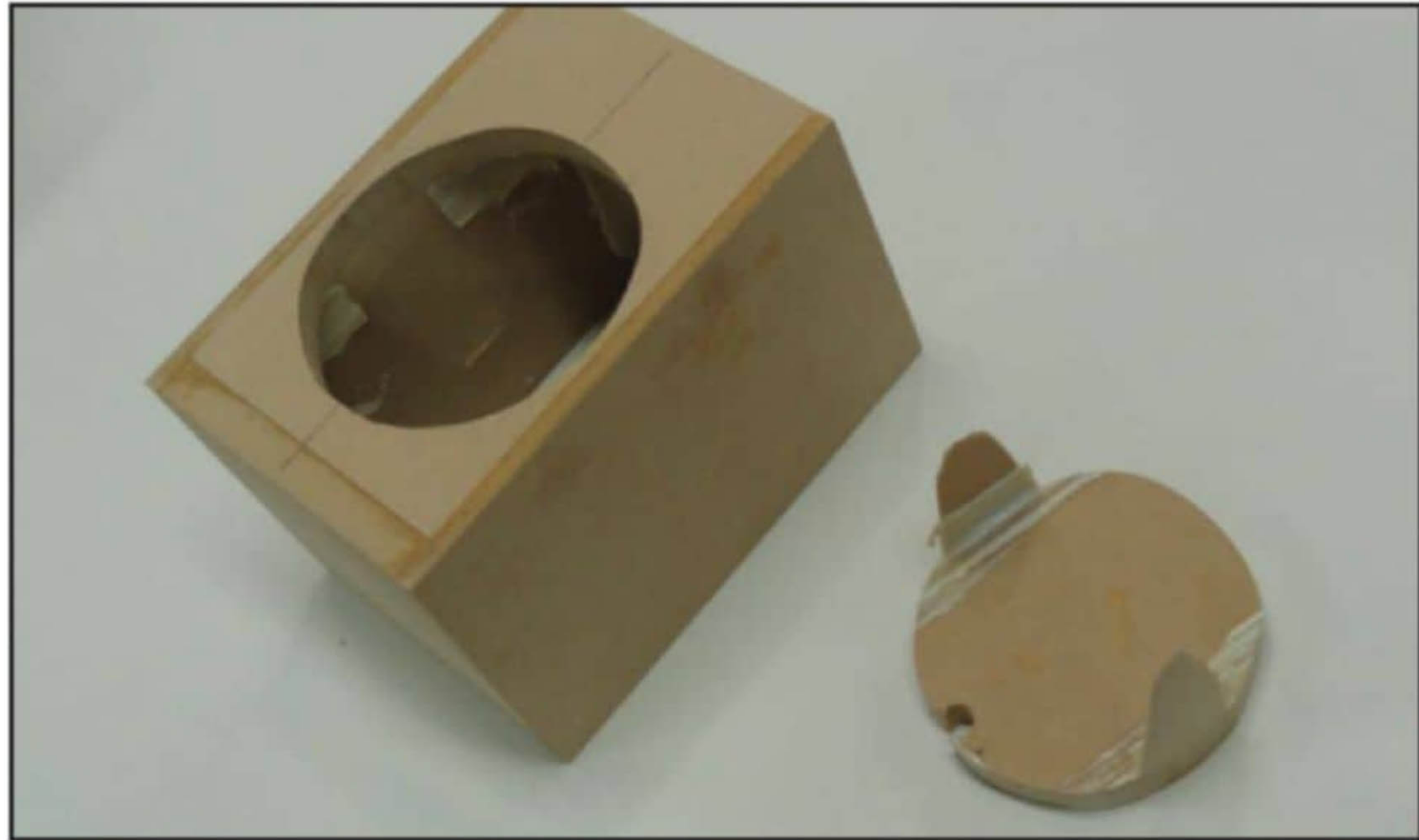


Photo 5: I cut out the opening for the passive radiator. Note the “wings” on either side of the resulting disc, which are actually thin slices of the cabinet walls.

Photo 6: I test fit the passive radiator and the binding post. It is important to pre-drill for the passive radiator mounting screws as it would be easy to split the 0.5” MDF otherwise.

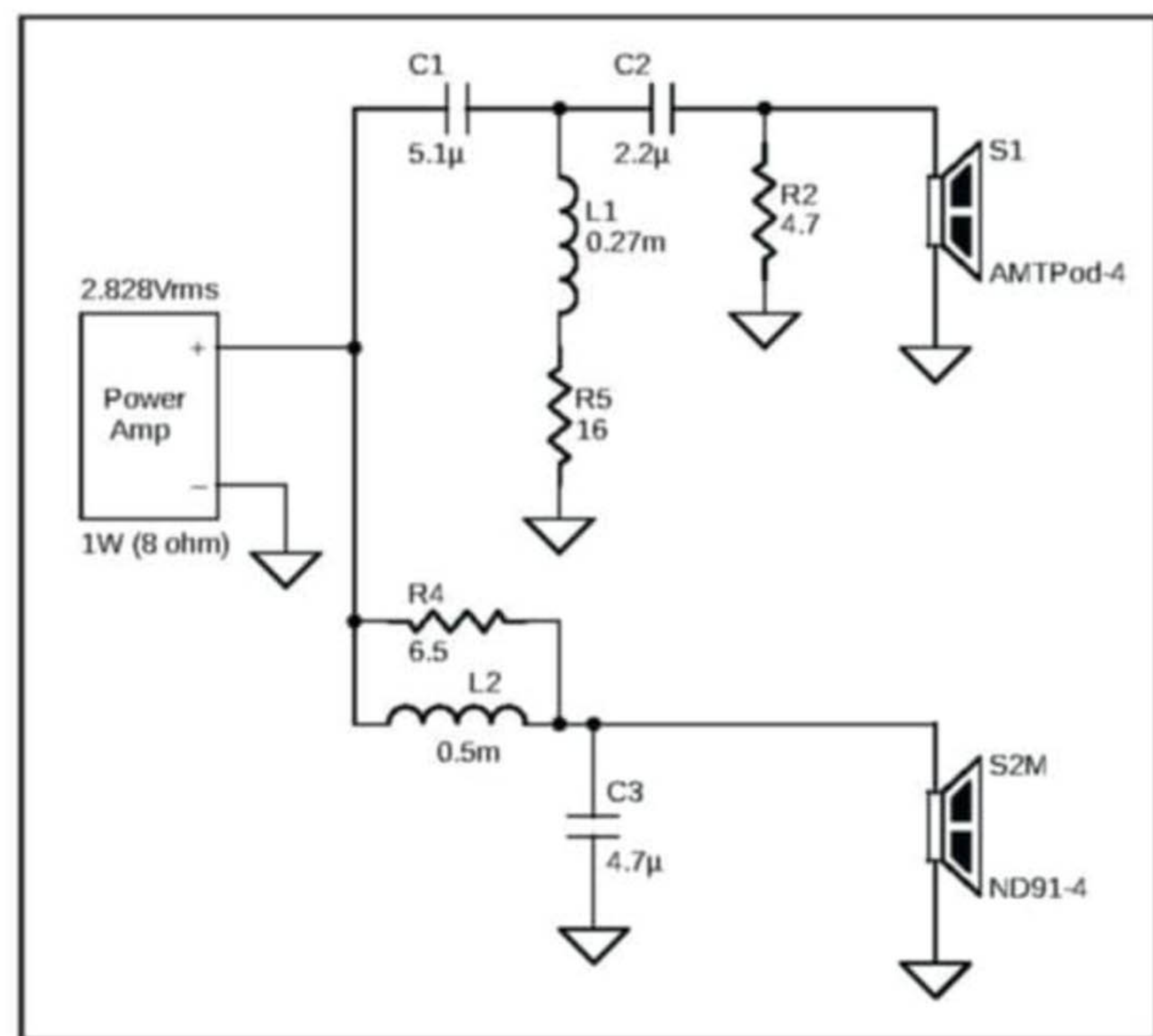


Figure 1: Bantams eight-element crossover schematic

The passive radiator opening is 3–3/4” and is on the rear of the cabinet, with its center located 2–15/16” down from the top, and centered left to right. I cut the passive radiator opening carefully with a jig saw, and as you can see in **Photo 5**, this is a snug enclosure for sure.

The holes for the binding post terminal are 5/8” from the bottom of the rear of the box and 3/4” apart. I chose this terminal because it is very compact and easy to install. I drilled these holes with a 3/8” drill bit, using the leverage of the tip of the drill bit against the inside bottom of the enclosure to ‘elongate’ the hole a bit. I had to bend the terminals closer to the binding post shaft to make everything fit. I marked and drilled appropriate pilot holes for the passive radiator and terminal at this time (see **Photo 6**).

### Testing the Design

As I previously mentioned, I used BassBox Pro for the cabinet simulations, and after receiving the woofers and breaking them in sufficiently, I used the Parts Express Dayton Audio Test System (DATS) to derive accurate Thiele-Small (T-S) parameters, as well as generate actual in-box impedance data. I also used Parts Express’ OmniMic measurement system to generate frequency-response measurements in-cabinet. Jeff Bagbys’ Passive Crossover Designer (PCD) and Response Blender programs were used for the crossover simulation, along with X-Sim to reaffirm, tweak and then print the schematics. After several days of modeling and tweaking until I felt my simulations were in the ballpark, I alligator-clipped a crossover together and used OmniMic to measure the frequency response and see what I was actually working with.

The end result, after some more tinkering, was a final crossover point in the neighborhood of 6,500 Hz, which was quite a bit higher than I anticipated going in, but there were actually a few good reasons to go with the higher crossover point.

First, there was a pronounced dip in the tweeters in-cabinet response centered around 6,900 Hz, and extending a good 1,000 Hz on both sides. That natural dip, combined with the electrical filter allowed me to use the low side of the tweeters natural response dip to my advantage as the beginning of the low-frequency roll-off. Trying to flatten out that dip to gain another few thousand Hertz of low-end extension wasn’t proving to be a good option and would have meant more crossover parts. Second, the woofers fairly smooth response out to nearly 8,000 Hz, and no real breakup to deal with allowed for that higher crossover point on the woofer side of the crossover as well.

The finished crossover consists of eight components employing a second-order crossover for the woofer with a response-shaping resistor in series with the inductor. The tweeter filter is third-order with two resistors for shaping and level-matching (see **Figure 1**).

Since the tweeter has no markings on the wire or driver itself to indicate what the polarity is, I took a guess and assumed that the copper colored wire (as opposed to the silver colored wire) was the positive (+) lead and wired it up under that assumption. I may be wrong on that assumption, but either way, I wired the copper colored line to the positive connection on the crossover as indicated in the schematic.

### Adding a Passive Radiator

As I noted earlier, it can be a challenge to fit all of the components needed to make a speaker function inside a tiny cabinet. Early on in the brainstorming stage, I toyed with the idea of possibly using the passive radiator as a crossover mounting point. I also realized that placing air core inductors next to steel such as the passive radiator frame would likely raise their inductance values. The big question was, how much?

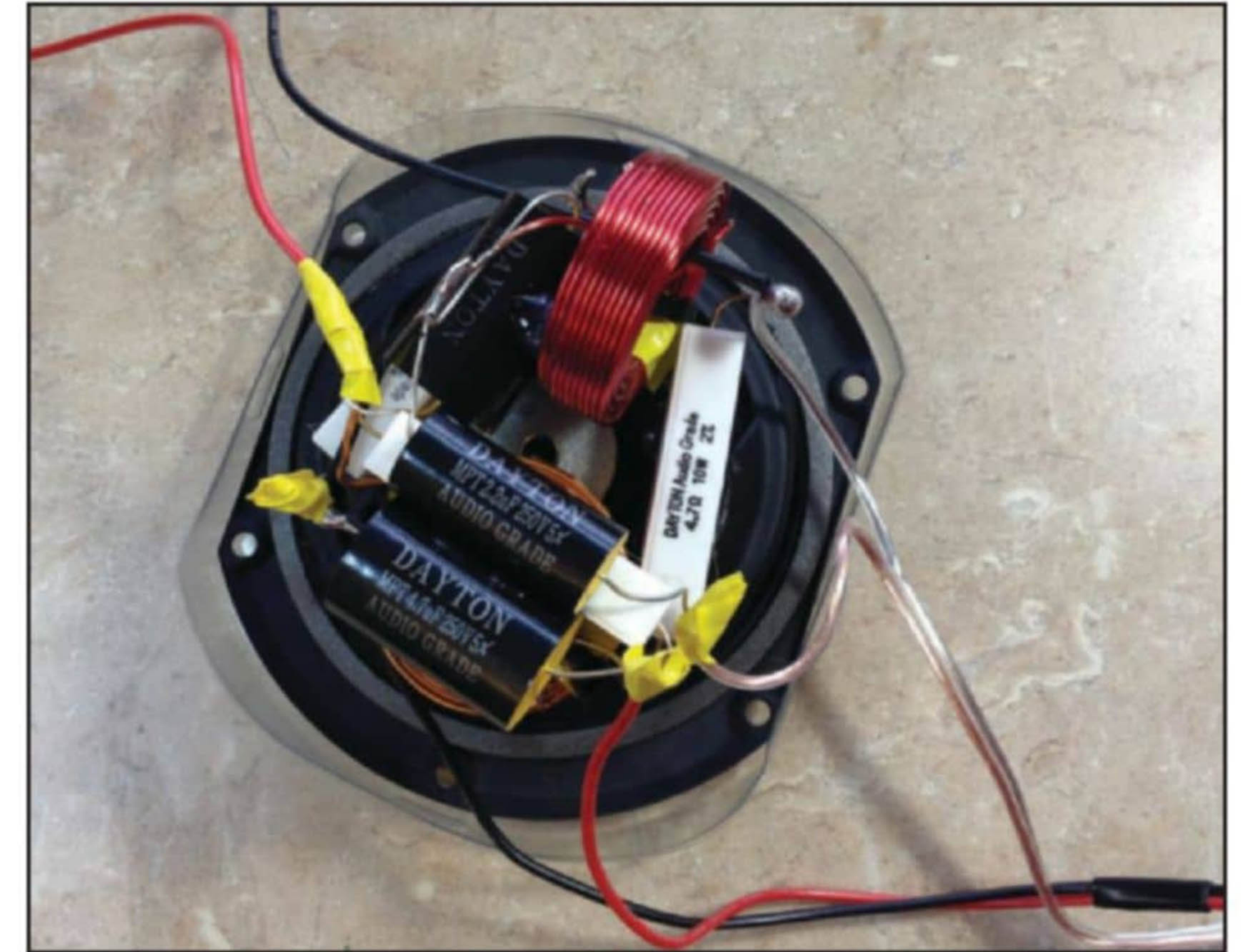
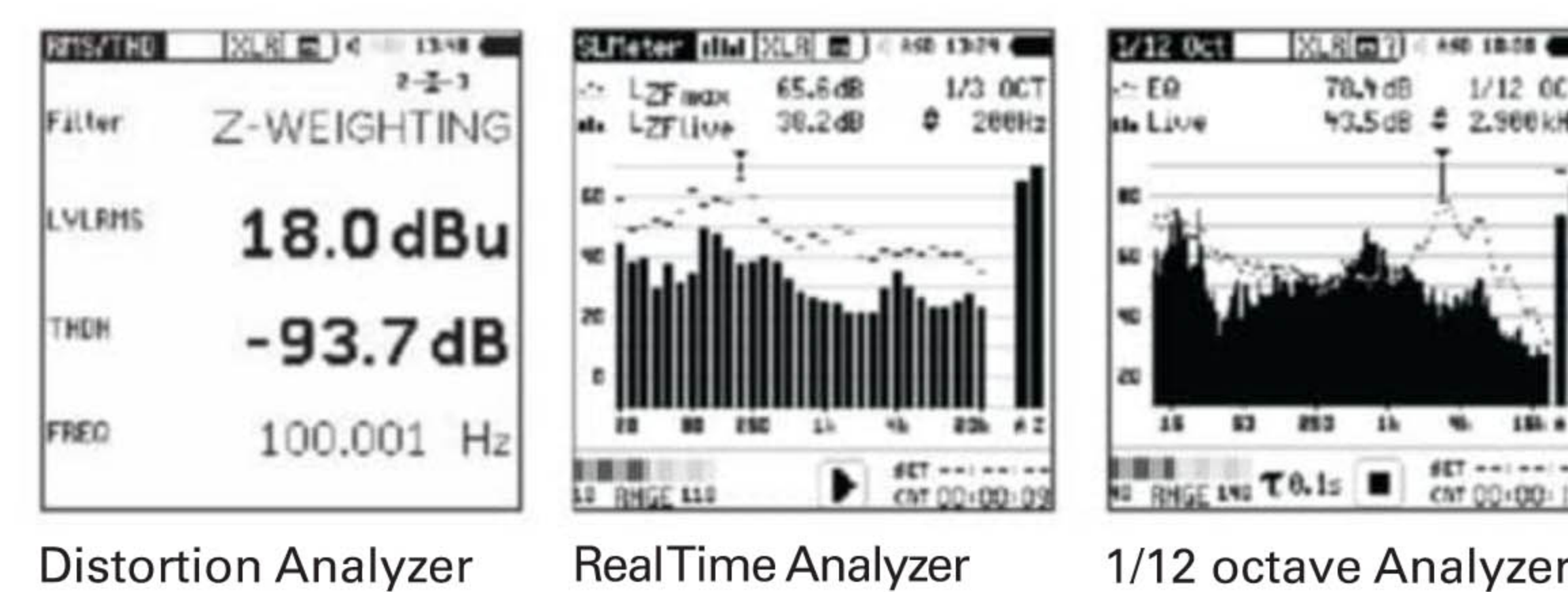


Photo 7: Finished eight-element crossover mounted on the rear of the passive radiator. Components were hot-glued to the passive radiator frame and test fitted after each step in the cabinet opening before zip-tying, and final soldering to be sure it fit through the opening.

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Photo 8: The baffle was painted with Rustoleum textured paint, then shot with rattle-can gloss black. Tweeter pressed in place on the finished baffle.



I tested the values of the inductors with the DATS while it was mounted on the passive radiator to see if the changes would be detrimental to the overall design. The larger 0.5 mH coil for the woofer measured 0.52 mH by itself, and 0.58 mH when mounted on the passive radiator frame. The smaller 27 mH coil measured 0.27 mH by itself, and 0.28 mH when mounted on the passive radiator. I felt the differences were minor enough to not be detrimental—and as I was to find later after final assembly and testing—no further adjustments needed to be made.

Feeling like I dodged a bullet at this point, I also tested the inductors mounted on the passive radiator with the magnet of the driver arranged as it would be in-cabinet—very close to the coils. To my surprise, there was almost no measurable difference in inductance on either coil factoring in that variable. I was now feeling quite relieved, because the thought of trying to squeeze in and mount a crossover board inside this tiny cabinet with my fat hands seemed like it would likely be an exercise in frustration. I then needed to figure out how to mount the crossover components efficiently onto the back of the small Peerless passive radiator. **Photo 7** shows my solution.

## Putting It All Together

I chose a light colored “figured maple” veneer for the sides, top, and bottom of the enclosure to provide nice contrast with the black baffle. I used the iron-on method of applying veneer as it is easy to do and yields great looking results. After the veneer was sanded smooth, I taped off the front where the baffle will glue onto. I used five or six coats of Rustoleum wipe-on polyurethane because it’s easy to apply, looks good, and doesn’t smell so strong that I can’t apply it in my basement.

I taped off the baffle and primed it with BIN shellac base primer, which does a nice job of sealing MDF, sanded smooth with fine sandpaper, and painted with Rustoleum textured paint. I used four long drywall screws driven slightly into the woofer screw holes to hold up the baffle as I painted it. The texture paint needs to be applied slowly in several coats, I used five very light coats; the first coat hardly covers up the white primer, and by the fourth or fifth I could barely see the light primer underneath. I changed the direction of spray for each coat for a more even texture. I finished up with gloss black paint by Krylon for a nice, glossy look that will contrast nicely with the light colored cabinet. Since the back of the box is mostly passive radiator, I opted to just roller paint it black with Rustoleum Painters Choice to make life easy.

I gave all finishes a few days to dry, then it was time for some assembly. I mounted the tweeter before attaching the baffle since it is a bit tricky to mount as it’s really designed for automotive stick-and-press usage. On the original Bantams, I adhered a full wrap around the inside of the tweeter opening with the Parts Express 0.5” speaker gasket foam tape. Then I twisted and pushed the tweeter in until the vents on the front were horizontal and the curved round-over on the tweeter was just sitting proud of the baffle (see **Photo 8**).

I thought that the foam gasket may have some bounce to it, so I filled the gap between the tweeter and baffle with construction adhesive on the rear of the baffle. This will firm up the tweeter and prevent it from vibrating. I also placed a bit of silicone where the wire exits the tweeter case as well, just in case the tweeter was not air-tight.

Although this mounting method worked fine, it was challenging to fit in the tweeter without disturbing the foam, so on subsequent Bantam speakers, I went with a much simpler mounting method. I drilled a 1-7/8” hole and just wrapped several thicknesses of black electrical tape around the tweeter body right up to where it starts to round over on the front. It took a little experimentation on a scrap board to get the thickness correct, but this

ended up being a much easier way to mount the tweeters, and the method I would recommend. I again used some silicone sealant to seal and secure the tweeter body to the baffle.

Next, I glued the baffle to the box with Gorilla brand construction adhesive (see **Photo 9**). I used this as opposed to polyurethane glue because it lays thick like caulking, but allowed for a good set up and close baffle-to-box alignment. Also, being thicker than wood glue, it negated the need to seal those seams with silicone later on, which in a cabinet this small would be extremely difficult.

Finally, I used a small 4” x 4” piece of UltraTouch denim insulation on the inside top of the cabinet. I ripped the piece of 2” thick insulation apart to yield two approximately 1” thick pieces, which I adhered to the inside of each cabinet with spray adhesive. My hope was that it would calm down any in-cabinet resonances a bit.

The finished enclosures are approximately 6.5” tall, 4.5” wide, and 5” deep. The passive radiator and its attached weight create serious enough vibrations to allow the speaker to “walk” around on most surfaces—a fact I discovered after leaving the speakers unattended for a few minutes while

playing at high volume. I returned to find them facing random directions within inches of falling off of the table they were sitting on. I quickly applied small rubber feet to the corners of the cabinet



Photo 9: Construction adhesive was applied to the front end-grain of the cabinet. I wiped off any squeeze-out with a damp rag immediately.

## About the Author

Thomas Zarbo, Jr., is a preschool teacher living in southern New Jersey, and he has been building speakers since he was a teenager. Thomas’ interest in speaker building began when he and his father took apart an old stereo console unit and transplanted the speaker drivers in some home-made enclosures. He said he was instantly hooked on the hobby. His love of DIY things “audio” has grown to also include various smaller amp and radio projects. Thomas said he is thankful for the kind folks on the “Tech Talk” forum who have helped him over the years gain a good understanding of how the various aspects of speaker design and construction fit together. Thomas added that he was grateful for their continued help and guidance, as well as to Parts Express for hosting the forum, and *audioXpress* for promoting DIY speaker building.

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Photo 10: The original Bantam pair are shown in the center with the full Striped Ebony veneer on the left and the Bubinga veneer on the right. The two newer pairs on the ends used a slightly larger round-over on the sides, and a truncated roundover on the top/bottom for a more sophisticated look.

bottom and reflected on how nice it felt to have learned a lesson the easy way for a change.

I have built three pairs of Bantams so far—the originals with the black baffles and two other pair for friends with Bubinga and Striped Ebony veneers with larger front round-overs that allowed for a full-wrap of a single piece of veneer (see **Photo 10**).

## The Sound

So how do they sound? I honestly admire people who can use colorful language to communicate how a particular speaker sounds. However I don't seem to possess that skill, so I'll simply say they sound very good. The Bantams don't do anything wrong sonically, and they project a clean and detailed soundstage. They also seem to image fairly well on certain material. They are currently my wife's favorite speakers for two years running now, and they really have two things going for them over most tiny speakers:

First, the ND91-4 woofer is an overachiever by all counts. It boggles my mind how a driver this small can reproduce music as cleanly, and loudly as it does. I use an 80 W per channel Onkyo amplifier with these, and they tolerate the "turn it up to 11" mentality just fine. This woofer just keeps on going and going.

Second, the AMT tweeter is kind of special—it's a sparkly driver with the ability to convey the finer nuances of music realistically—like the sliding of fingers on a guitar string or the gentle breath taken by a vocal soloist. This combination of drivers just seems to blend perfectly for a nice sonic treat.

The frequency response is roughly flat within  $\pm 3$  dB out to 16 kHz. Although the system's F3 (the point at which the system response is 3 dB lower than reference or where a person is likely to begin to notice a reduction in output) is technically in the mid-90 Hz range, that's really a little deceiving. The modeled low-end roll-off below that is very shallow (less than first-order) with an F6 of 59 Hz and an F10 of 45 Hz, at which point the "knee" in the response takes over and output falls off dramatically (see **Figure 2**).

The fact that the response drops like a rock after 45 Hz is actually a good thing. It's one of the benefits of using a passive radiator as opposed to a vent in this case—it protects the woofer from over-excursion in the lowest frequencies that it couldn't reproduce very well anyway.

The Bantams are a nice stand-alone mini-speaker that punches above its weight class, but like most micro-speakers, they really shine when paired with a subwoofer.

## Next Month

In the follow-up article, I detail the design of my Tenacious Bass-8 powered subwoofer with a

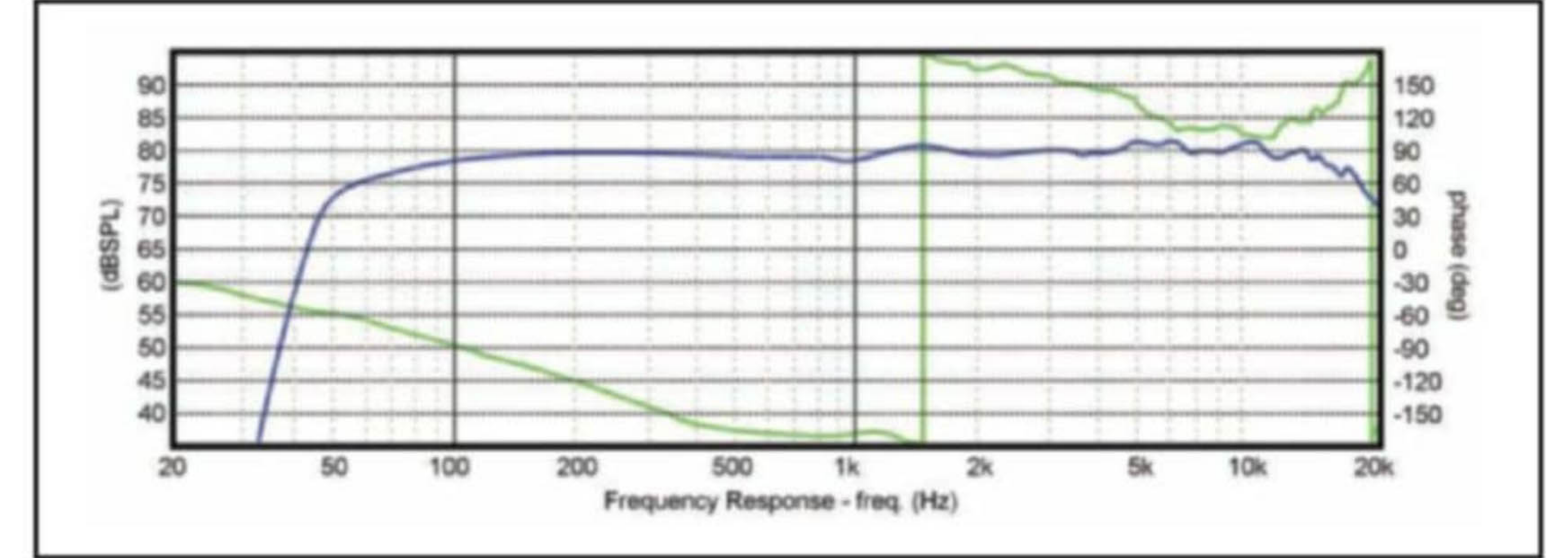


Figure 2: I measured the frequency response of the Bantams with a 5 ms gate time 20" from the tweeter axis (modeled frequency response included below 500 Hz).

front-mounted passive radiator, which I paired with my first truly tiny Bantams speakers to complete this 2.1 system.

For more information on the Bantams, visit the Parts Express website and click on "Project Gallery" at the bottom of the page. A search for Bantams will take you to the online write-up with a full clickable parts list. Additional information, along with step-by-step photo crossover assembly instructions, is also available at the Parts Express "TechTalk" forum if you perform a similar search.



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# The First Electronic Music Synthesizer

According to vocabulary.com, “A synthesizer is a musical instrument that has a keyboard like a piano but can sound like just about any instrument in the world.” In this article, Richard Honeycutt takes a step back in time to explore the first electronic music synthesizer.

By  
**Richard Honeycutt**  
(United States)

If asked to name the inventor of the music synthesizer, most people would probably mention Bob Moog, who first demonstrated his voltage-controlled modular synthesizer at an Audio Engineering Society (AES) convention in New York on October 12, 1964. But in fact, Laurens Hammond demonstrated the 150-tube Novachord synthesizer (see **Photo 1**) at the 1939 World’s Fair in Chicago, IL. While its cost and weight prevented it from becoming a financial success (only about 1,000 units were produced, of which about 200 remain today), the Novachord was certainly the first polyphonic electronic music synthesizer.

Well, almost. Laurens Hammond’s most famous creation—the Hammond organ (see **Photo 2**)—is also a polyphonic music synthesizer containing electronic modules, although its tone generator



Photo 2: Laurens Hammond at the keyboard

is electromechanical. This incredibly successful invention was specifically designed to sound as much as possible like a “true” organ.

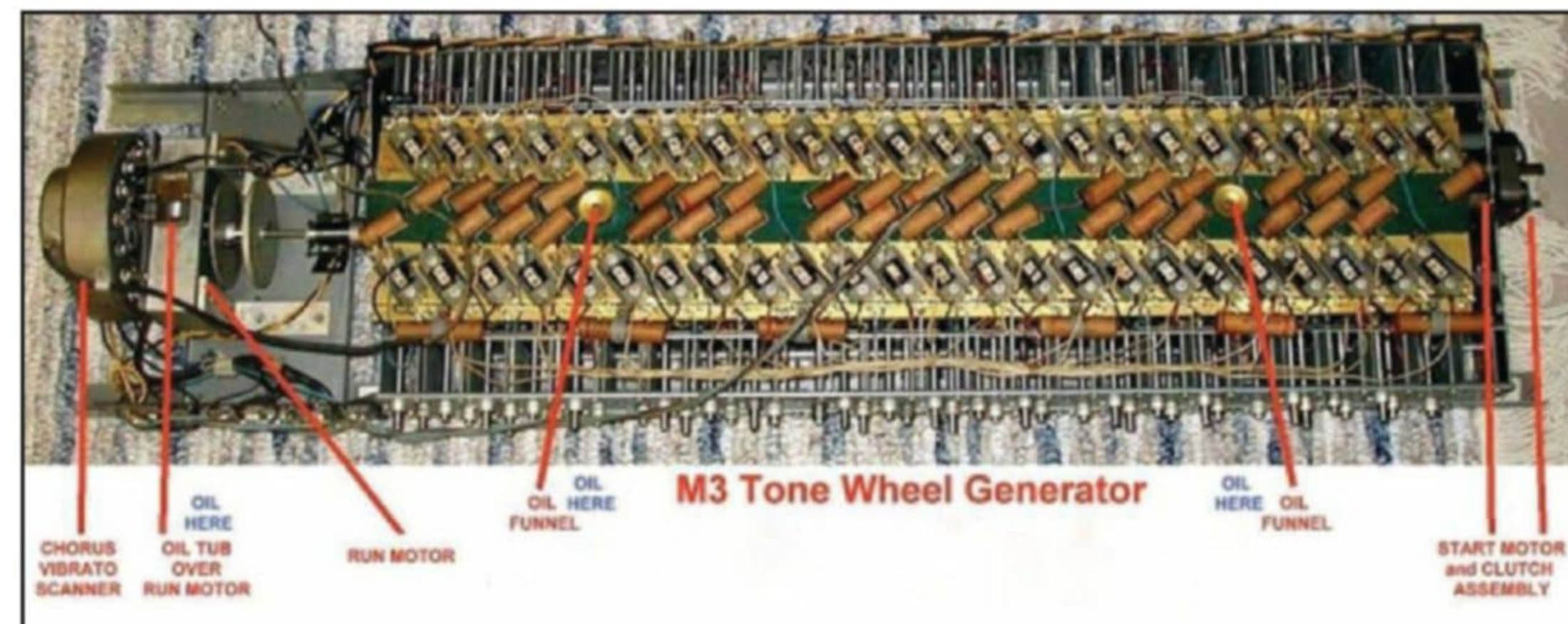
Although many professional organists disdain the Hammond and refuse to call it an organ, much of their criticism is unfair. First, a professional organist’s trained ear is accustomed to hearing pipe organs and their more recent electronic imitations in their natural habitat: an auditorium or church sanctuary. As any competent organ expert will quickly point out, the room is part of the organ. Specifically, the reverberation signature of the room is an essential part of a concert organ’s sound. If it were possible to hear a fine pipe organ played in an anechoic chamber, it would sound like a calliope. In contrast, Hammond organs are very seldom heard in such acoustical environments; rather, we hear them in living rooms, clubs, bars, and coliseums (through powerful amplification systems).

Second, fine pipe organs are usually played by highly trained professionals, whereas players of Hammond organs cover a wide variety of skill and taste levels. Certainly there are virtuoso Hammond players, but one dose of what may be called “funeral-parlor Hammond sound” with its excessive spring reverberation and mechanical vibrato, plus inept drawbar settings, can certainly sour a critical listener on the instrument!

## The Hammond’s Inner Workings

Laurens Hammond manufactured electric clocks, whose synchronous motors were synchronized with the highly stable 60-Hz AC electric power grid’s

Photo 1: A 1939 Hammond Novachord synthesizer (Image courtesy of Antiquity Music)



frequency, and so they kept excellent time. He created a tone generator driven by a synchronous motor and using notched wheels of ferromagnetic materials that were spinning near pickup coils in the field provided by permanent magnets to create sinusoidal tones.

The Hammond tone generator (see **Photo 3**) used in the “console” organs (the B3, the C3, the A-100, etc.) creates 91 pitches spanning the frequency range of about 32 Hz to 5,925 Hz (C1 through F#8). Each key activates nine contacts, corresponding to a fundamental and eight overtones of the note. The relative levels of the fundamentals and overtones of all keys that are depressed at a given time are controlled by drawbars: one drawbar for each harmonic. (Remember that the fundamental is the same as the first harmonic.) Thus, the drawbars function much like the stops on a pipe organ, and each Hammond organ also has a way to pre-set various drawbar combinations, as does a pipe organ.

So, in fact, a pipe organ is a sort of synthesizer, although not an electronic one. This fact explains why various pipe organ stops have the names of other musical instruments, such as flute, bassoon, cornet, and some obsolete instruments such as the krummhorn. But I digress.

Some Hammond organs incorporate a pedal solo generator. This feature was added because some organ literature includes solo passages played on the pedals. The normal 16’ and 8’ pedal sounds from the tonewheel generator blend well with various keyboard settings (drawbar combinations) for most music, but are not sufficiently bold for solo use.

The pedal solo generator’s master oscillator creates a fundamental pitch corresponding to which pedal is pressed, then divides it via a five-stage all-tube frequency divider. The resulting pitches can be added to the fundamental according to the player’s preference, by use of rocker switches and a potentiometer. Thus, bold pedal voices such as the attention-grabbing bombarde can be produced by a Hammond organ. To listen to a Hammond organ, visit [www.youtube.com/watch?v=q7M6\\_k05S4k](https://www.youtube.com/watch?v=q7M6_k05S4k). The demo begins at the 4:47 mark.

Some Hammond Organ models incorporate “percussion,” which adds keyed harmonics (second or third—switch-selectable) to generate chirp at the onset of a note, mimicking the sound of a celeste or the “chiff” produced by an organ pipe when it begins to speak. The signature sound of a B-3 with percussion has been prominent in a lot of popular music, and examples can be heard at: [https://www.youtube.com/watch?v=IvFk\\_CNv0GQ](https://www.youtube.com/watch?v=IvFk_CNv0GQ).

## Hammond Models

Some Hammond organs were manufactured as spinet models, (“L”, “M”, etc.). More feature-rich console models include the “A” (the first model introduced), “B”, “C”, “D”, “E”, “RT”, etc., and later solid-state models. By far the most popular is the B-3, used by numerous jazz and rock musicians. **Photo 4** shows the RT-3, which is similar to the B-3, but has a standard American Guild of



Photo 4: Hammond RT-3 organ

Photo 3: This is a Hammond tonewheel generator. (Image courtesy of the Hammond Orgel Club, Holland)



Organists (AGO) concave radial 32-note pedalboard, and the solo pedal unit. Further information on the various Hammond models can be found at: [https://en.wikipedia.org/wiki/List\\_of\\_Hammond\\_organ](https://en.wikipedia.org/wiki/List_of_Hammond_organ)s.

As previously mentioned, the pedal solo unit was completely hollow-state, as was the preamp inside the console of each Hammond organ. There are actually three preamps on the “preamp” chassis. The top two (non-vibrato channel and vibrato channel) consist of a 6AU6 pentode

Photo 5: Hammond Tone Cabinet and the connecting cable



Photo 6: Leslie Model 122 speaker

feeding a triode—12AX7 for the non-vibrato channel and 12BH7 for the vibrato channel. Both are transformer-coupled at the input. The non-vibrato channel has additional triode stages for the expression pedal interface, a radio-phono input, and a tone control. It has a transformer-balanced output that feeds the external power amplifier.

The third preamp has a transformer-coupled input for the percussion signal, and actually constitutes a gated push-pull amplifier using a 6C4 triode as a pre-driver, a driver transformer for phase splitting, and the two halves of a 12AU7 dual triode as the push-pull pair. The 6C4 triode at the bottom gates the amplifier to provide decay at the end of the percussion chirp. The percussion and vibrato signals are mixed with the non-vibrato signal after the 6AU6 pentode stage of the top preamp.

Although many Hammond organs manufactured for home use contain power amplifiers and speakers, the professional models use external tone cabinets (see **Photo 5**). Various models of tone cabinets were made, containing different amplifiers and combinations of speakers. Each tone cabinet contained a power amplifier whose balanced input signal and AC power were supplied through a six-conductor cable. The schematic of such a tone cabinet amplifier consists of two balanced 6SN7 dual-triode stages driving a parallel-push-pull output stage using four 6V6s. According to the *RCA Receiving Tube Manual*, this circuit should produce about 28 W maximum output.

### Use of Reverberation

Since the pure tonewheel waveform is a combination of sine waves, a Hammond organ can share the tendency of a pipe organ to sound like a calliope unless some reverberation is added. In the case of the pipe organ, the reverberation is supplied by the large room in which the organ is located. Hammond organs are typically used in smaller rooms, so another method had to be used to provide reverberation. Seeking a way to synthesize reverberation for his organ, Laurens Hammond discovered that Bell Labs had invented an electromechanical device to simulate the signal delay experienced on long distance calls. The device used two springs to transmit the delay signal, and four additional springs to stabilize the mechanism. One of the springs transmitting the delay signal ended in a short tube partially filled with oil, and varying the amount of oil in the tube varied the decay time. Hammond modified the device to produce multiple decaying signals. The large Hammond spring reverb



Photo 7: Leslie 122 power amplifier

unit was mounted in the tone cabinet. (After numerous years of development, this device evolved into the famous Hammond/Accutronics Type 4 reverb that was pretty much the standard for guitar-amplifier reverb effects.)

### The Speaker

Donald Leslie began working in the late 1930s to create a speaker for a Hammond organ that more successfully emulated a pipe or theatre organ. He discovered that baffles rotating along the axis of the speaker cone gave the effect he was seeking. Hammond was not interested in marketing or selling the speakers, so Leslie sold them himself as an add-on, targeting other organs as well as Hammond. Leslie made his first speaker in 1941.

The Leslie model 122 speaker (see **Photo 6**) has a rotating drum with one side opening, placed below the 15” woofer, plus a pair of rotating horns driven by a compression driver. It produces a combination of amplitude and phase modulation, using the Doppler Effect. The drive motor has three speeds: off (no effect), “chorale” or low speed, and “tremolo” or high speed. Organists can use the gradual ramping up and down of the Leslie’s speed to great musical effect.

**Photo 7** shows a Leslie 122 power amplifier. The balanced input and AC power are supplied via the same 6-pin connector as is used for Hammond tone cabinets. Both triodes of a 12AU7 are employed to drive a push-pull pair of 6550 beam power tubes using fixed bias, providing a maximum output power of about 40 watts. Donald Leslie, like Laurens Hammond before him, was interested in a clean sound. However, during the 1960s many rock players, and later some jazz players, adjusted the Hammond organ preamp so it would overdrive the Leslie power amplifier providing a “Hammond growl” that was used effectively in quite a bit of music. One example is the opening G7 chord about 26 seconds into “Gimme Some Lovin” by the Spencer Davis Group (<https://www.youtube.com/watch?v=xcxYX8KPhGk>). Other examples can be heard in “The Greatest Hammond Organ Solos, Part 3” ([https://www.youtube.com/watch?v=IG5-s\\_G2csM](https://www.youtube.com/watch?v=IG5-s_G2csM)).

Many blog posts discuss this particular variant of the Hammond organ sound. The technical details on its creation will be the subject of a future Hollow-State Electronics article.

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

**November 1-3, 2019**

## Capital Audiofest

Hilton Hotel at Twinbrook Metro,  
1750 Rockville Pike, Rockville, MD  
[www.capitalaudiofest.com](http://www.capitalaudiofest.com)



Capital Audiofest (CAF) is now established as one of the East Coast's most enjoyable audio shows. CAF started out in 2010 as a very casual, DIY show and has evolved into a well-organized and well-attended event with which almost everyone in the audio industry is familiar.

CAF is recognized as a fun, friendly, and family-oriented show, where parents are often seen with their children strolling from room to room, listening to music, auditioning gear, and browsing the multiple bins of vinyl records, CDs, and accessories in the Atrium Marketplace.

In the evenings there is live music in the bar and special live performers in the Theater, and local distillers offering tastings. Visitor turnout has been gradually expanding and the show benefited from the Rockville Hilton's renovations. The hotel has a major Metro line right out the back door, connecting from/to the center of Washington DC with no transfers. Also, Rockville has great restaurants and shopping.

**November 8-10, 2019**

## New York Audio Show

Park Lane Hotel, Central Park South,  
New York, NY

[www.chestergroup.org/newyorkaudioshow](http://www.chestergroup.org/newyorkaudioshow)



The New York Audio Show will return to Manhattan, once again taking place at the luxurious Park Lane Hotel on Central Park South, one of New York's most prestigious locations. Taking place a week after Capital Audiofest, this sequence has been working great for the promoters of The New York Audio Show, Chester Group, allowing many exhibitors to participate in both shows. The New York Audio Show takes up three classic floors on consecutive levels in the hotel, with spectacular views that overlook Central Park. Some of these rooms are among the largest available, allowing perfect demonstrations of the best in high end audio and home theater.

Located directly on Central Park South (aka 59<sup>th</sup> Street), in Manhattan's toniest midtown location, the Park Lane Hotel is just steps from Fifth Avenue shopping, Broadway theaters, the Museum Mile, Carnegie Hall, Radio City Music Hall, the Lincoln Center, and many other NYC activities and attractions.

**November 9-10, 2019**

Shanghai Headphone Festival  
Shanghai Marriott Hotel City Centre,  
Shanghai, China

[www.canjamglobal.com/shanghai2019](http://www.canjamglobal.com/shanghai2019)



CanJam is the world's premier headphone audio show with annual events in New York City, Singapore, Los Angeles, London, and Shanghai and is produced by Head-Fi.org. The second edition of CanJam Shanghai promises to be the busiest thus far, and larger than ever given the success of the CanJam formula in China.

The show is established as the headphone audio industry's leading platform for exhibiting brands to expand their product distribution channels, interact with their customers and industry peers, and engage with the international audio, lifestyle, and technology press.

The Shanghai Headphone Festival 2019 will feature more than 100 leading brands in over 20,000 ft<sup>2</sup> of exhibition space, taking place at the fantastic Shanghai Marriott Hotel City Centre located in the heart of downtown Shanghai and within minutes from Nanjing Road, the Bund, and other major tourist attractions, providing easy access to explore the whole city.

**November 13-15, 2019**

## InterBee 2019

Makuhari Messe, 2-1, Nakase, Mihama-ku,  
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Promoted by the Japan Electronics and Information Technology Industries Association (JEITA), and run by the Japan Electronics Show Association (JESA), with the support of the Japanese government, all major broadcasting organizations and associations in Japan, and certified by the United States Department of Commerce, the International Broadcast Equipment Exhibition (Inter BEE) is always an important event for the professional audio and video industries. The event traditionally focuses on television technologies and is chosen by all major Japanese companies for key technology and product presentations—sometimes well ahead of any other show. Inter BEE is an important opportunity to promote solutions in the Japanese market with a large impact across Asia. The location, at the Makuhari Messe, is highly convenient and exhibitors recognize a high level of visitors, including many attendees from remote areas of the world. Professional audio equipment companies represent close to 100 exhibitors, including all major microphone manufacturers. In anticipation of the 2020 Tokyo Olympics the show will reflect important industry's efforts to enhance broadcasting services with 4K and 8K production and transmission.

**November 18-24, 2019**

## Live Design International (LDI) Show

Las Vegas Convention Center, Las Vegas, NV  
Exhibition: November 22-24

[www.ldishow.com](http://www.ldishow.com)



Founded in 1988, LDI (Live Design International) connects design, technology, and production professionals to the live audience experience. As the leading conference and trade show for live design professionals from all around the globe, LDI hosts more than 14,000 members of the production community, working in a wide range of international live and broadcast venues—festivals, corporate events, theatre, concerts, clubs, theme parks, archtainment, and houses of worship. From leading industry training, hands-on product demos, vendor-designed presentations and real-world product applications to unique experiential networking events, LDI offers something for everyone. Nearly 400 companies exhibit, mostly stage and lightning vendors but also many professional audio companies, providing live demos of cutting-edge gear.

**December 2-6, 2019**

## 178<sup>th</sup> Meeting of the Acoustical Society of America

The Hotel del Coronado, San Diego, CA  
[acousticalsociety.org](http://acousticalsociety.org)



The 178<sup>th</sup> Meeting of the Acoustical Society of America (ASA) will be held during the first week of December 2019 in San Diego, CA. The Society holds two meetings each year—one in the US and the second one abroad. At each meeting, invited and contributed papers and poster presentations, usually number between 850 and 1,100, reflect the diversity and interests of the membership. Tutorials, lectures, technical and standards committee meetings, and visits to university, government, and industrial laboratories are regular features, along with receptions and a variety of social programs. These meetings offer opportunities for students and young researchers as well as experienced acousticians to share information. Members who are able to attend meetings regularly find the personal contacts a rewarding experience. 2020 will be the International Year of Sound! IYS2020 will be a global initiative to highlight the importance of sound in all aspects of life on earth and will work toward an understanding of sound related issues at the international level. <https://sound2020.org>.



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