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By Jim Wood

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Fast, Wired, Wireless?

By now, every electronics designer should be aware of the updated Universal Serial Bus (USB) specifications — including the fact that there is a new USB Type-C standard connector and a new USB Power Delivery (PD) 3.0 specification, which is complemented with a certification program for USB Fast Chargers and supported Programmable Power Supply (PPS) feature of the updated specification.

Since 2014, the USB Implementers Forum organization (www.usb.org) has been continuously working on these USB updates, its logo, compliance, and certification programs, supporting USB 3.2 Gen 2 product development, and the critical Certified USB Charger initiatives. More importantly for our readers, in September 2016, the USB-IF announced the USB Audio Device Class 3.0 specification to establish USB Audio over USB Type-C as the primary solution for all digital audio applications, while the USB 3.2 update, published in September 2017, introduced support for up to two lanes of 10 Gbps operation, doubling the performance across existing cables. This could allow device manufacturers to eliminate the need for multiple ports and efficiently deliver data, power, and video over a single connector with USB Type-C, with compelling user benefits of device interoperability, ease-of-use across all digital audio applications and added capabilities to reduce power consumption. More than simply paving the way to the removal of the 3.5 mm analog audio jack on mobile devices, this could open up a whole new world to designers of devices with built-in digital signal processing (e.g., headphones, portable audio players, and recording interfaces).

But as I wrote in one of my editorials back in 2014, the audio industry is increasingly relying on standards and technologies that it doesn't control, and all these initiatives are directly supported by major technology companies in the consumer, mobile, and IT spaces. These companies have their own priorities according to their own - quickly shifting - market requirements, and we still don't see audio-specific requirements reflected as they should be on the products introduced to the market.

While everyone is praising the fact that USB Type-C is smaller and reversible, the main reason manufacturers are adopting USB-C on new devices has to do with USB Power Delivery. Meanwhile, the mobile industry is also quickly moving toward wireless charging since the Wireless Power Consortium' Qi Wireless charging technology gained support from the most important brands, including Apple. Of course, Qi Wireless charging pads still need a power adapter, and the models supporting up to 10 W wireless charging specifically need USB-C PD. No wonder USB-C power adapters and fast chargers are the hot product for 2019.

Precisely for this edition of *audioXpress*, we intended to publish one of our Market Update features, focusing on these USB-C topics – including obviously USB Audio 3.0. We plan those features in advance, and it seemed logical to discuss the huge implications for DSP, headphone audio output, active noise cancelling, wireless audio dongles, etc.

Indeed, USB-C and Power Delivery have been one of the most active areas for semiconductor companies, but when we looked at the announcements made during 2018 it became painfully obvious that we are still missing many critical pieces from the audio product development perspective. We could gather lots of interesting information relevant for electronic designers in general, but not specific to an audio-focused publication.

Ironically, as I was reading those traditional “best headphones you can buy” end-of-the-year lists published by some consumer electronics websites, I couldn't help noticing how the “best” models in 2018 all mentioned USB-C on the “pros” side, while the “cons” specifically mentioned the use of “older Micro USB charging” as one of the reasons why 2018 models didn't make it to the top of the list. The message here for consumers is that micro USB connectors are not only highly inconvenient, but they are a stand out indicator that a product doesn't benefit from the latest fast charging possibilities, which is supposed to be the requirement consumers are noticing.

It seems that the promise of the technology still needs to be realized in terms of USB Audio 3.0 adoption. For those reasons, we have decided to wait, and reschedule that Market Update for later in 2019.

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.

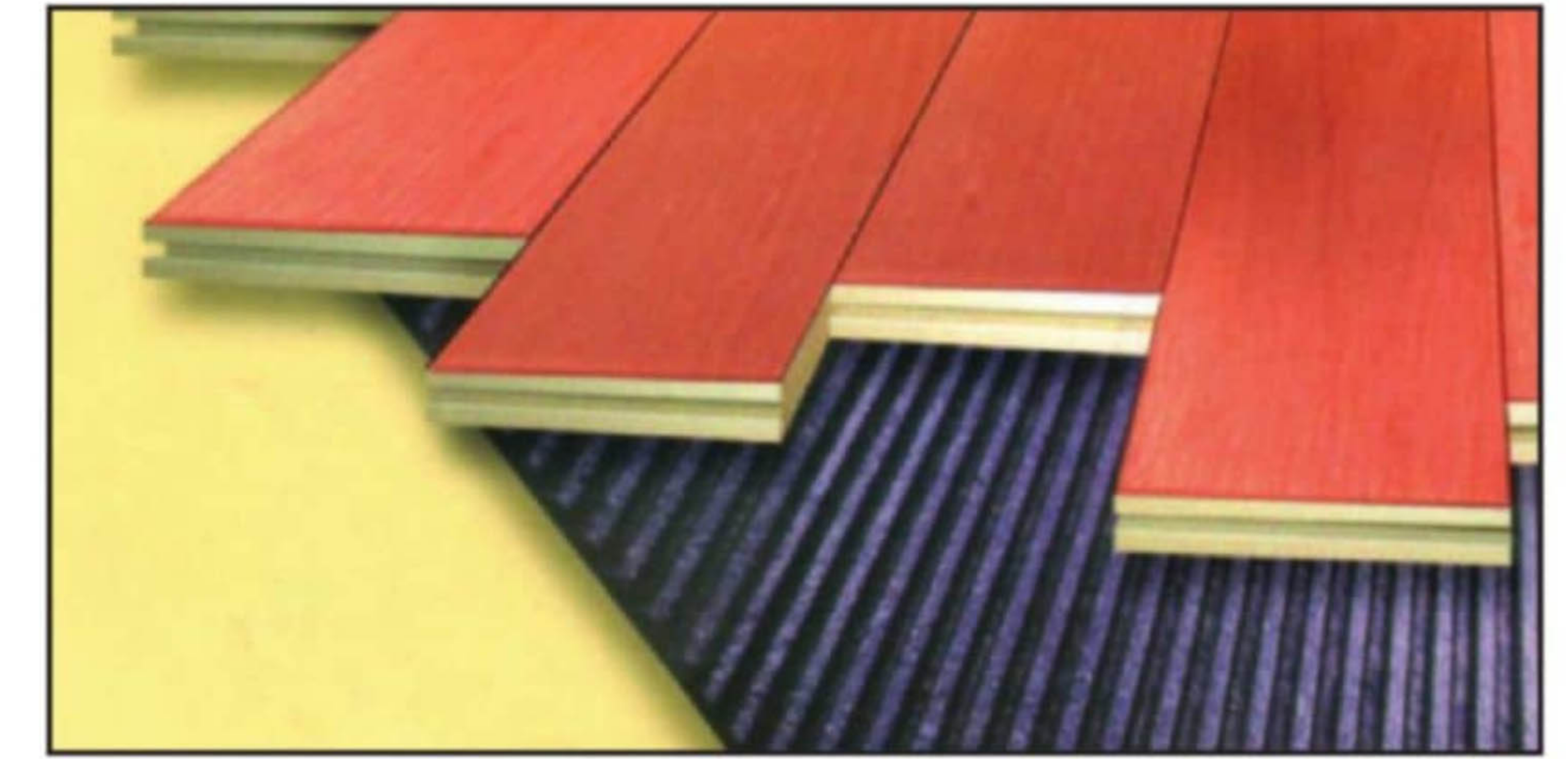
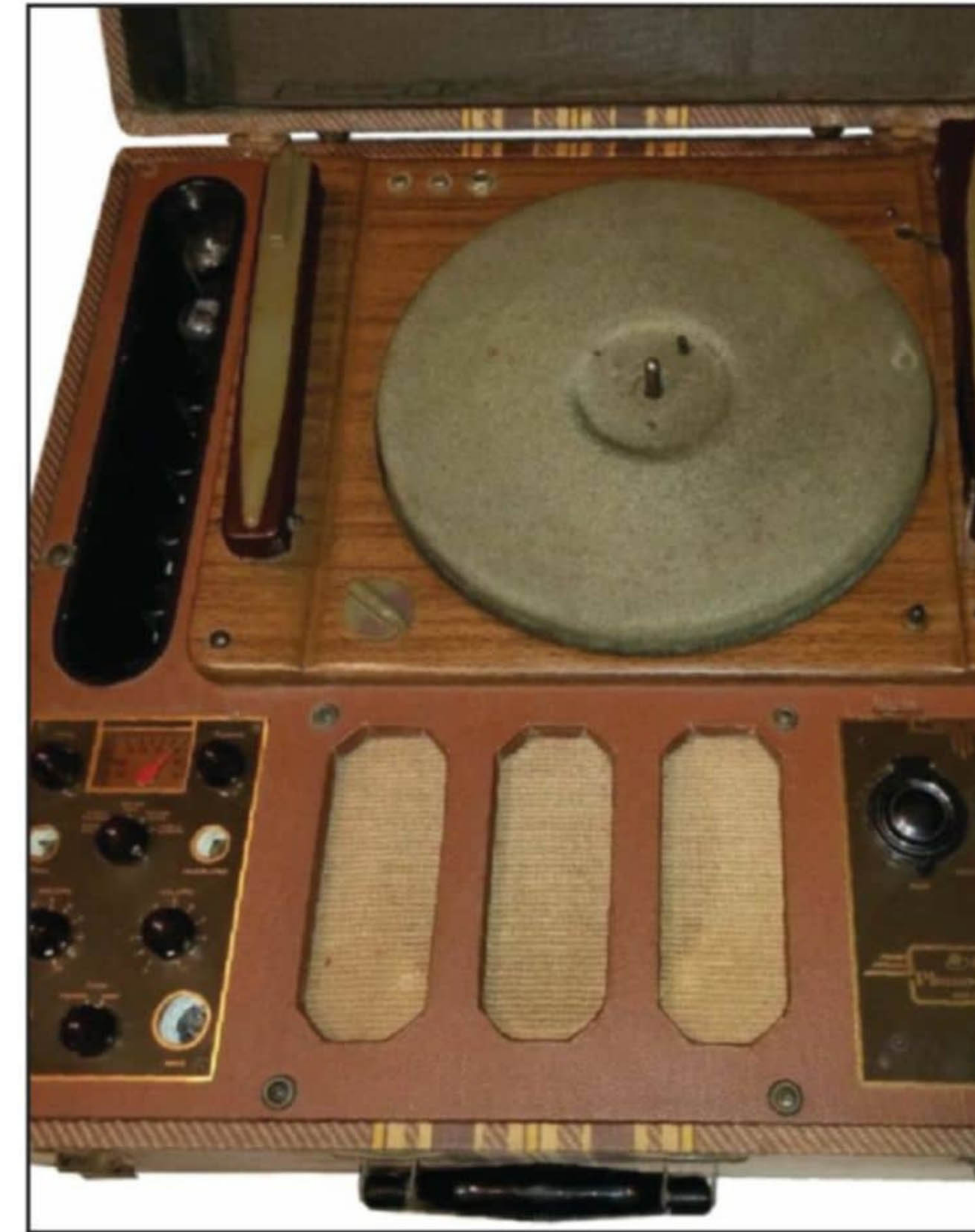
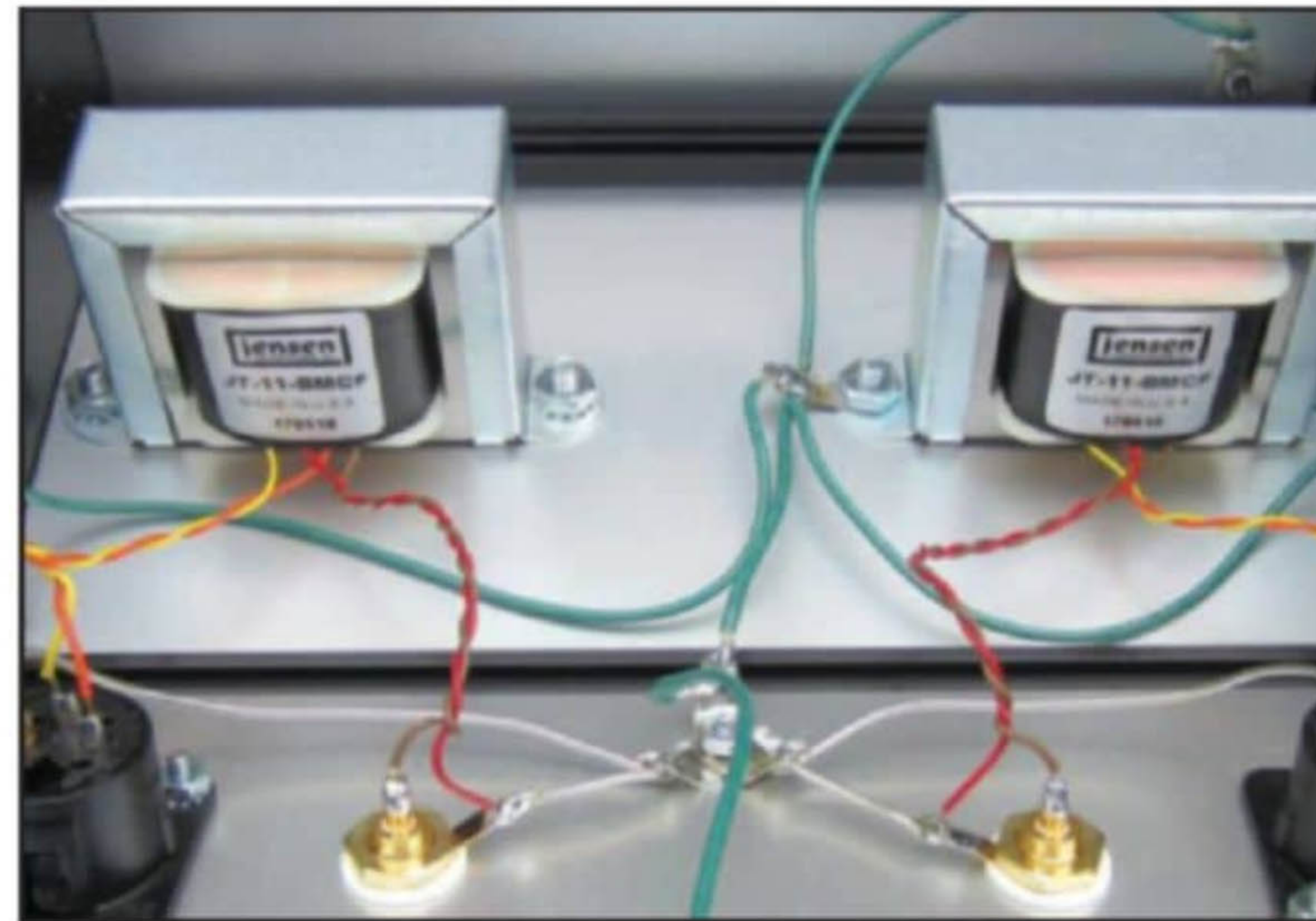
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Spotlight on Pascal

PASCAL

Pascal is a recognized OEM manufacturer of high-power amplifier modules at the global level. Focused in the professional loudspeaker market with power solutions based on proprietary technologies and design standards, the company is now in the top one percent of highest performing Danish companies, doubling its business every second year, over the last six years.

By
Ward Maas
(The Netherlands)

That Denmark and the Danish capital Copenhagen are a focal point for the audio industry in Northern Europe might be a recognized fact, but apparently, they are getting ready to take on the world! During the last Copenhagen High End show that I attended in October 2018, there were a lot of high-tech startup audio companies, all eager to present their products to the world. It might not be well known, but Copenhagen is now the Class-D amplifier capital of the world. In earlier *audioXpress* articles focused on suppliers of amplifier modules, I discussed and visited companies such as ICEpower and Merus Audio (now Infineon), all based in Copenhagen. But there are more. Among the OEM manufacturers, there's also Pascal, which I recently visited, and is a recognized name in the professional audio segment. A "niche" leader, as they like to say.

The Technical University of Denmark in Lyngby has a very active power electronics and switch-mode technology research department, and it is

no wonder that it would result in a lot of startups focused in the Class-D amplification field.

You might ask where all those companies will find a market or markets that suit them. Although Class-D amplification is still relatively new, you can already determine the specialization on certain market segments by the larger companies. While companies such as Hypex or ICEpower profile themselves as high-quality brands competing in the high-end audio segment, Pascal has chosen a different path. It has a clear focus on the professional market.

Pascal was founded in February 2006 by Lars Rosenkvist Fenger, Jesper Lind Hansen, and Peter Frenzt, with CAMCO (the German audio company) as a minority shareholder. Pascal's founding directors identified a market opportunity in a demand for dedicated power amplifier modules—optimized for professional audio, both for self-powered speaker and power amplifier manufacturers—that was not

Photo 1: The newly expanded Pascal headquarters and facilities building is located on the outskirts of Copenhagen.

being met. In 2014, the company moved into larger headquarters, doubling the number of employees in just 12 months. At the beginning of 2017, after CAMCO was taken over by L-Group, the holding company for L-Acoustics, a buyout took place, executed through a joint holding company that is now the sole owner of the company. Pascal A/S is now 100% in hands of its Danish founding team.

Having thrice outgrown its operating base in the 12 years since it was founded, in early 2018 the company completed a 55% expansion of its current headquarters in the Herlev suburb of Copenhagen (see **Photo 1**). The expansion of its premises, now occupying an area of 1,726 m², was driven by rapid organizational growth—particularly in areas of R&D, operations, and sales. The new building project was also planned to accommodate a new Electromagnetic Compatibility (EMC) testing laboratory, allowing staff to test all electrical equipment compliance onsite.

The state-of-the art facility has been specifically developed with audio equipment in mind, such as self-powered speaker systems and 19" rack amplifiers. The chamber is highly automated, being fitted with an automatic turntable and antenna polarization. Measurements and control data are remotely accessible to staff in the company's engineering facilities for optimal convenience.

From its inception in a lakeside cabin, the company grew rapidly and at an accelerating rate due in equal measure to its advanced technology, engineering, and a visionary business model, which focused on the requirements of professional audio companies. Just recently, Pascal became one of only 20 companies in Denmark to be awarded the Børsen Gazelle Award for six years in a row, placing it firmly among the country's top highest performing companies (see **Photo 2**). Awarded by *Børsen*, a leading Danish business daily newspaper, the Gazelle award recognizes continuous growth in turnover or gross profit over a period of four consecutive years, where either of those metrics has doubled during that period.

When Senior VP Business Development Peter Frenzt accepted the award on behalf of the company and fellow founding directors, he said that although the award is in recognition of financial success and continued growth, in fact, it is the result of the company's unique "approach to engineering, our sales and marketing strategies, and defines our client relationship management. Most of all it is the DNA of our company, our people, and our organizational culture."

But as always, a company story isn't always made of straight lines. Pascal learned the hard way

that a failure can be the best learning experience. The failure of its first product launch taught the team a few lessons in a very short time. It was Pascal's first and last failure, and a very clear reason to completely re-evaluate product development and market approach.

Finally, this resulted in Pascal's high-power density amplifier module design, and a marketing strategy focused on unique levels of customer support and development partnership. That way it offers clients significant added value and strategic advantage, a faster time to market, and risk mitigation within their product development programs.

An Expanding Range

The Danish manufacturer leveraged its proprietary patented UMAC class-D amplifier technology and UREC power supplies to claim a leading market position in the pro-audio self-powered PA sector, where it provides benchmark designs. Pascal's UREC Power Factor Correction (PFC) power supply technology also enables universal AC mains operation, delivering consistent, regulated power worldwide and eliminating susceptibility to local power related reliability issues.



Photo 2: Pascal's founding directors, CEO Lars Fenger, Senior R&D Specialist Jesper Lind Hansen, and Senior VP Business Development Peter Frenzt, proudly displaying six consecutive Børsen Gazelle Awards.



Photo 3: Pascal's founding team is pictured (L-R): CEO Lars Fenger, Senior R&D Specialist Jesper Lind Hansen, and Senior VP Business Development Peter Frenztz with the flagship S-PRO2 module launched in 2013. At the time it was the world's smallest 1 kW amplifier module with an integrated power supply.



Photo 4: Pascal U-PRO Series modules feature extremely high voltage rails, providing unique dynamic power ratings at a competitive price to dynamic power ratio.

Following the success of the original S-PRO modules, the company introduced its flagship S-PRO2 module in 2013 (see **Photo 3**). At the time, it was the world's smallest 1 kW amplifier module with an integrated power supply. The S-PRO2 was extremely well-received in the market thanks to its integrated universal power supply including PFC, combined with Pascal's UMAC class-D technology and wide range of protection features as well as auxiliary power for DSP cards. This made the S-PRO2 module an ideal candidate to power a wide range of one- and two-way cabinets (e.g., monitors, line arrays, and subwoofers).

In 2014, Pascal expanded the possibilities of the S-PRO2 module with the S-A2 and S-A1 expansion boards, which are able to function as additional identical amplifier output channels, powered by the S-PRO2 power supply. With the new extension modules, the S-PRO series gained a wide range of output channel configurations, allowing efficient three-way power configurations by deploying the S-PRO2 in bridged configuration with the addition of an S-A1 or S-A2 module providing either 500 or 1000 W to the low-frequency section.

In June 2015, Pascal launched its new T-PRO Series amplifier modules with a dedicated design optimized for ultra-compact high SPL integrated power applications—including asymmetrically rated two-way power channels. This allows for more compact high-powered speaker cabinets, and low-profile applications such as sub bass cabinets and other compact amplifier components. The T-PRO Series also boasts better efficiency, minimizing the requirement for heatsinks and cooling, and contributing to the amplifier's long-term reliability.

The series features ultra-low standby power consumption for EuP2013 and green energy star compliance, with an Auto Standby/Wake-up feature with selectable time settings. Additionally, the T-PRO Series has identical interfacing and identical output voltages to the popular higher current capable S-PRO2 module, making the front-end electronics of both "plug-and-play" compatible.

In 2016, the company launched another key product. The new Pascal U-PRO Series (see **Photo 4**), featuring extremely high voltage rails that provide unique dynamic power ratings, at a competitive price to dynamic power ratio. The design once again met the specific commercial requirements for self-powered, lightweight, portable PA and ultra-compact high-output fixed install cabinets. Offering both single and asymmetrically rated two-way power channels, with the smallest form factor, the U-PRO Series was also widely adopted for professional studio monitors.

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Photo 5: Pascal's EMC facility with EMC Engineer Claus Momme Thomsen



Photo 6: Peter Frentz with Pascal's latest 1500 W L-PRO2 module and the similar sized 1000 W S-PRO2

Parameter	Typical Value
Total output power (1% THD+N, 1 kHz at 4 Ω)	810 W + 810 W
BTL power at 8 Ω	1500 W
Dynamic BTL power at 4 Ω	2500 W*
Total system efficiency (SE, 2x 400 W at 8 Ω)	89.5%
Peak output current	
(Channel 1)	40 A
(Channel 2)	40 A
THD+N (1 kHz at 1 W)	0.003%
Dynamic range	120 dB(A)
Idle noise	54 µV(A)
Output resistance (1 kHz)	6.5 mΩ
Mains input voltage	85 VAC to 265 VAC
Standby consumption	0.23 W
SMPS power	1500 W

*Output voltage of 160 V peak, yielding a dynamic headroom of more than 2 dB and a peak SPL rating equivalent to a conventional 2500 W amplifier

Table 1: L-PRO2S product specifications

According to Pascal, due to its extremely high voltage rails relative to its RMS power output, the U-PRO series features RMS ratings of 280 W. The modules can also deliver ultra-short bursts of 500 W RMS or 1000 W peak power. At a lower cost point than normal market expectations for such power capabilities, these are optimal performance characteristics, even for fixed install cabinet designs.

In September 2018, Pascal surprised the market with the release of the new L-PRO2 series amp modules, providing 50% more power from the same form factor as the S-PRO series, and featuring the world's smallest 1500 W RMS amp module. Basically, the new L-PRO2 modules inherited the form factor from the S-PRO series—heralded as the world's smallest 1000 W amplifier module when it was launched in 2012.

An ultra-compact power platform for two-way and subwoofer applications, featuring extreme dynamic power ratings and integrated power supply with PFC, the L-PRO2 modules are optimally designed for integration within high power sound touring, fixed venue PA and line array systems, and large subs. Three variants address specific applications: the L-PRO2 (1200 W + 400 W RMS) and the L-PRO2HP (high power—1500 W + 800 W RMS) feature asymmetrical power configurations, with feature sets and a form factor uniquely optimized as a dedicated power solution for self-powered two-way loudspeakers, and the symmetrical output channels of the L-PRO2S (1500 W RMS in BTL mode) optimized for use in large-scale subwoofers.

Exceptionally high voltage rails combined with an ability to drive a low (4 Ω) impedance load, provide massive power headroom. As a result, the low-frequency channel of the L-PRO2HP and the L-PRO2S (in BTL mode) deliver an output voltage of 160 V peak, yielding a dynamic headroom of more than 2 dB and a peak SPL rating equivalent to a conventional 2500 W amplifier.

All modules are equipped with comprehensive protection features and designed on an “audio-at-all-times” principle, rendering them fully self-protecting. They all support Pascal's plug-and-play interconnect, which makes the complete range of 26-pin power amplifier modules inter-connectable to a single front-end I/O/DSP board design. A single front-end module can be used in multiple self-powered loudspeaker design configurations, reducing the required engineering work across multiple product design projects.

Already in October 2018, the company announced its entry into a new business sector, with the development of a comprehensive range

of dedicated modules for rack amplifier designs—targeting both live sound and integrator/installation markets. A new range of six modules enables both constant voltage (100/70 V) and low impedance output mode operation and a wide range of power configurations, from half rack 1RU to full 19” 2RU form factors. And depending on the combination of modules used within a power amplifier, it is possible to drive any share of the total power capability reservoir to any output stage. This offers installers and integrators complete freedom to drive the required amount of power precisely where it is needed. And as with other Pascal products, these also keep the 26-pin interconnect, making them inter-connectable to a single front-end I/O/DSP board design.

With a steady pace of product launches and continued innovation, it is no wonder that the company has been showing such growth, year after year. As Pascal's CEO, Lars Fenger, details “Our continual process of innovation is grounded in the unique abilities of our people to creatively exploit technologies to provide proprietary solutions. It is based on a culture of complementary individual and interpersonal development; something that we have nurtured from the very beginning. Human activity is team based at every level in our organization.”

A truly remarkable fact is that the company's eighth employee was a human resource manager. A strategic move that emphasizes the importance of having the right people in the right place. And where other companies write down their vision and mission statements, Pascal lives them. That way the implemented quality assurance and environmental protection standards (ISO9001, ISO14001, and ISO13485) are not felt as something they had to do, but as something natural.

Manufacturing and Implementation Support

In order to deliver Pascal's promise of tight integration with loudspeaker manufacturers and on-time deliveries, two CMS providers in China produce the line of amplifier modules. One specializes in high-mix low-volume production; the other, VTech, in Dongguan, specializes in high-volume low-mix production. Pro audio is a focus market for VTech, an industry sector in which it is one of the world's largest and most proficient CMS providers. Like Pascal, both companies comply with the highest quality assurance and environmental protection standards (ISO9001/ISO14001/ISO13485).

All amplifier modules are subjected to several quality tests during manufacturing. The final

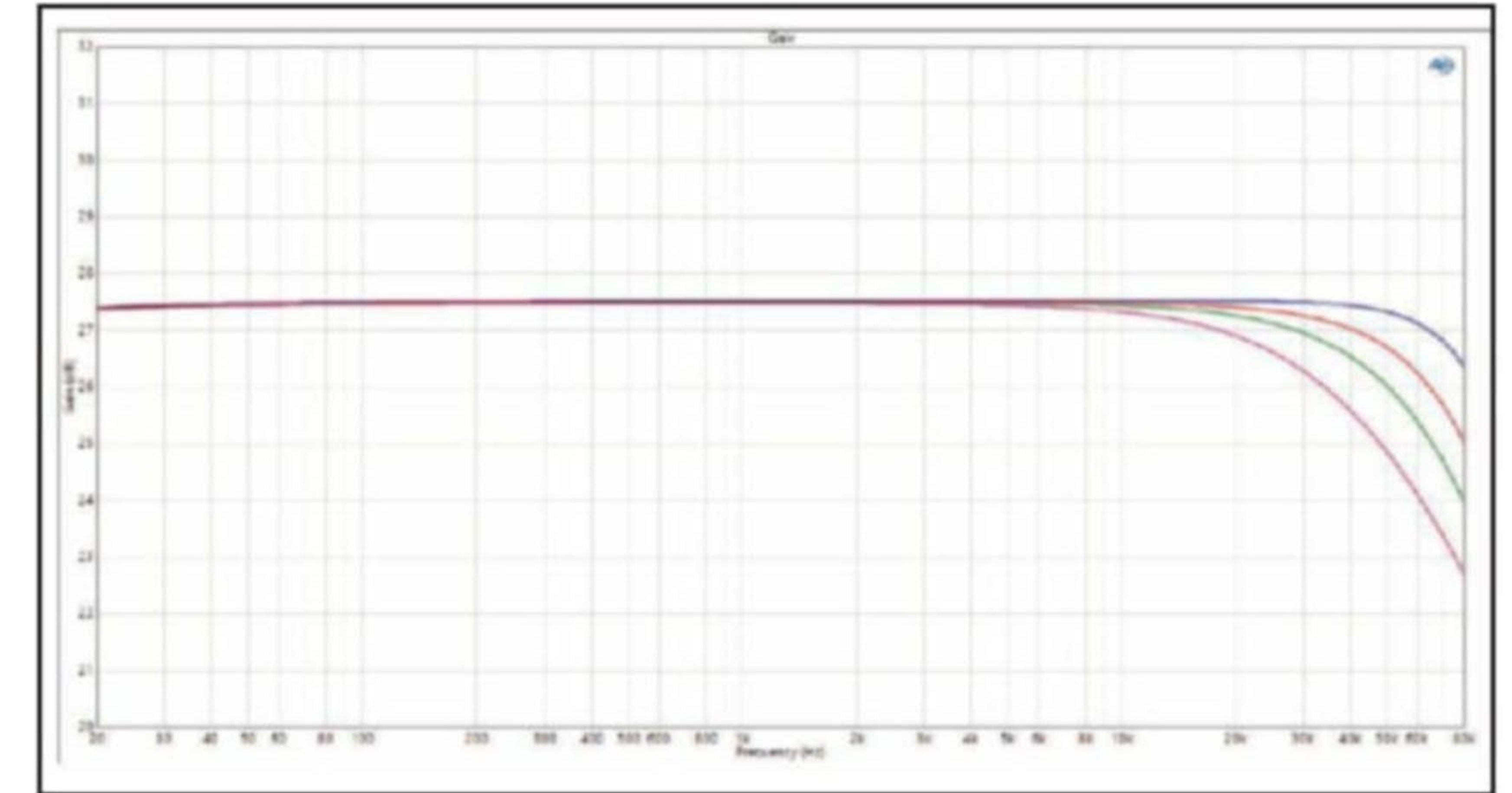


Figure 1: L-PRO2S frequency response, Amplitude 2 Ω (purple), 4 Ω (green), 8 Ω (red), and Open Load (blue)

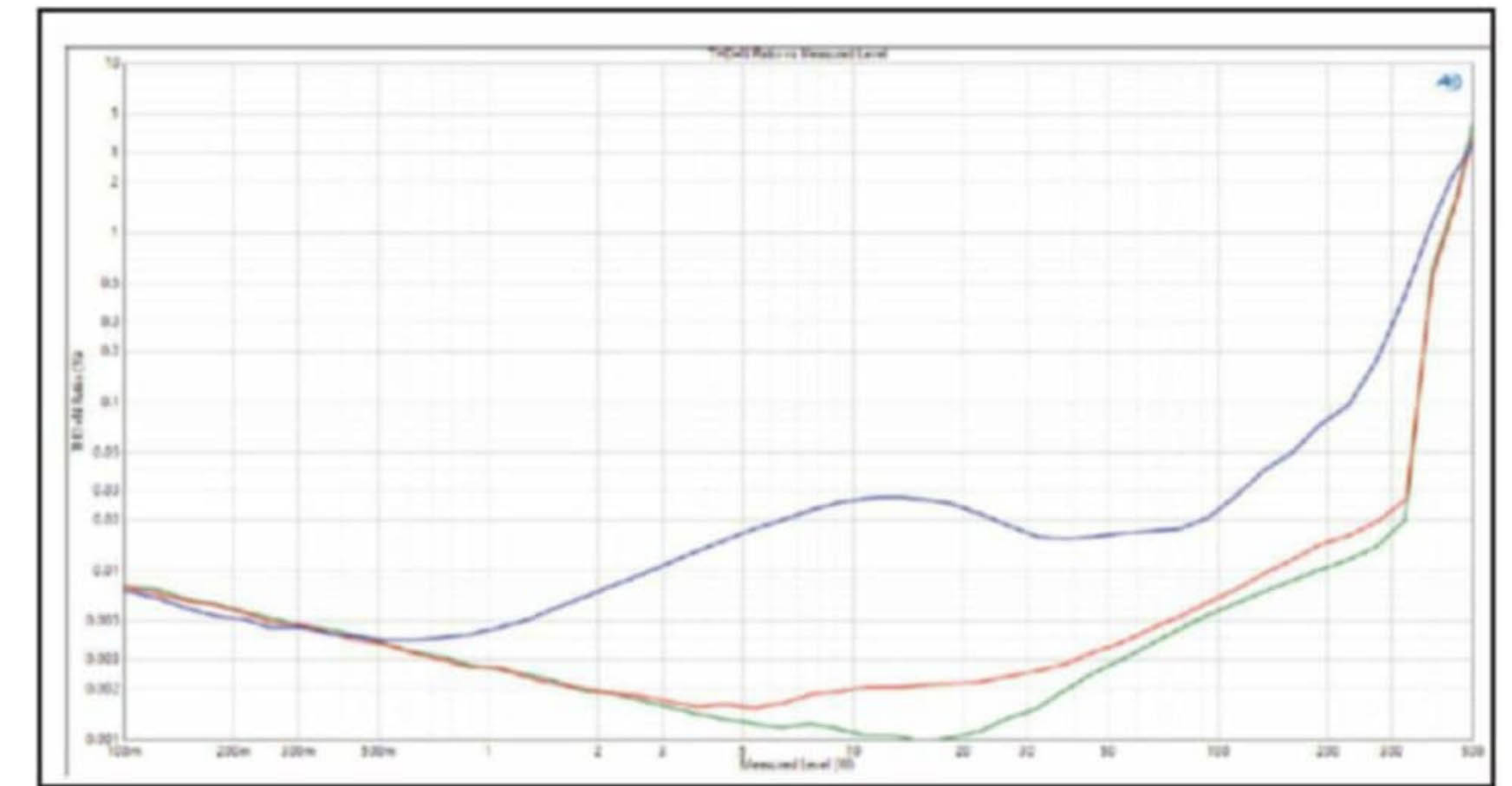


Figure 2: THD+N vs. Power at 8 Ω, 230 VAC/120 VAC (100 Hz, green), (1 kHz, red), and (6.67 kHz, blue) for Pascal's L-PRO2S amplifier module

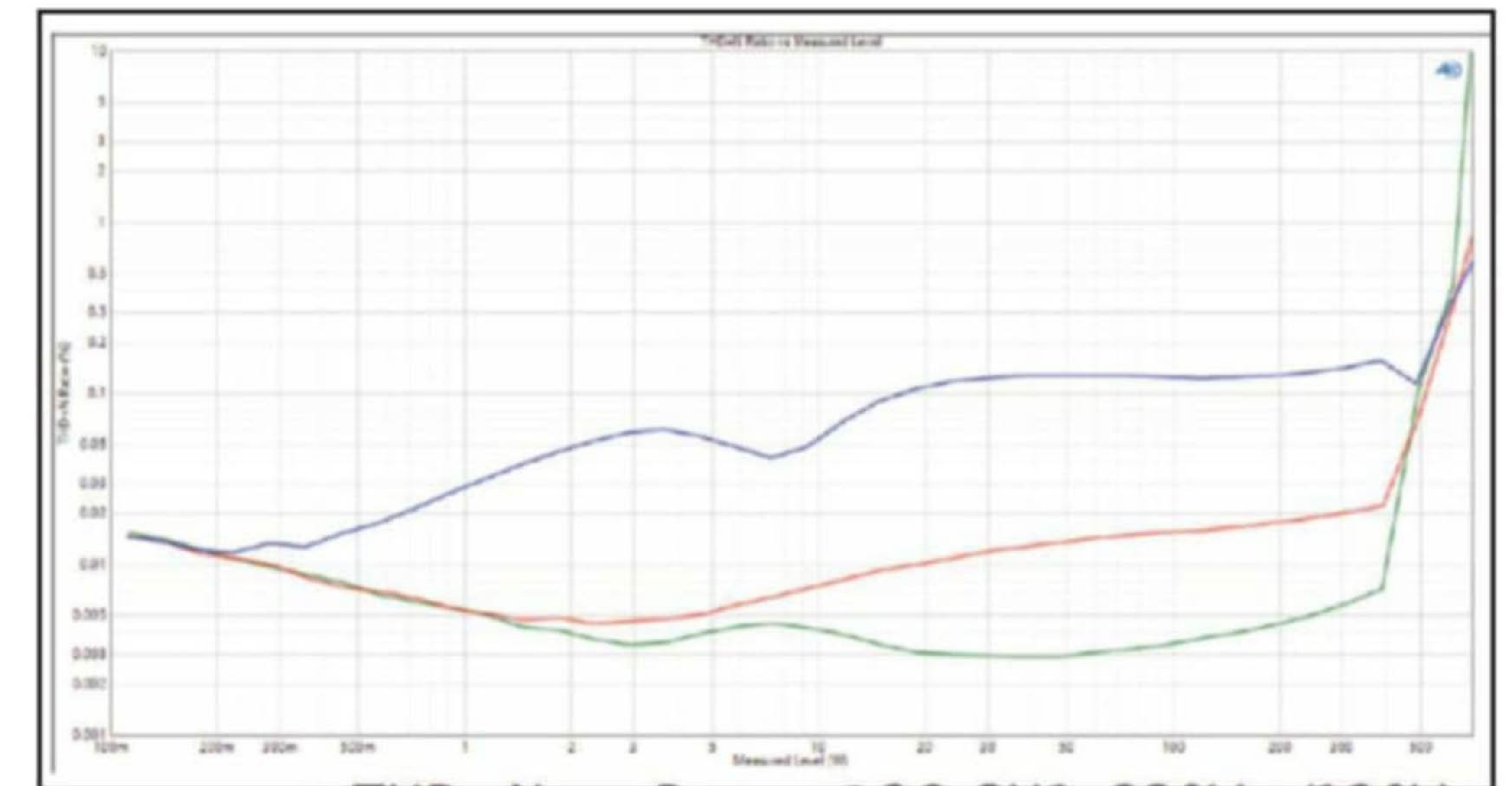


Figure 3: THD+N vs. Power at 2 Ω, 230 VAC/120 VAC (100 Hz, green), (1 kHz, red), and (6.67 kHz, blue) for Pascal's L-PRO2S amplifier module

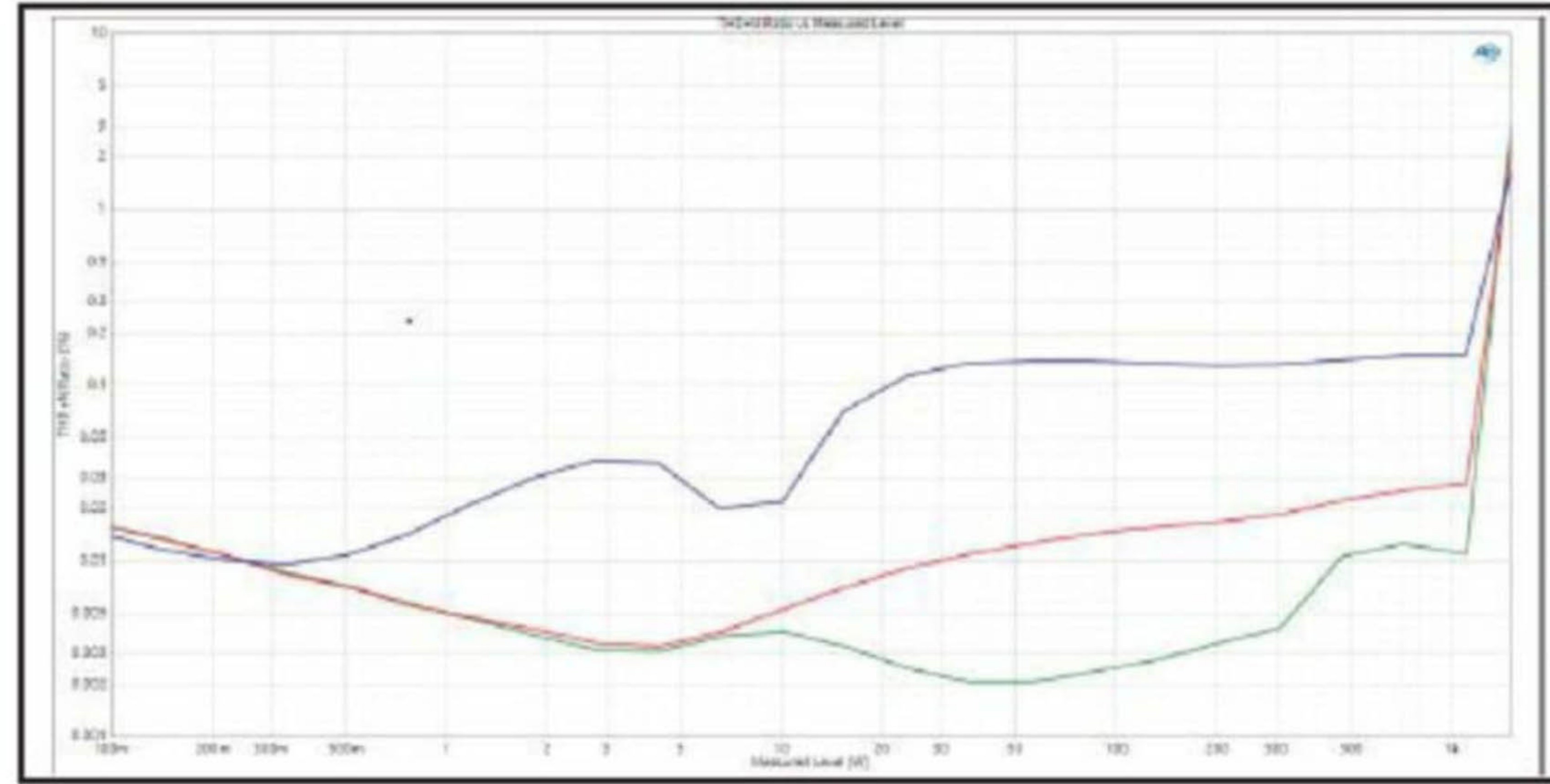


Figure 4: Pascal's L-PRO2S THD+N vs. Power at 4 Ω, 230 VAC/120 VAC (100 Hz, green), (1 kHz, red), and (6.67 kHz, blue)

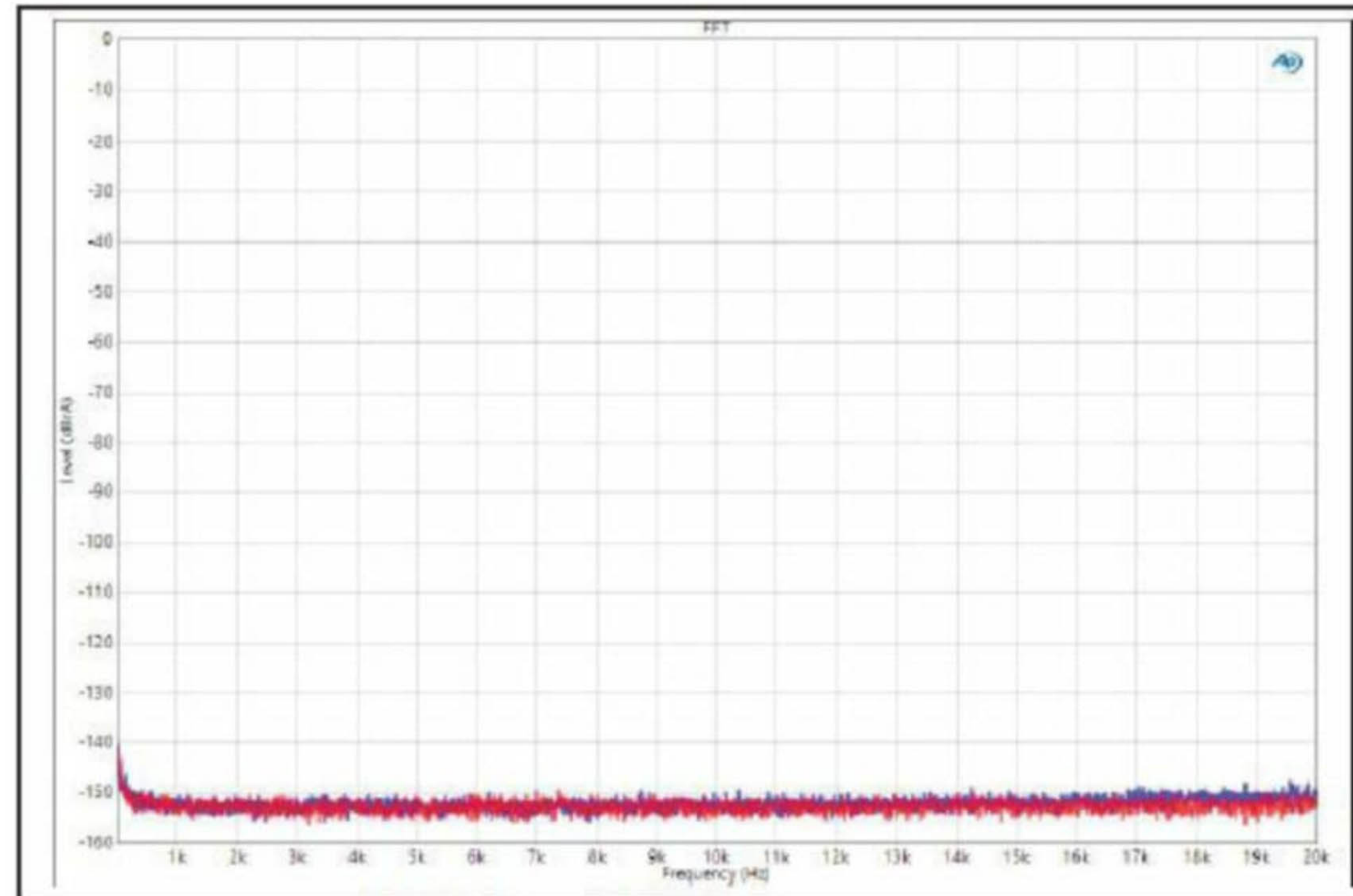


Figure 5: FFT idle (SE) 8 Ω, Channel 1 (blue) and Channel 2 (red) of Pascal's L-PRO2S amplifier module

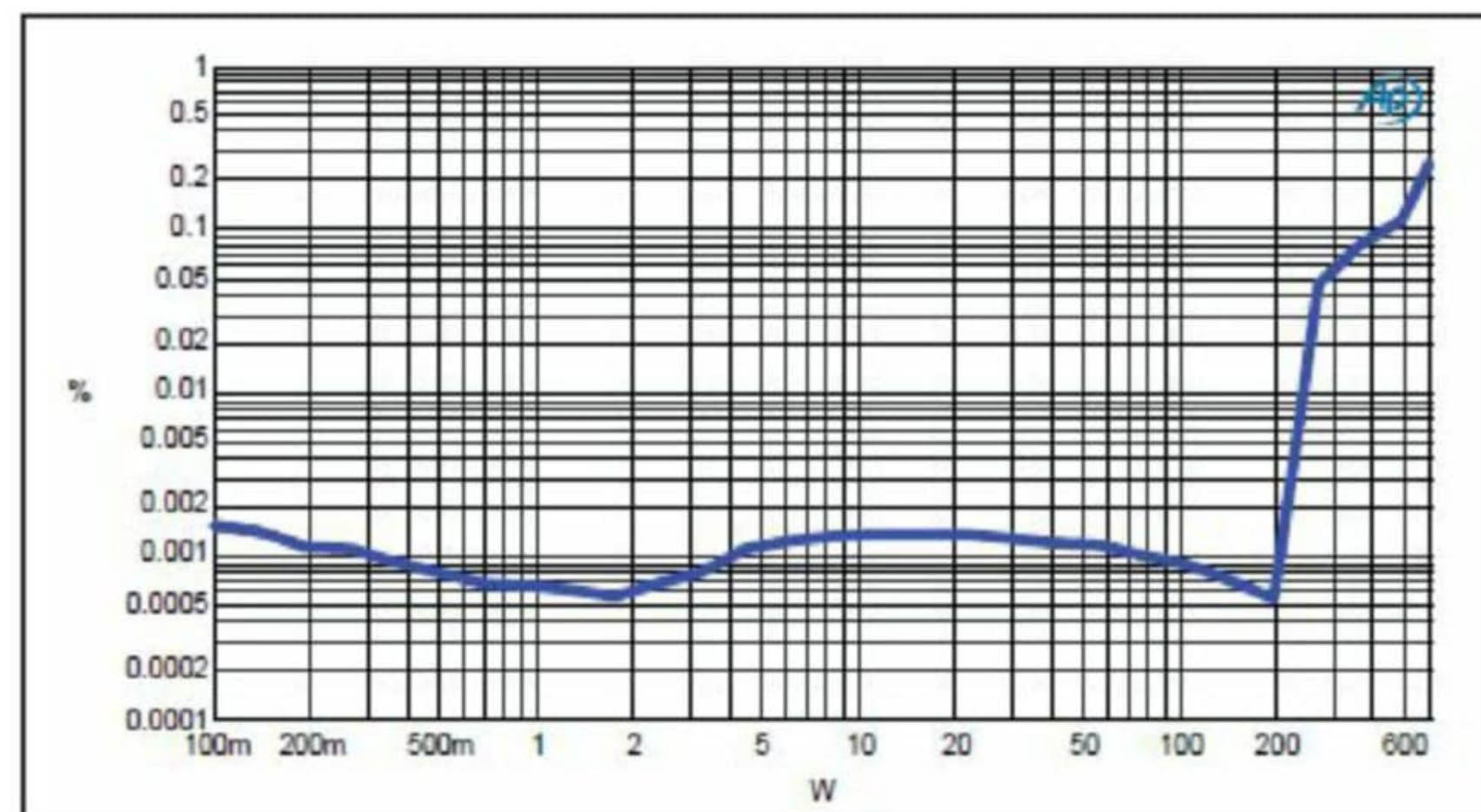


Figure 6: CCF vs. Power at 4 Ω, f1 = 18 kHz, f2 = 19 kHz of Pascal's L-PRO2S module

assembled amplifier module passes between 100 and 200 end-of-line test steps (depending on the specific model). These fully automatic custom test systems are designed by Pascal's own test and quality engineers, who are able to access all measurement data for each test step. All amplifier modules are tagged with unique barcodes, which are logged in the test systems. This makes any individual amplifier module fully traceable.

Of course, the focus on the pro-audio market requires that all power modules offer maximum reliability first. Also, the power supply is integrated with the amplifier. Pascal was one of the first to acknowledge the importance of the (switch mode) power supply for the amplifiers' performance and reliability. Also, as illustrated in the description of the product range, the front-end is the same for all products, offering clients a steady platform for their own developments.

In this professional environment, a strong product application support is essential, and Pascal helps to implement an amplifier module in the client's product. As I learned during my visit, that starts with knowledge of how to handle products in the desired application, includes the often-needed DSP support, and ends with EMC/EMI pre-compliance testing. This is what enables Pascal to be an OEM provider of amplifier electronics for many pro audio manufacturers in the industry.

The EMC facility enables Pascal to offer enhanced implementation support to help clients pass regulatory challenges for their end products. This capability is further enhanced by the recent expansion of the application support team with an extra Application Engineer who has been working as an EMC measurement specialist at a leading Danish accredited EMC test house (see **Photo 5**). Due to the high demand for EMC testing at accredited test-houses in Europe, waiting times are long. Not to mention the costs involved in such official testing. This made it easy to see why this capability to do quick pre-compliance tests is a major asset for Pascal and its clients.

With an extensive range of high-power density amplifier modules, Pascal is able to offer dedicated solutions for almost any number of output channel and power configurations. And these can be integrated within everything, from large-scale sound touring line arrays to portable PA cabinets and ultra-compact fixed installation systems.

A Close Look at a Flagship Module

The L-PRO2S can be justifiably named as Pascal's current flagship module (see **Photo 6**). As with all its amplifier modules, this design features

the company's proprietary core technology, the patented UMAC Class-D amplifier platform. Detailed in US patent 7893743, it was granted in 2011 to Pascal A/S and its two founders, Lars Rosenkvist Fenger and Jesper Lind Hansen.

Furthermore, this design also uses the UREC PFC power supply technology, automatically adaptive to any mains voltage between 100 V and 240 V. The "sauce" for the UREC technology is in the encrypted firmware in Pascal's own chipsets, which the company has preferred, as opposed to filing a patent. This means no public insight exists over Pascal's technology but also there is no risk of this knowledge being circumvented. Reverse engineering this encrypted firmware would be extremely difficult.

The L-PRO2S specifications (see **Table 1**) has to be read with the company's focus on the pro audio market in mind. With its outside dimensions of 215 mm × 80 mm × 55 mm, the power density is phenomenal (see **Figure 1**). A 1500 W BTL in 8 Ω and a dynamic 2500 W BTL in 4 Ω for such a size is impressive. Indeed, somewhere the power supply becomes the limitation, not the amplifier... With the option to add an LA2 module (which comes without a power supply) four times 810 W is available, again with the power supply of the L-PRO2S as a limiting factor.

With an efficiency of almost 90% and a thermal limit at 85°C, the L-PRO2S is a pleasant product to incorporate in a 2HE amplifier cabinet or to be used in self powered speaker cabinets. Due to the high output voltage the L-PRO2S is easily capable of driving 70/100 V systems.

As mentioned, Pascal strives for the best audio quality, but not at the cost of reliability. The harmonic distortion figures are good (see **Figures 2–4**) but not the best we have seen. The noise floor is very low (see **Figure 5**). The CCIF intermodulation distortion (an intermodulation test recommended by the International Telephonic Consultative Committee, and referred to by the initials CCIF) and transient intermodulation (TIM) are nice (see **Figure 6** and **Figure 7**). The output resistance is so low that the resistance of the output connector does matter (see **Figure 8**). Crosstalk is okay as well (see **Figure 9**).

More interesting is the extensive attention to AC mains and thermal behavior. In standby, the unit uses 0.17 to 0.36 W, depending on the mains voltage. In active mode but muted, the power consumption is between 3.8 to 5.5 W depending on the mains voltage. In idle mode, around 12 W is consumed. The capability to deliver its full power under very different mains conditions is again

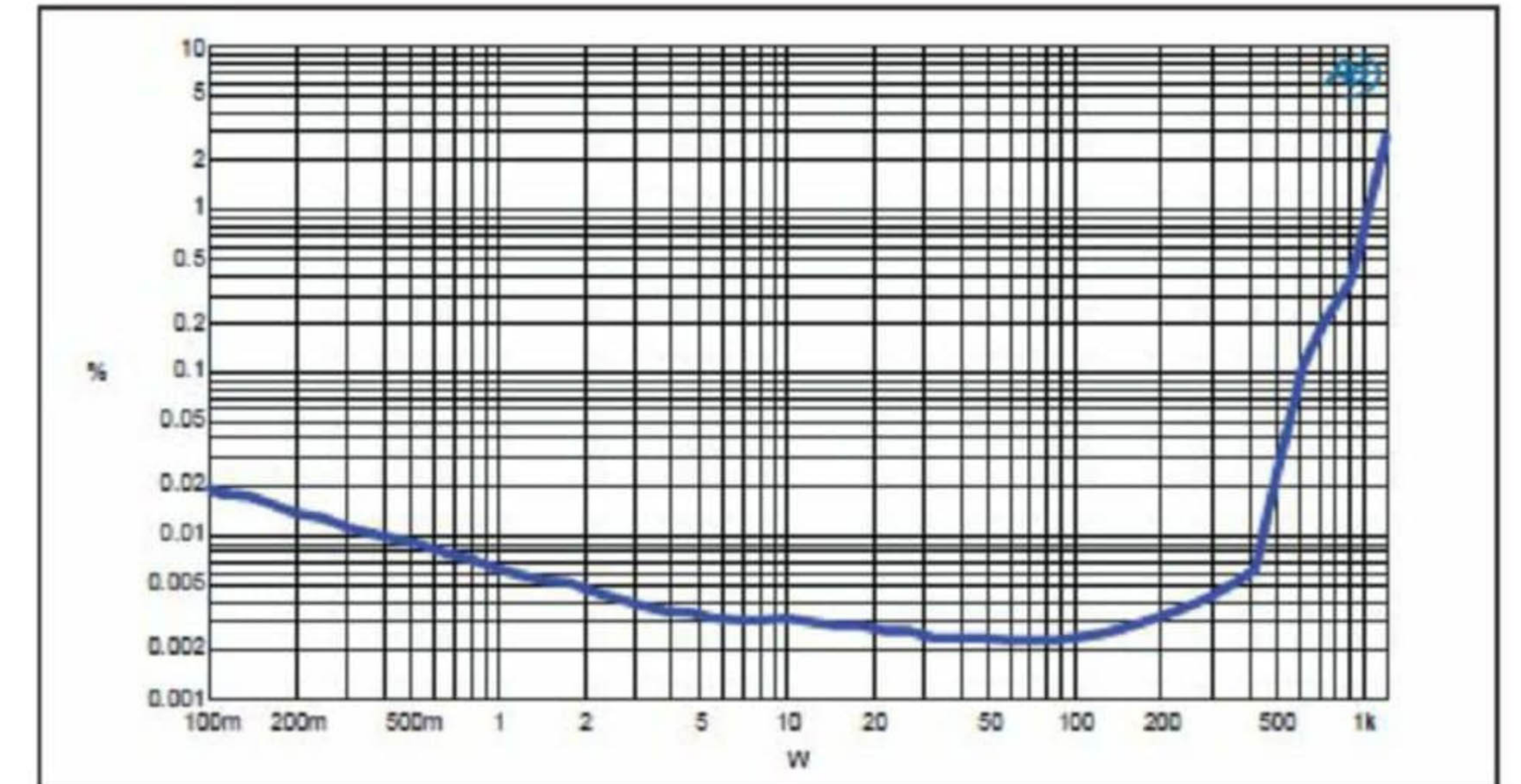


Figure 7: TIM vs. Power at 4 Ω of Pascal's L-PRO2S module

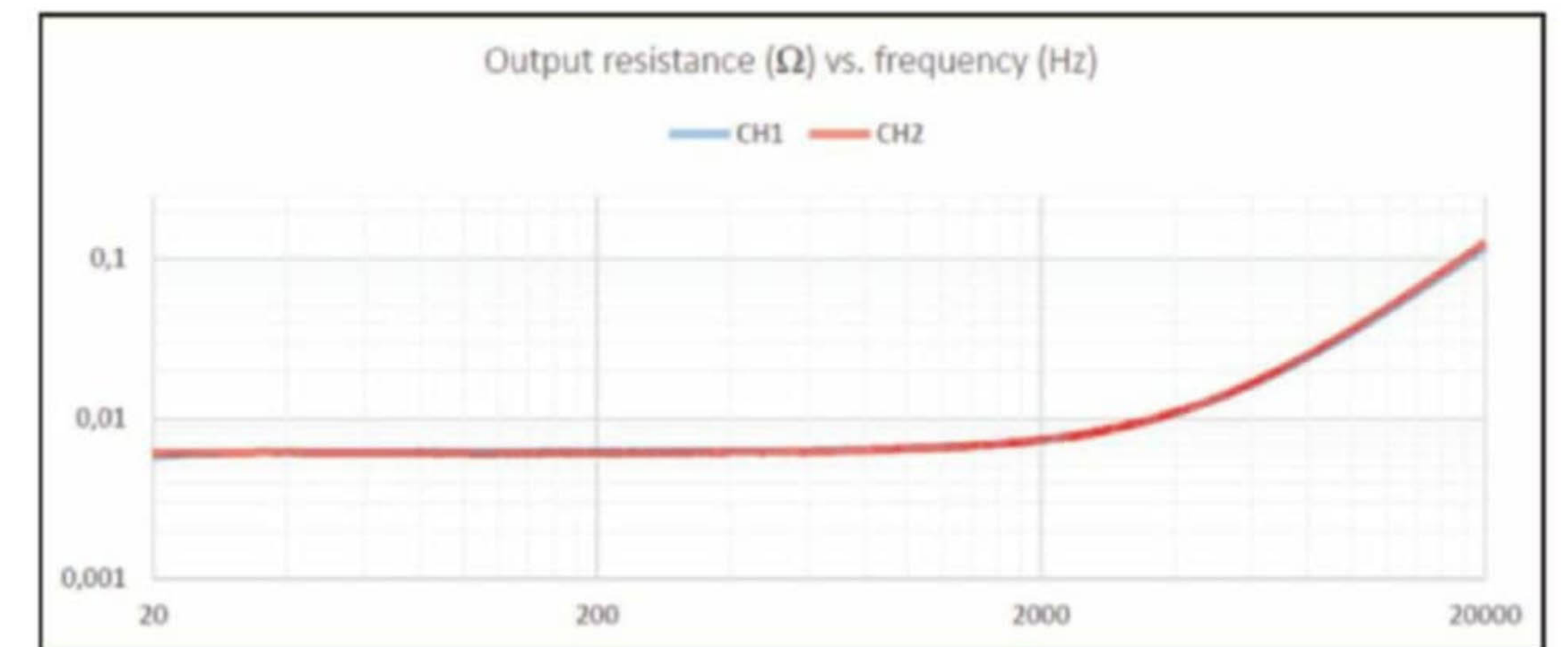


Figure 8: Output resistance (Ω) vs. Frequency (Hz) of Pascal's L-PRO2S module

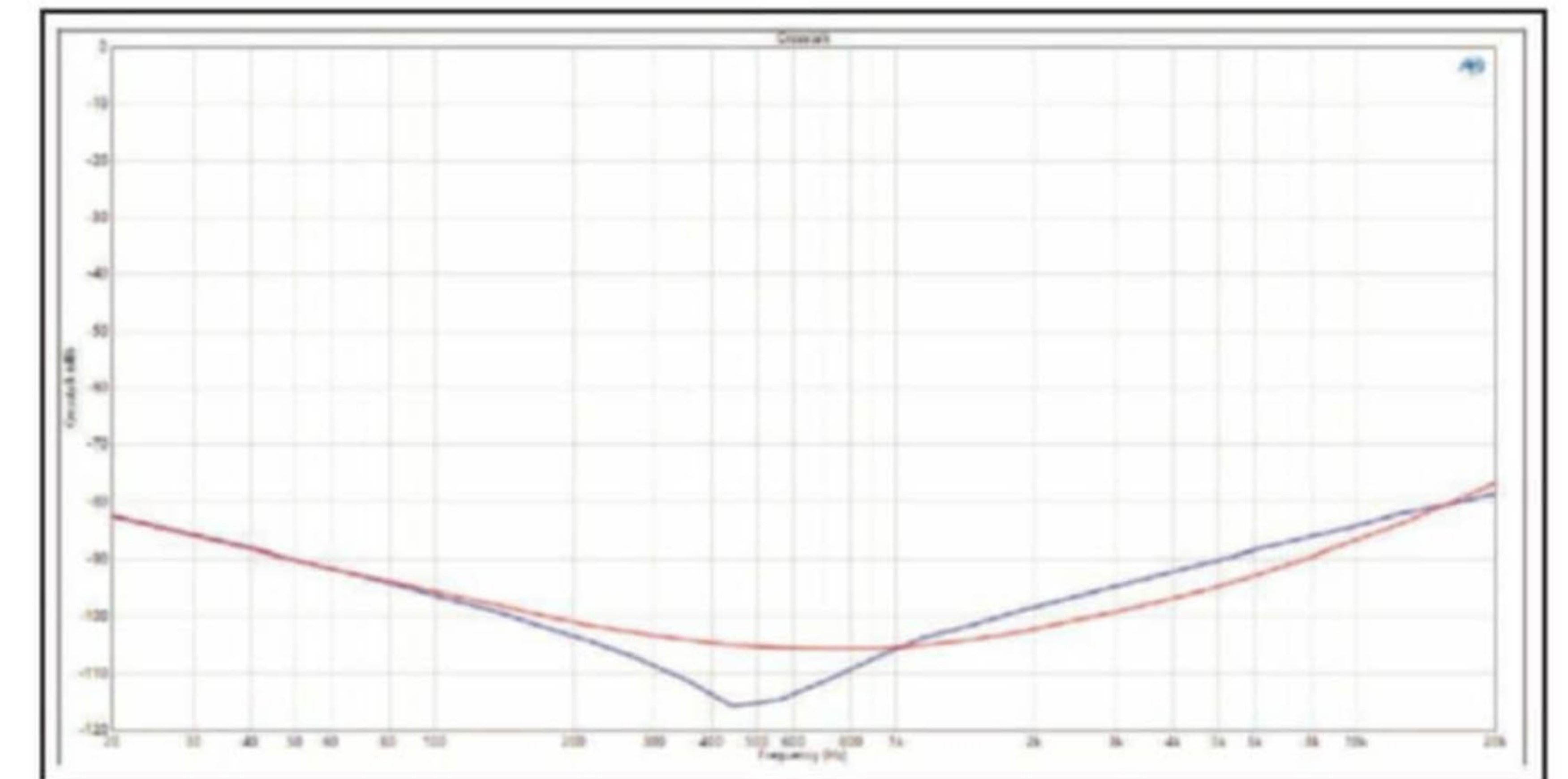
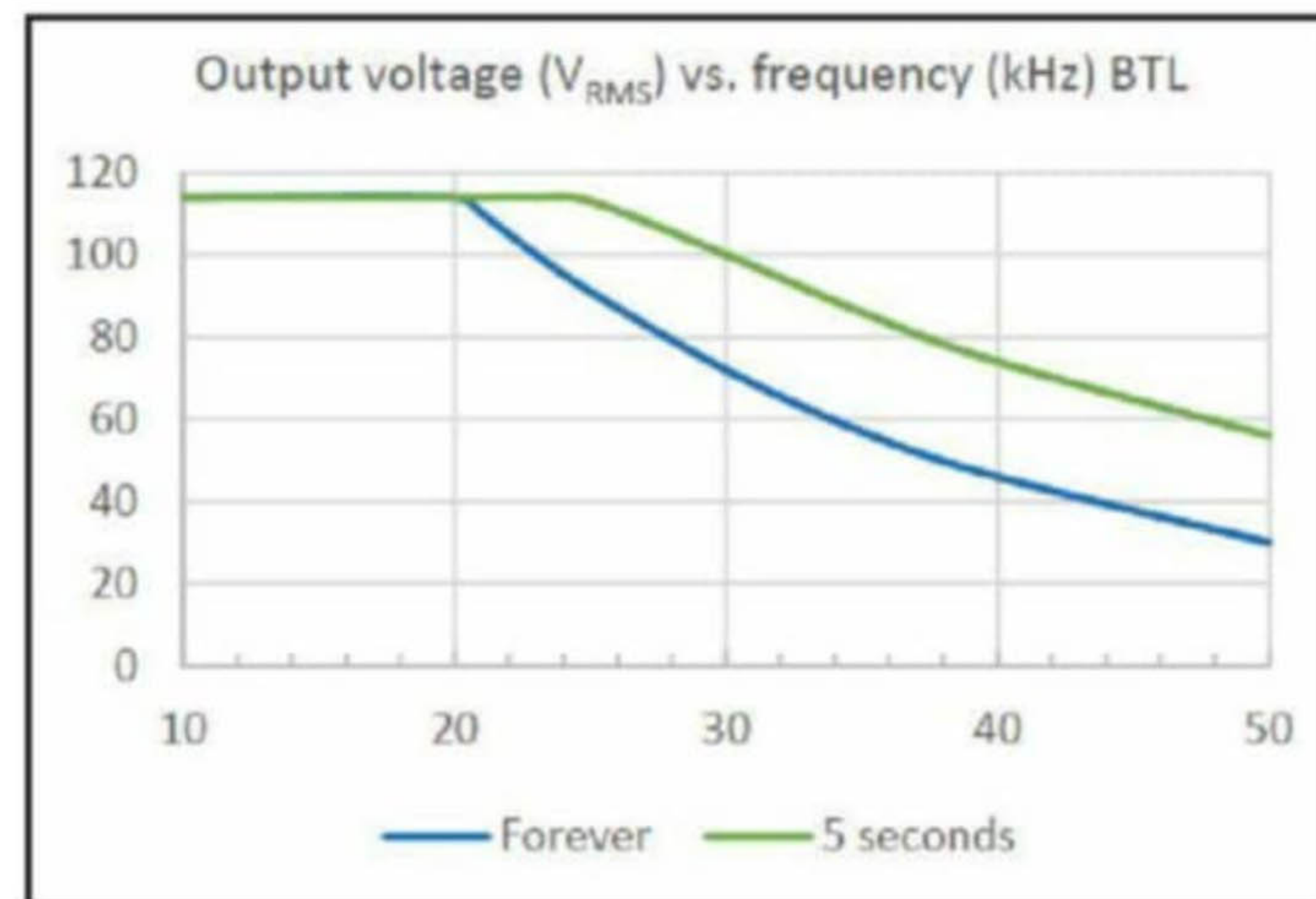


Figure 9: Crosstalk Channel 1 at P0 Channel 2 = 50 W 8 Ω (blue), Channel 2 at P0 Channel 1 = 50 W 8 Ω (red) of Pascal's L-PRO2S module

About the Author

Ward Maas is the owner of Pilgham Audio. He studied electronics, marketing, and amplifier design. During his career in consumer electronics, Ward worked in areas ranging from CD standardization to radio and television to personal GPS navigation. Ward has worked on an extreme low-noise magnetic cartridge preamplifier and several special amplifier products. As the CTO of "Witchworld," a theme park near Amsterdam, he also works with animatronics. He lives in Almere, Netherlands, with his wife and son.

Figure 10: Pascal L-PRO2S maximum Vout vs. Frequency vs. Time (BTL mode)



impressive. The unit is working (guaranteed) from 85 VAC to 265 VAC at 45 to 65 Hz. As mentioned before, in the end, the power supply is the limiting factor. From 240 VAC down to 120 VAC, the unit is delivering its 1500 W. Only at 100 VAC the output power is 1300 W. **Figure 10** confirms that Pascal knows what its amplifier is doing. Conclusion: this is a very, very rugged amplifier.

The L-PRO2S is safety tested according IEC/EN 60065:2001(7th E) + A1:2005 + A2:2010, IEC/EN/UL 62368-1:2014 (2nd E) fulfills the requirements of EN 60065:2002 + A1:2006 + A11:2008 + A2:2010 + A12:2011, EN 62368-1:2014, CSA C22.2 NO. 62368-1-14:2014, UL 62368:2014. EMI compliance was tested according to emission and immunity: EN 55032:2012 with EN 55032:2012 AC 2013, EN 61000-3-2:2014, EN 61000-3-3:2013, FCC part 15 subpart B, EN 55103-

2:2009, EN 55035:2017. Yes, this is a long list. Yes, both safety and electromagnetic interference (EMI) compliancy under so many different circumstances is no mean feat.

If you would like to incorporate a Pascal product in an amplifier cabinet or use it in a self-powered speaker, you would probably want a solution that is flexible, adaptable, and one where your initial investment goes a long way. One of the key features here is the interconnect ability to a single front-end I/O/DSP board design. Plug-and-play connectivity is achieved via the 26-pin interconnect that is standard across the Pascal amplifier module range. A single front-end module can be used in multiple self-powered loudspeaker and rack amplifier design configurations. This means less engineering work across multiple product design projects.

On this 26-pin connector you will find control pins like Mute, Standby, Signal present, Signal TimeOut, and Temp/VAC/Low rail mode. With these you can set the behavior of the unit depending on several situations. The L-PRO2S has various readouts to monitor the state of the module such as temperature, mains voltage, amplifier output voltage, and clipping read out and a disable/protect output.

The L-PRO2S also has all the relevant protection features built in for temperature, over current, DC protection, and high-frequency protection—both to protect the amplifier module against abuse/extreme use scenarios, and to protect the speaker drivers from being damaged in case of a malfunction. With these standard functionalities accessible for DSP and network control implementations and I/O stages, it is easy to tailor fit this unit to the required needs. Better yet, due to the standardization you can reuse the efforts done for other Pascal products and even better, use them in your next product as well. The built-in auxiliary power supply comes in handy there. Pascal also works closely with providers of proven high-performance signal processing technologies, making advanced complementary DSP solutions for its amplifier modules readily available.

With this way of working, Pascal is not only very able to support a manufacturer with its knowledge of the amplifier, but also with the implementation knowledge, both in terms of extra hardware and additional software needs. The icing on the cake is the help Pascal can provide with EMI/EMC performance measurements.

All these elements together will result in a very predictable product development at very predictable costs and a faster time to market. During my visit Peter Frentz and Senior Application Engineer Thomas Taul (see **Photo 7**) served as my



Photo 8 a-d: The company's research and development facilities (a-c) are an optimal environment not only for creating new products and technologies but also for supporting the company's customer implementations and designs featuring Pascal's power modules; d: Customer support at work with a multi-channel amplifier that uses Pascal modules.



Photo 7: Peter Frentz and Senior Application Engineer Thomas Taul were our guides during *audioXpress*' visit to the facilities.

tour guides. **Photo 8** shows R&D and customer support/application engineering at work.

On November 26, 2018, Denmark's Minister of Industry, Business and Financial Affairs visited Pascal's HQ in Copenhagen (see **Photo 9**). The purpose of Mr. Rasmus Jarlov's visit, which received television coverage, was to learn first-hand about the company's remarkable commercial success and rapid rate of growth. The Minister was especially interested in the specific aspects of Pascal's business that have engendered its rapid success in becoming a global market leader. "Pascal's dedicated focus on a specific market segment and its ambition to be number one is very inspiring," he said. "Not to be satisfied until you are the best in your market space globally is key to your success."



Author's Note: I would like to thank Peter Frentz and Thomas Taul (shown in Photo 5) for their time and the opportunity to visit the Pascal premises.



Photo 9: Ministerial visit: (L-R): Pascal CEO Lars Fenger, Danish Minister of Industry Rasmus Jarlov, and Business and Financial Affairs and Senior VP Business Development Peter Frentz

How to Design and Specify Loudspeaker Drivers

Our longtime columnist Richard Honeycutt shares his insights on Geoff Hill's recently released book, *Loudspeaker Modelling and Design: A Practical Introduction*, which demonstrates modern software and hardware being applied to the processes behind loudspeaker design and modeling.

Review By
Richard Honeycutt
(United States)

Ever since the introduction of the dynamic loudspeaker by Edward Kellogg and Chester Rice in 1925, hobbyists and professionals have been interested in building these electroacoustic marvels. In order to work optimally, loudspeakers require some sort of enclosure. These were fair game for hobbyists, as experimentation, or building a finished product, only required competence in woodworking. Not so with the driver—the loudspeaker itself. By far the greatest focus among speaker-loving amateurs has been enclosure design: open-back, sealed, ported, horns, and various combinations.

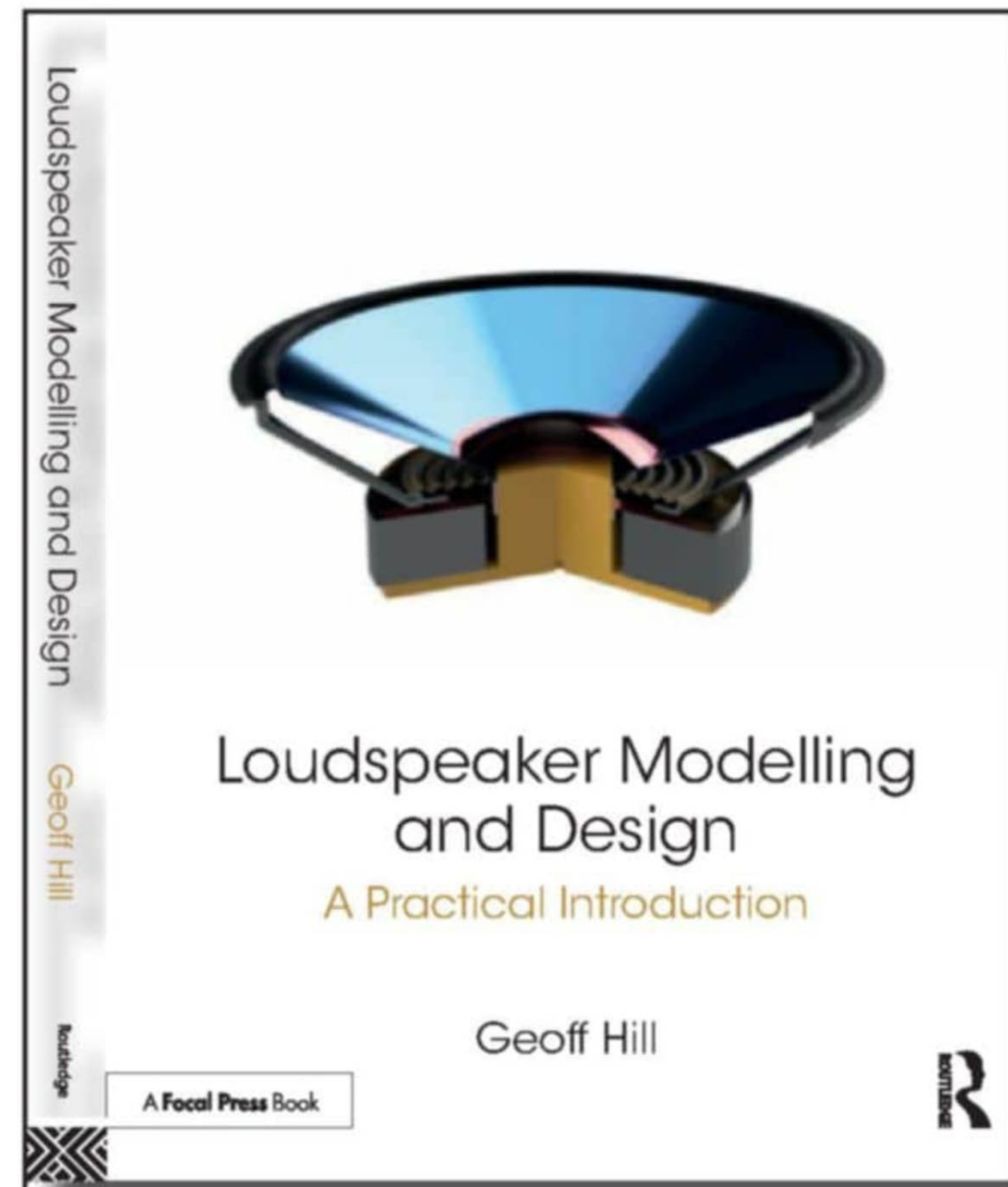
For many years, information on drivers was available only in patent documents or the journals of professional societies. Even when information appeared in books such as Harry Olson's *Acoustical Engineering* or Leo Beranek's *Acoustics*, it was largely theoretical. Olson included illustrations and discussion of the effect of cone profile, and both books covered

the theoretical aspects of magnetic field strength and directionality's dependence upon cone diameter. Electro-mechano-acoustical equivalents of speaker behavior permitted physicists and mathematicians to investigate the behavior of highly simplified models—within certain limits. But driver design was largely the domain of professionals, and was as much an art as a science, passed along by mentors.

In the latter decades of the twentieth century, new computer-modeling tools became available for predicting the actual performance of loudspeaker drivers. These tools included Finite Element Analysis (FEA), the Boundary-Element Method (BEM), 3-D modeling, and other methodologies.

Engineer/Author Geoff Hill has spent much of his career employing these state-of-the-art technologies in designing loudspeakers, and his latest book concisely presents the how-to's he has learned and employed as an engineer for Goodmans and other speaker manufacturing companies, and as a consultant at Hill Acoustics.

Loudspeaker Modelling and Design discusses exactly what the title advertises. It is intentionally very light on theory and math, and does not cover manufacturing methods at all. Thus, it is primarily of interest to budding loudspeaker engineers. It will appeal to those amateurs who just want to understand the process of designing a loudspeaker using up-to-



the-minute tools. Even DIY-ers who have access to a fully equipped machine shop will not be able to “roll their own” drivers using the information in this book, because of unavailability of parts—cones, voice coils, surrounds, magnets, pole pieces, top plates, baskets, etc.—in small quantities.

For the professional engineer just starting to engage in speaker design, the book can act almost as a mentor, starting with creating a project schedule and a bill of materials. From there, it guides the reader through the process of modeling a speaker (with emphasis on the driver, not the enclosure), and simulating the magnetic, mechanical, and acoustical behavior of the unit as each subsystem is designed. Hill illustrates the design/modeling/simulation process using a number of software packages, many of which are available as shareware.

The book is laid out in nine parts. The first four, comprising 15 chapters, provide an introduction, basic theory, the concept of modeling, and a quick run-through of the design process. The five chapters of Part 5 cover the inner workings of a speaker driver. Part 6 uses seven chapters to introduce the use of FEA, BEM, and integrated modeling methods. Parts 7 and 8 (three chapters each) discuss mechanical design and the science of measuring a loudspeaker, respectively. Part 9 includes appendices C through K and M, which demonstrate the use of the various software packages employed in modern loudspeaker development. Appendix A is a glossary, and B is a reference list of books and websites for learning about loudspeakers. Appendix L covers statistical analysis, an essential topic for a designer of loudspeaker drivers for manufacture.

Before diving into *Loudspeaker Modelling and Design*, a novice, even one having a BS or equivalent in Electrical Engineering, will benefit from familiarizing him/herself with the background information found in such books as Olsen's and Beranek's volumes mentioned earlier, as well as *Loudspeaker and Headphone Handbook* (ed. By John Borwick) and Glen Ballou's *Electroacoustic Devices: Microphones and Loudspeakers*, plus books on loudspeaker system integration such as Vance Dickason's *Loudspeaker Design Cookbook* to gain an understanding of the variables at play in loudspeaker operation and optimization. Membership in the Audio Engineering Society (<http://www.aes.org/membership/>) provides access to a mind-boggling array of original papers on loudspeaker-related subjects.

But Geoff Hill's book fills a unique niche in providing insight into the methods actually used in designing loudspeaker drivers. If you're interested in loudspeakers, you should certainly consider buying a copy!



Wolfgang Klippel meets with Geoff Hill (at right) and discusses Hill's new book, *Loudspeaker Modelling and Design*, at the 2018 Audio Engineering Society convention in New York.

Loudspeaker Modelling and Design: A Practical Introduction
By Geoff Hill
Publisher: Routledge; 1st edition (October 10, 2018)
Paperback: 390 pages
ISBN-10: 0815361335
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Assessing the Impact of Device Enclosure Materials

By
John Quigley
(Application Development Scientist,
Eastman Chemical Co.)

In this article for *audioXpress*, John Quigley discusses how polymers might improve acoustic performance in audio applications and reveals findings from multiple tests of specialty plastics, including copolyester and cellulosic resins used for in-earphone design.

Audio products impact our lives in ways that were unimaginable just a few years ago—integrating seamlessly into our home theaters, audio streaming, and a growing list of smart speakers and other voice-enabled devices. While advances in technology and applications continue to increase user convenience and enjoyment, material selection has often not kept pace.

As the market demands the latest innovative application, users also demand an ever-higher level of sound quality. The convergence of these demands leads manufacturers to look for any distinction that can create a performance advantage in the marketplace.

Global specialty plastics provider Eastman collaborated with respected design and engineering firms to assess how material selection can impact the audio performance of audio devices. Representatives from Eastman and DW Designs presented their test methods and results at the annual Rocky Mountain International Audio Fest (CanJam) in October 2018. Eastman has also presented these test results in a recent webinar (see Resources).

Based on our knowledge of the unique vibrational damping of our resins, our hypothesis was that

materials with desirable viscoelastic response could improve the acoustic performance of audio devices. We developed testing methods to quantify inherent vibration damping properties in candidate materials. Further testing of fully molded in-ear monitors (earbuds) compared cumulative spectral decay (CSD) and total harmonic distortion (THD) along with an expert panel of listeners to demonstrate the role of housing material on resonance, distortion, and listener experience.

Evaluating the Damping Properties of Materials

The vibration damping properties of Eastman materials have been demonstrated at a wide range of frequencies in previous works (see Resources). The current testing applies similar recognized research methods to compare damping loss factors for Eastman Tritan copolyester, Eastman Trêva engineering bioplastics, polycarbonate (PC), and acrylonitrile butadiene styrene (ABS) resins under controlled conditions, geometries, and temperatures.

Collaborating with Kolona & Saha Engineers, a center point impedance (CPI) method was used to compare how the vibration damping properties of test

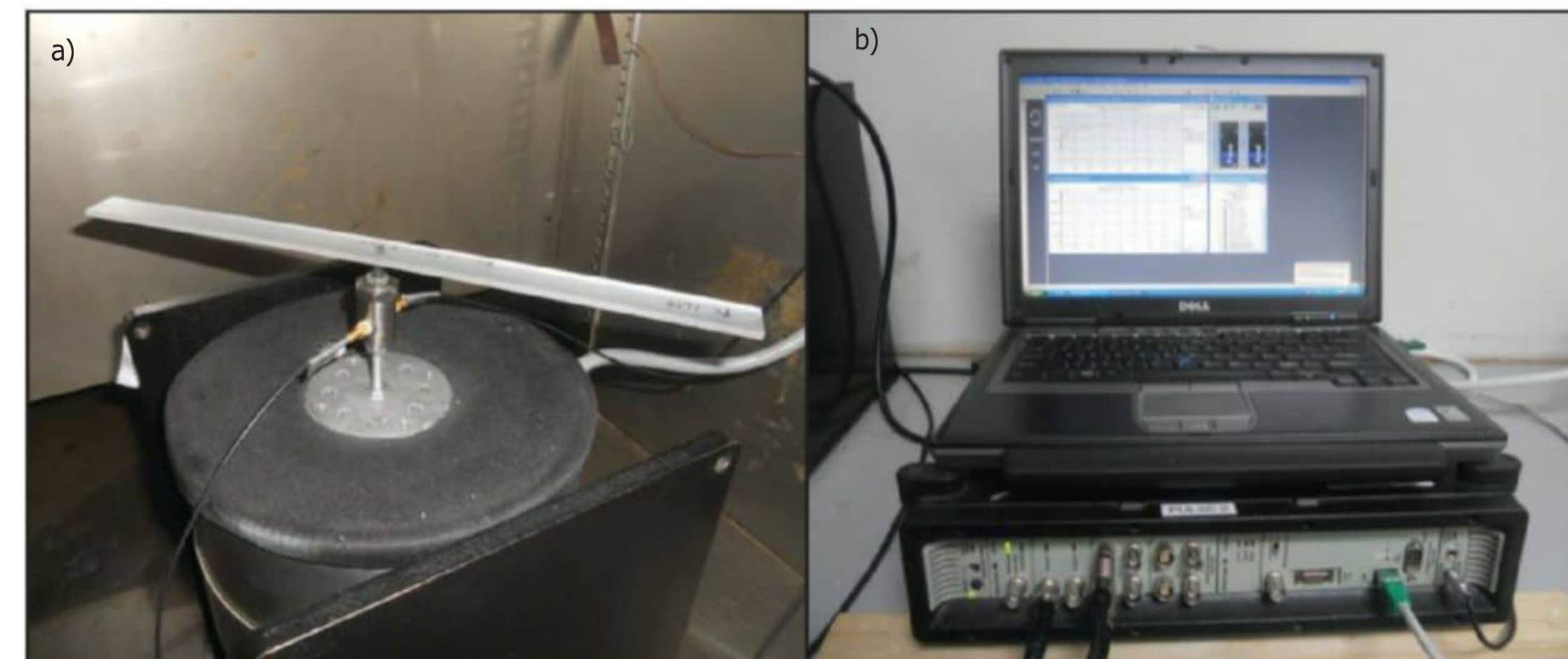


Photo 1: Center point impedance (CPI) was used following the SAE J3130 standard. The rectangular bar is vibrated at the center, and the accelerometer and the impedance head measure force and velocity as vibration travels through the part (a). The CPI testing equipment includes the impedance head/accelerometer and the data acquisition frequency software generator (b). (Images courtesy of Kolona & Saha Engineers, which collaborated on all center point impedance testing)

materials can affect acoustic performance—based on a working SAE standard (SAE J3130). The CPI testing employed a frequency generator, an accelerometer, an impedance head, and data collection software (see **Photo 1**).

In tests conducted at room temperature, the frequency generator created vibration at the center of rectangular test bars. An accelerometer measured the motion of each bar as the input vibration traveled through the part. The magnitude of this input was measured via the impedance head, measuring only longitudinal modes. Damping loss factor, often expressed as $\tan(\delta)$, was determined from vibrational response at discrete resonance modes. Frequency vs. force/velocity was plotted as the frequency response function (FRF).

FRF, as measured by CPI, allows the comparison of vibrational damping capabilities of different materials in a similar geometry by analyzing their resonance modes. **Figure 1** plots the FRF curves for the four test materials.

The damping loss factor (DLF) was calculated for each resonance peak. **Figure 2** uses the fourth resonance mode to illustrate how this calculation is performed. Mathematically, the loss factor is stated as $\eta_c = \Delta f_n / f_n$.

Using these calculated values, researchers can make direct comparisons between the different test materials as long as they are molded with the same geometry.

For a broader context, researchers gathered CPI data over extended frequencies—beyond the frequency ranges relevant to consumer audio applications (see **Figure 3**). Damping loss factor,

as calculated from frequency response function, provides valuable information for determining the vibrational damping capabilities of different materials.

Damping loss factors represented in Figure 2 are consistent with comparisons at frequencies that are higher than those relevant for audio applications shown in Figure 3. Eastman copolyester (Tritan) and cellulosic (Trêva) demonstrate improved damping properties compared with PC and ABS over a wide range of frequencies. Differences in loss factor indicate the potential for superior performance in

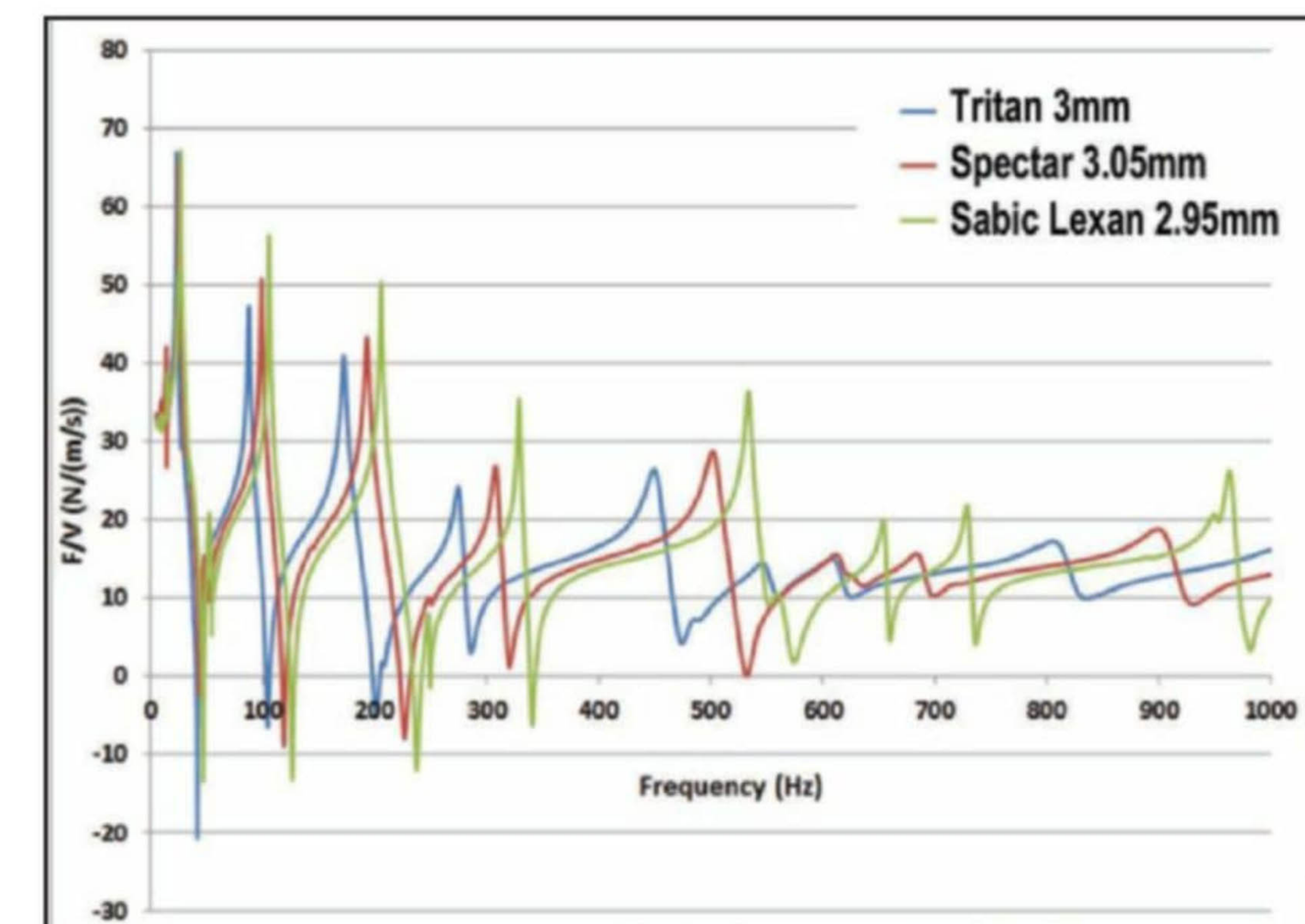


Figure 1. The frequency response function (FRF) was plotted for the four materials. Pronounced modal resonance peaks indicate less damping value. (Image courtesy of Kolona & Saha Engineers)

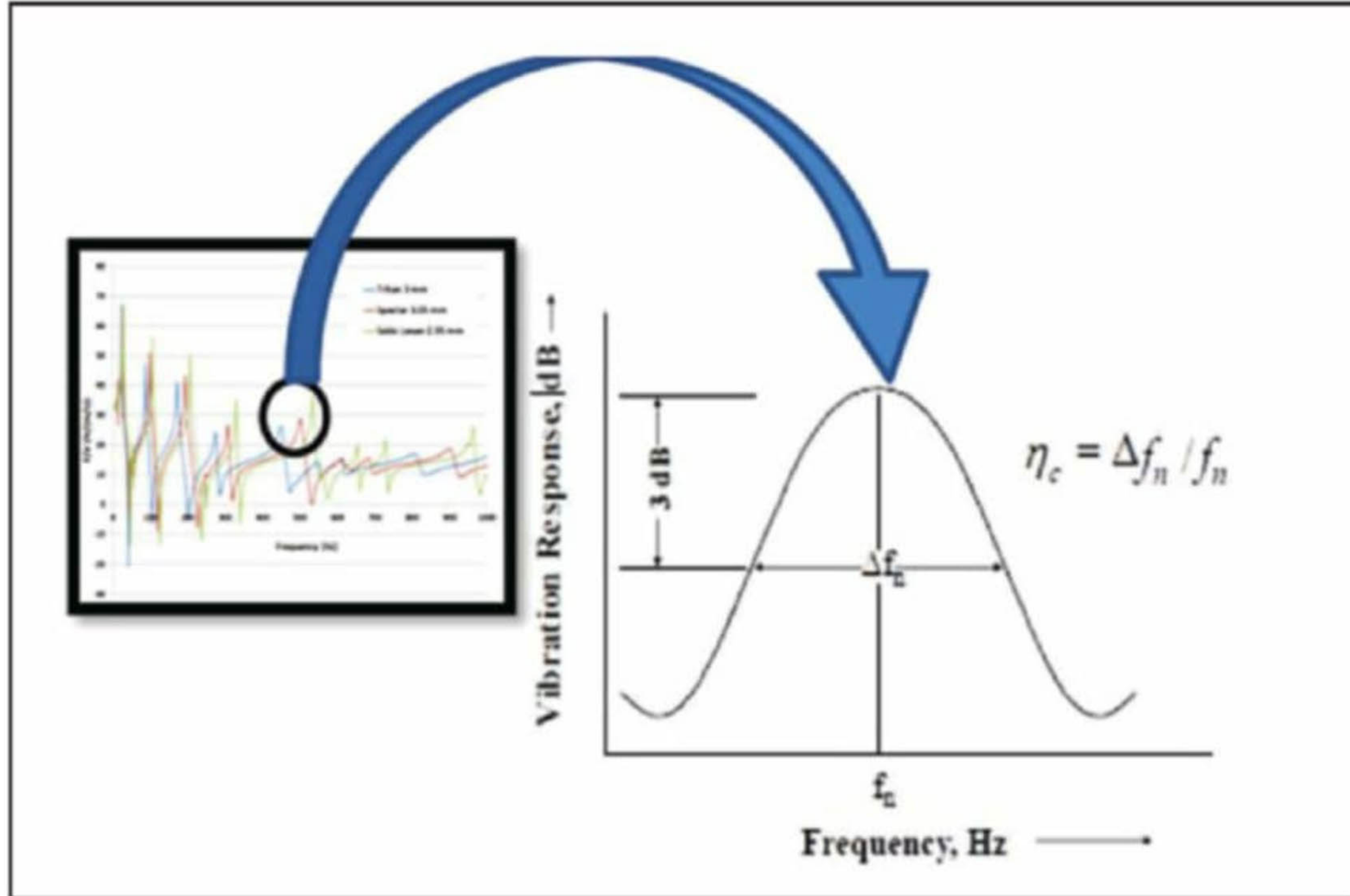


Figure 2: Peak width is determined 3 dB below the maximum (r_{f_n}) and divided by the resonance frequency (f_n). This gives the damping loss factor (η_c) at that discrete resonance. (Image courtesy of Kolona & Saha Engineers)

audio applications that require reduced resonance, less distortion, and improved acoustical performance.

Evaluating Performance in a Fully Molded Device

Eastman collaborated with a second external partner to evaluate the audio performance of polymers when fully molded and assembled in device enclosures. Working with engineers at DW Designs, Eastman evaluated test housings of Periodic Audio Be (beryllium) in-ear monitors (IEM), shown in **Photo 2**. The housings were molded with one of three materials: the incumbent PC, Eastman copolyester (Tritan), or Eastman cellulosic (Trêva) resins.

To determine the housing material's role on the audio quality of the test IEMs, researchers measured acoustic performance based on cumulative spectral decay (or CSD, presented as waterfall plots), total

harmonic distortion (THD), and subjective listening tests by an expert panel. This particular IEM model provides a valuable benchmark for assessing enclosure materials. The Periodic Audio Be model with incumbent polycarbonate (PC) has been rated the "lowest THD IEM on the market" by third-party tests (1% THD at 1 mW).

If you've worked with CSD waterfall plots, you know they present FRF data with the added dimension of time (Z-axis). The wavelets on the Z-axis graphically show how the cycles decay after the input signal has stopped. The wavelet-based plots improve resolution at higher frequencies because of how the time effects are calculated.

For CSD testing, IEMs were connected via a fixture to a microphone directed with a 3D printed fixture measuring 20 mm wide and designed to mimic the human ear canal. To eliminate noise floor and reduce ambient sound, an FM slide was emitted by the transducer to the microphone at 110 dB SPL nominal. The microphone set to a 50-dB range measured the response up to 40 kHz. This required two times frequency generation to accurately measure response in the desired region.

The results of CSD testing are presented in the waterfall plots shown in **Figure 4** where:

- Polycarbonate housing provided a benchmark, with ~20 cycles of resonance at 6 kHz and 14 kHz—the first decaying quickly; the second, less quickly.
- Copolyester housing experienced a more energetic initial transducer response than PC—with a similar ridge at 6 kHz and a very rapidly decaying ridge at 14 kHz.
- Cellulosic housing experienced a 6 kHz ridge that decayed noticeably faster than either of the others; the 14 kHz ridge was almost nonexistent. Acoustically, Trêva is a very dead material.

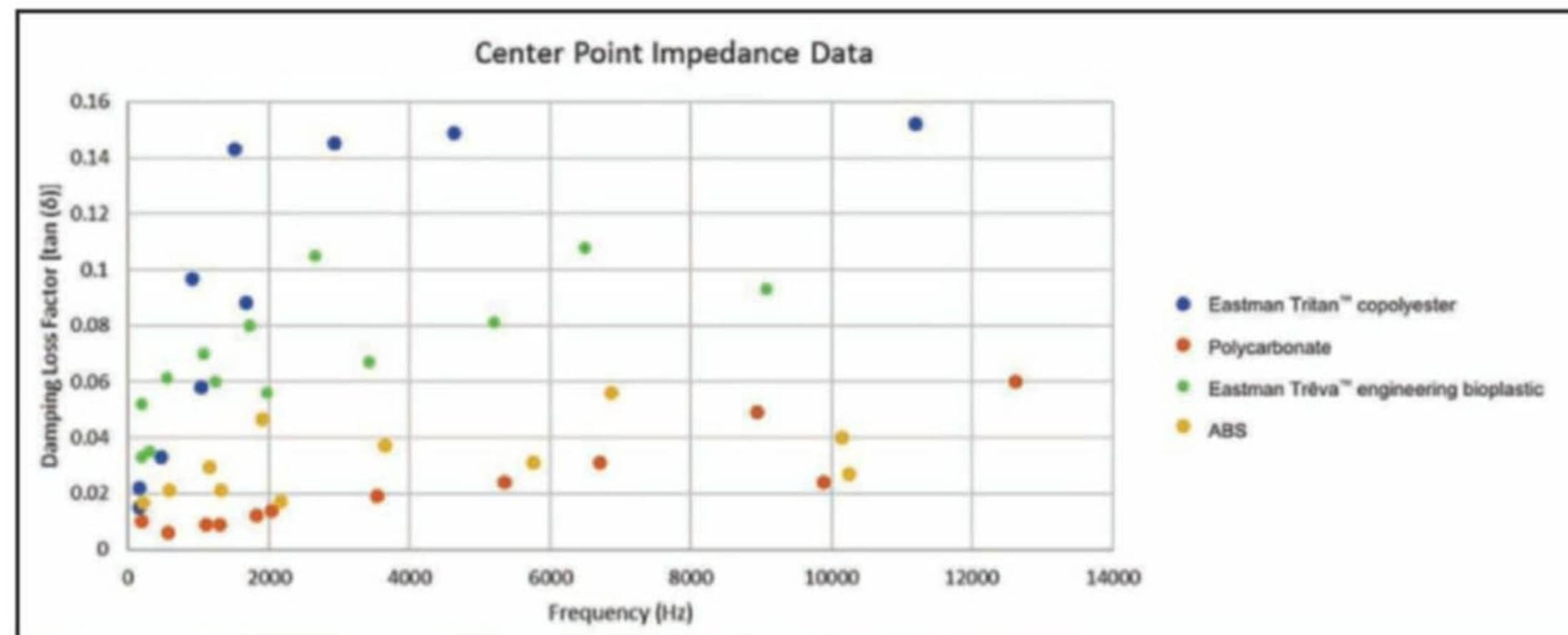


Figure 3: Results in Figure 1 are consistent with testing beyond consumer device applications. (Image courtesy of Kolona & Saha Engineers)

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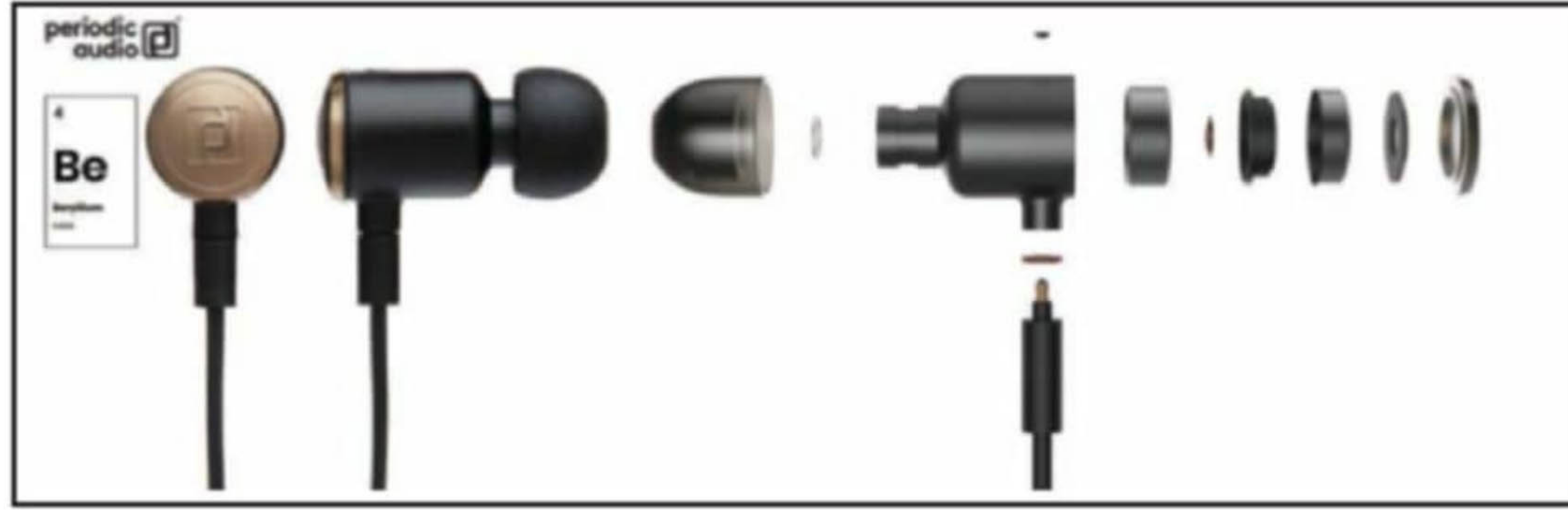


Photo 2. Periodic Audio Be (beryllium) in-ear monitor used for testing. (Image courtesy of Periodic Audio)

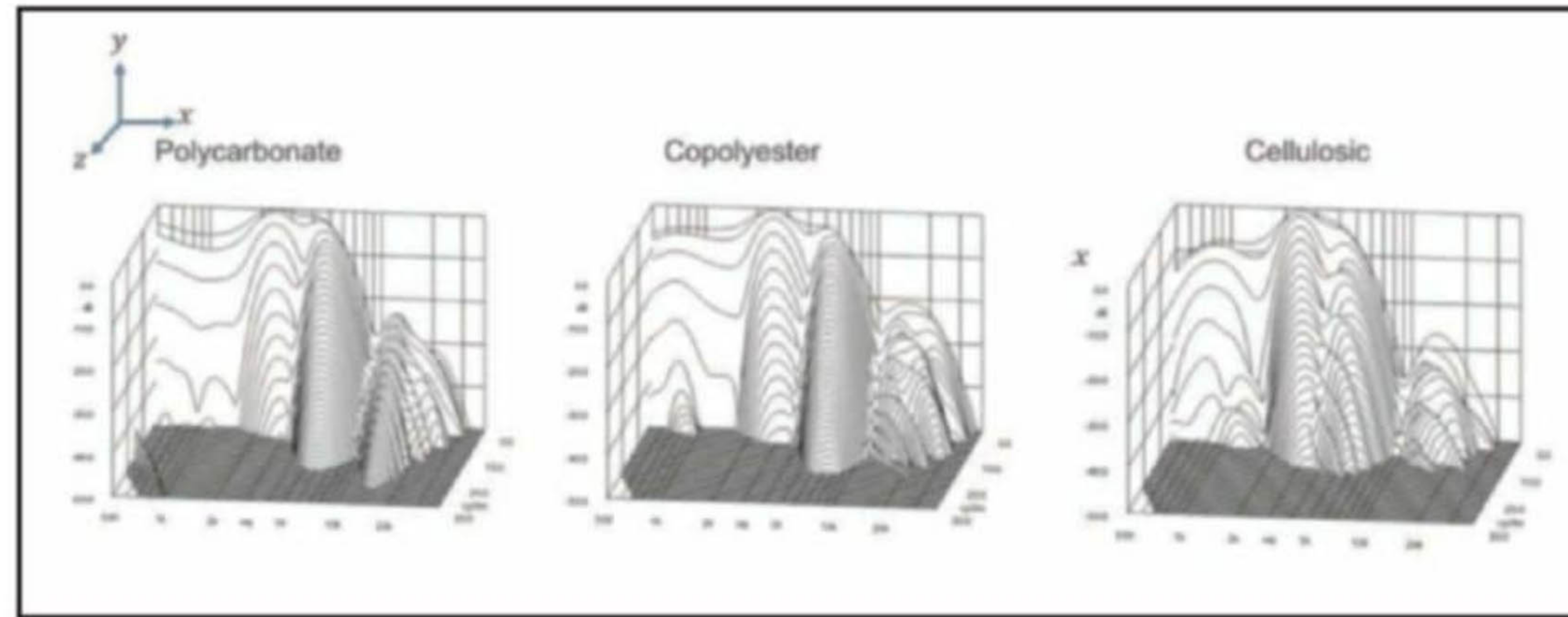


Figure 4: Incumbent polycarbonate housing provides a good benchmark. The limited number of ridges with resonance of ~20 cycles decay smoothly (a). Copolyester housing shows similar response through 6 kHz, but ridge at 14 kHz decays very rapidly (b). Cellulosics housing shows significantly reduced ridge at 6 kHz and virtually nothing at 14 kHz. Acoustically, it is a very dead material (c). (Image courtesy of Periodic Audio)

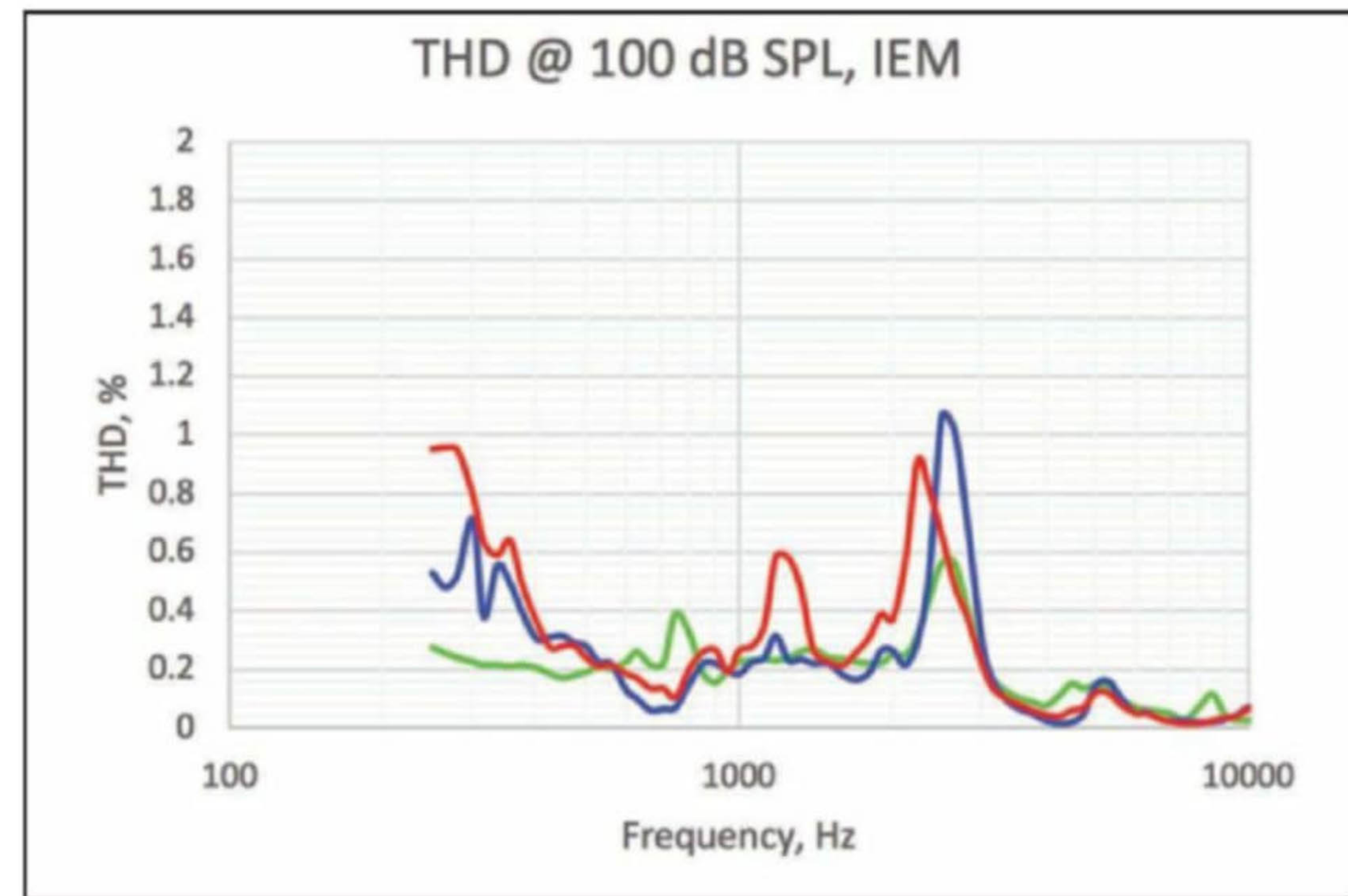


Figure 5. In this graph, PC shows a distortion peak (>0.5% THD) at 1250 Hz. PC and copolyester show pronounced peaks (>0.8%) at ~2500 Hz. (Image courtesy of Periodic Audio)

Compared with PC, both Tritan and Trēva provide less of their own audio signature to interfere with the listener’s audio experience.

For THD testing, researchers ran sweeps for each IEM at 100 dB SPL nominal. Results plotted in **Figure 5** reveal that PC experienced the most pronounced distortion peaks. Tritan had fewer peaks (although there is one large primary peak at 2500 Hz), while demonstrating better top-end extension than PC. Trēva performed the best of the three materials with the lowest overall level of measured THD.

To translate these results to devices with higher THD, we would expect improvements related to proportional gains rather than absolute magnitude differences. That is, a 30% improvement here could represent a much larger improvement on a product that has a larger initial THD value than that of the Periodic Audio Be IEM.

Compared with PC, both Tritan and Trēva provide measurable improvements in THD, with Trēva providing superior performance at nearly all frequencies.

Hearing the Difference

Results presented in the featured CPI and molded IEM tests established that differences in audio performance of enclosure materials can be measured. But can they be heard?

Eastman conducted subjective listener testing to determine whether changing enclosure material can provide a discernible benefit for audio engineers, designers, and consumers. A panel of three listeners with audio expertise and familiarity with the Periodic Audio Be IEM experienced 15 demo tracks selected from a wide range of music styles. Tracks were chosen for characteristics that highlight potential issues with the IEMs. Music was streamed through a SONOS Connect:Amp with optical audio cables using a Channel Islands Audio VDA•2 D/A Converter and a Channel Islands Audio VHP•2 Headphone Amplifier.

For expert listeners, the differences between the IEMs were “subtle but audible and consistent.” Other interesting comments noted differences such as:

- Reduction of sibilance
- More “space”
- Resonance is toned down
- Sound is more full and overwhelming
- Reduction of “buzziness”
- More even amplitude

The majority of improved listener experiences favoring copolyester and cellulosics correlated with improved CSD plotted in Figure 4 and reduced THD described in Figure 5.

Innovating with a Material Difference

Innovative applications and greater IoT integration will continue to raise the bar for audio performance. Acoustic engineers, material scientists, mechanical engineers, and designers will be tasked with improving the integrity of voice input as well as audio output.

Eastman looks forward to building on the research, beginning with understanding other applications where vibration damping materials could enhance audio performance and provide value to its customers. A few of the working hypotheses include:

- Reducing resonance of cabinets and enclosures
- Improving active noise cancellation
- Preventing feedback and feedforward systems from going unstable by reducing resonance and interference contributed by housing
- Reducing load on digital signal processing (DSP)
- Reducing computational load by removing resonances, leading to longer battery life
- Reducing computing power of PCB to mitigate cost
- Increasing efficiency of speaker/microphone systems by reducing echo
- Allowing higher SPL of speaker systems incorporating microphones by reducing echo and reverberation

Every new innovation and application in this rapidly growing market involves the challenge of reevaluating designs. Each reevaluation can be welcomed as an ideal opportunity to consider how different materials can improve device performance—and strengthen brand presence in the marketplace.

For more information about Eastman, visit the company’s consumer electronics device housings page at www.eastman.com.

About the Author

John Quigley, Ph.D., is an application development scientist with Eastman Chemical Co. In recent years, his work has focused on understanding the link between engineering thermoplastic viscoelastic properties and performance enhancements for audio components.

Resources

S. Givens and A. Dudal, “How to Reduce Small Appliance Noise Without Reducing Performance,” UL Prospector, 2017, Webinar.

J. P. Quigley and A. Dudal “Using Eastman Plastics to Reduce Unwanted Resonance in Audio Enclosures” UL Prospector, 2018, Webinar.

J. P. Quigley and D. Wiggins, “Eastman and DW Designs showcase audio clarity of Eastman Tritan™ copolyester and Eastman Trēva™ engineering bioplastics,” Rocky Mountain International Audio Fest (CanJam), 2018.

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DPA d:vote 4099 Instrument Microphone

By
Stuart Yaniger
(United States)

In this article, Stuart Yaniger checks out DPA's d:vote 4099, which features the latest DPA CORE technology. The idea behind CORE is to reduce distortion and increase dynamic range. See if this mic measures up to our reviewer's standards.



Photo 1: The DPA d:vote 4099 is a small, light, and versatile microphone for close-miked recording and performance use with acoustic instruments.

Since splitting off from Bruel & Kjaer (the Mercedes Benz of test and measurement microphones) in 1992, DPA has similarly established itself in the musical performance and recording market as a premium choice for top artists and audio engineers. Its range of microphones is extensive, with product offerings that include lavalier, desktop, studio pencil, headset, and hand-held form factors. My own previous experience with DPA's mics includes the 4006, a studio omnidirectional condenser microphone featuring a low noise floor and a clean, uncolored frequency response. It was always a reliable go-to for live recording applications in a spaced omni configuration, and I sorely miss having them in my mic locker.

DPA's mic ranges for live performance have traditionally included hand-held, lavalier, and pencil mics for vocals and acoustic instruments, but with the introduction of the d:vote 4099, DPA has extended its reach into clip-on mic applications for both live performance and recording. The d:vote 4099 features a new interface circuit that DPA calls CORE. The idea behind CORE is to reduce distortion and increase dynamic range.

The Product

The d:vote 4099, which I'll just call the 4099, is available in two versions: the Loud SPL and Extreme SPL (there's no "medium SPL" or "Quiet SPL"). The Loud SPL version will accommodate most instruments, with a rated noise floor of 26 dB A -weighted, 20 Hz to

20 kHz, and a maximum sound pressure level (SPL) of 142 dB at clipping. For a few very high SPL situations (e.g., close miked drums and brass), the Extreme SPL extends the limit to 152 dB, at the sacrifice of a few decibels of noise floor, which is 31 dB A-weighted. Both versions have a hypercardioid polar pattern, which can be useful in performance situations where feedback could be an issue.

Physically, the 4099 is a small and lightweight foam-covered lozenge at the end of a thin gooseneck for maximum flexibility in placement (see **Photo 1**). It is shock-mounted via suspension from two elastomeric arms. DPA offers a range of instrument clips. After clipping onto the instrument, the mic's gooseneck snaps into the clip and the mic is then easily aimed to the desired point. Available clips include upright bass, cello, guitar, mandolin/violin, drums, piano, saxophone/brass, and a Universal for most other woodwinds (see **Photo 2**). There's also a stand-mount option. A color-coded band at the gooseneck's connector identifies the mic as a Loud or Extreme version, a useful bit of foolproofing when multiple mics are being used.

With the 4099, DPA includes a 1.6-m long, 2 mm diameter flexible cable that screws onto the end of the gooseneck and at the other end, screws onto a supplied XLR adapter, which converts phantom power (12 to 48 V) into the correct voltage for biasing the mic's capsule and to power the built-in electronics.

The mic capsule is a rectangular electret (self-

polarized) module with a 4 mm × 5 mm gold-plated mylar diaphragm. Built into the capsule is a field-effect transistor (FET) preamplifier. The geometry and the construction of this capsule are illustrated in one of DPA's YouTube videos (see Resources for a link).

DPA's product manager Rene Mørch tells me that the capsule's electronics, the CORE technology, are responsible for the mic's low distortion. Unlike the typical single FET interface used in the majority of electret mics, DPA uses a multi-transistor circuit that provides a more constant current to the electret and can amplify the signal in a more linear manner. Although Mørch would not provide details of the actual circuit used in the CORE microphones, some possibilities can be inferred from a pair of articles on optimizing FET interfacing to electret mics by analog guru Scott Wurcer (see Resources).

The 4099 is supplied in a small, attractive, fabric-covered hard-shell case with a compartment for storing clips. Also available is a touring kit with four 4099 mics and a customizable assortment of instrument clips, all packed in a Pelti hard case with pouches to hold the accessories.

Along with the mic, DPA sent me instrument clips for guitar, mandolin, and a Universal clip. These clamp onto the respective instruments with a friction fit. In the case of the Universal clip (which I used for my flute), the attachment is via a Velcro strap that can be tightened around the instrument. The strap is long enough to accommodate larger woodwinds—even bassoons.

As befits a microphone likely to be used in performance, the 4099 is quite rugged—dropping a light and well-padded unit like this is unlikely to damage it, and that happened several times during the in-use testing phase of this review. DPA also claims that the 4099 can be immersed in water for several hours without any permanent damage. I did not evaluate this claim, but having spent a lot of time with musicians, I think that imperviousness to liquids (especially those containing ethanol) is a very useful feature.

Measurements

Before use-testing the 4099, I ran a basic set of measurements. The test setup included an Audio Precision APx525 audio analyzer, the APx1701 transducer interface, and a calibrated PCB 376A32 phantom power microphone as a reference. I used a Kali LP6 monitor speaker driven by the balanced analog output of the APx525 as an acoustic source.

I first measured the 4099's sensitivity, which is specified at 6 mV/Pa (1 Pa = 94 dB SPL). This was done by the substitution method (i.e., frequency sweeps were played through the acoustic source with the



Photo 2: The 4099 is available with a variety of clamps for attachment to at least 10 different kinds of instrument. (Photo courtesy of DPA)

calibrated reference microphone positioned about 1 meter away, then the 4099 was substituted in the same physical position as the reference mic). I measured the sensitivity of my particular unit at 7.6 mV/Pa, about 2 dB more sensitive than spec.

Source impedance was measured with the XLR adapter attached and powered by 48 V, and was 94 Ω across most of the audio band. This is slightly better than the 100 Ω specification.

Next, I used the BLANK chamber for noise measurement. The BLANK chamber, described in several of my previous *audioXpress* reviews, is basically a plastic tub filled with a mixture of sand and cat litter, placed on a concrete floor. The microphone is wrapped in a plastic bag and duct tape, then buried in the sand mixture with the lead trailing out of the chamber for electrical connection. This measurement method gives a noise floor of under 5 dB SPL over a 20 kHz bandwidth, A-weighted.

The measured noise spectrum is shown in **Figure 1**. The small peaks at 60 and 120 Hz are power line pickup, which is likely a grounding issue in the measurement setup. Because they are narrowband and very low level, and thus contribute little to the overall noise measurement, I did not spend much time moving grounds around to minimize them. The noise floor follows the A-weighting curve, so there are no significant contributions from either cavity or diaphragm resonances. When integrated over a 20 Hz

About the Author

Stuart Yaniger has been designing and building audio equipment for more than half a century, and currently works as a technical director for a consumer products 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.

to 20 kHz bandwidth and A-weighted, the overall noise was 23 dB SPL, comfortably lower than the specified 26 dBA maximum. This is more than adequate for close-miking applications.

The on-axis frequency response was measured similarly to sensitivity, by placing the 4099 about a meter from a small acoustic source, performing frequency sweeps, then ratioing the output to that of the PCB-calibrated reference microphone. **Figure 2** shows the measured response after gating to remove room reflections.

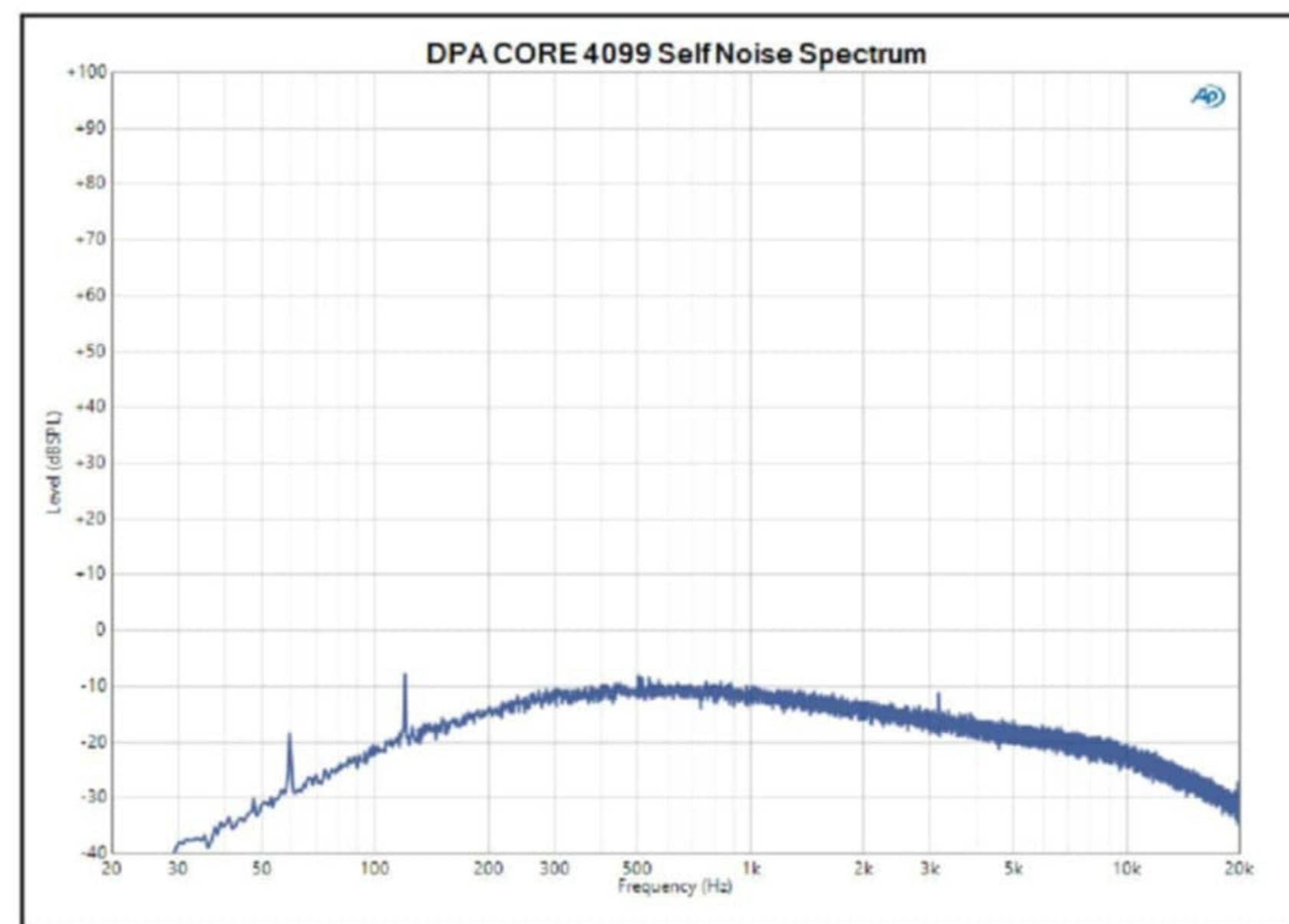


Figure 1: The 4099's self-noise spectrum is low enough for it to comfortably exceed specifications and has no odd resonances. The 60 Hz and 120 Hz peaks are most likely test artifacts and do not significantly affect the overall noise.

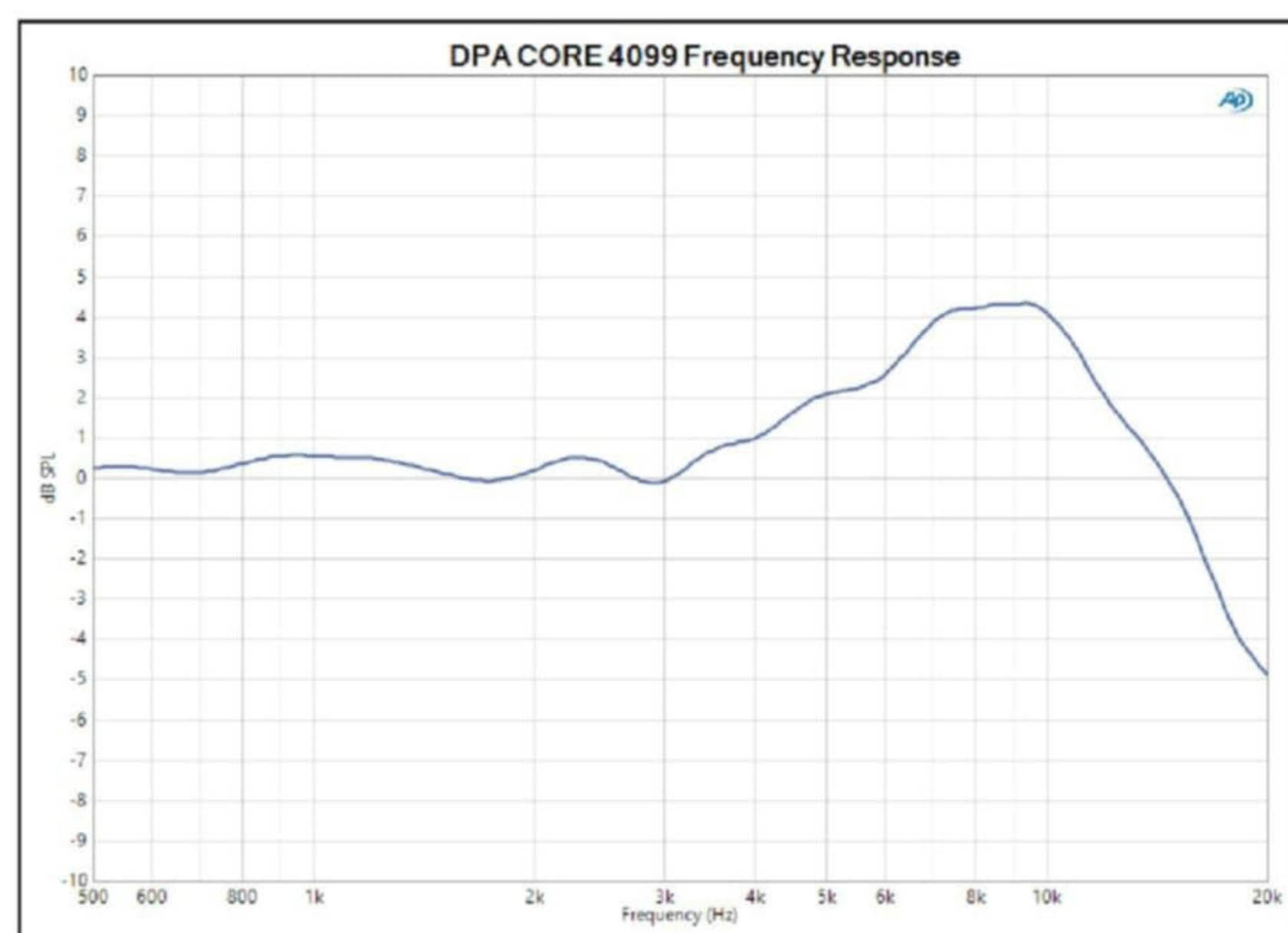


Figure 2: The on-axis frequency response of the 4099 shows a broad rise in the treble, giving the microphone a rather bright sound.

The most unusual feature is the nearly 5 dB rise in the brightness range between 7 and 10 kHz. DPA shows this rise in its frequency response curves and position it as a feature for enhancing the sound of an instrument, however the rise it shows is on the order of 2 dB. I repeated this measurement with a different acoustic source and got an identical result. Perhaps the cause of the difference between DPA's published measurement and mine is the microphone-to-source distance or some other detail in the measurement setup, but in my recordings, I think the subjective result is more consistent with my measurements—the treble rise is not subtle.

To get a sense of the rejection of sound from the rear, I turned the mic to face directly away from the acoustic source, and ran the same gated sweeps as before. The response was then ratioed to the on-axis response from Figure 2. The results show more than 20 dB of rejection in the midrange, and no loss of that rejection in the region where the treble response is elevated on-axis (see **Figure 3**). This is excellent performance.

Since DPA touts the distortion performance of the 4099 from the CORE technology, I set the mic up in the near-field of a 170 mm Peerless woofer and drove the woofer with a 200 Hz sine wave, sweeping the level and measuring distortion as a function of SPL. The results are shown in **Figure 4**.

Note that at 105 dB SPL (the beginning of the sweep), the distortion is well below 0.1%; at the maximum before the test setup protested, 127 dB SPL, the distortion still remained below 0.15%. The distortion performance at high SPL is superb, easily beating the 1% maximum specification. Although I cannot attribute this performance directly to the CORE technology, whatever it is the DPA engineers are doing, they're doing it right.

The 4099 In Use

I enlisted the help of noted Chicago, IL-area multi-instrumentalist LJ Slavin for a live performance and recording testing. We first made a few recordings in my basement studio (aka my test lab) using a Scarlett 2i2 interface at 96 kHz/24-bit settings. I should note that the acoustic treatment in that room is minimal, with ceiling diffusion, a combination of diffusion and absorption on the side walls, and a rather reflective floor. Pickup of room sound in this case is not desirable, and I restrict my room's use to close-miking, so this was a perfect application for the 4099.

Our first instrument for recording was Slavin's flute. The Universal clip held the mic securely in place between the lip plate and the crown, and we bent it around so that it was about 3 cm from the embouchure-hole and pointed down at it at about a 45° angle. Listening to the unprocessed tracks,

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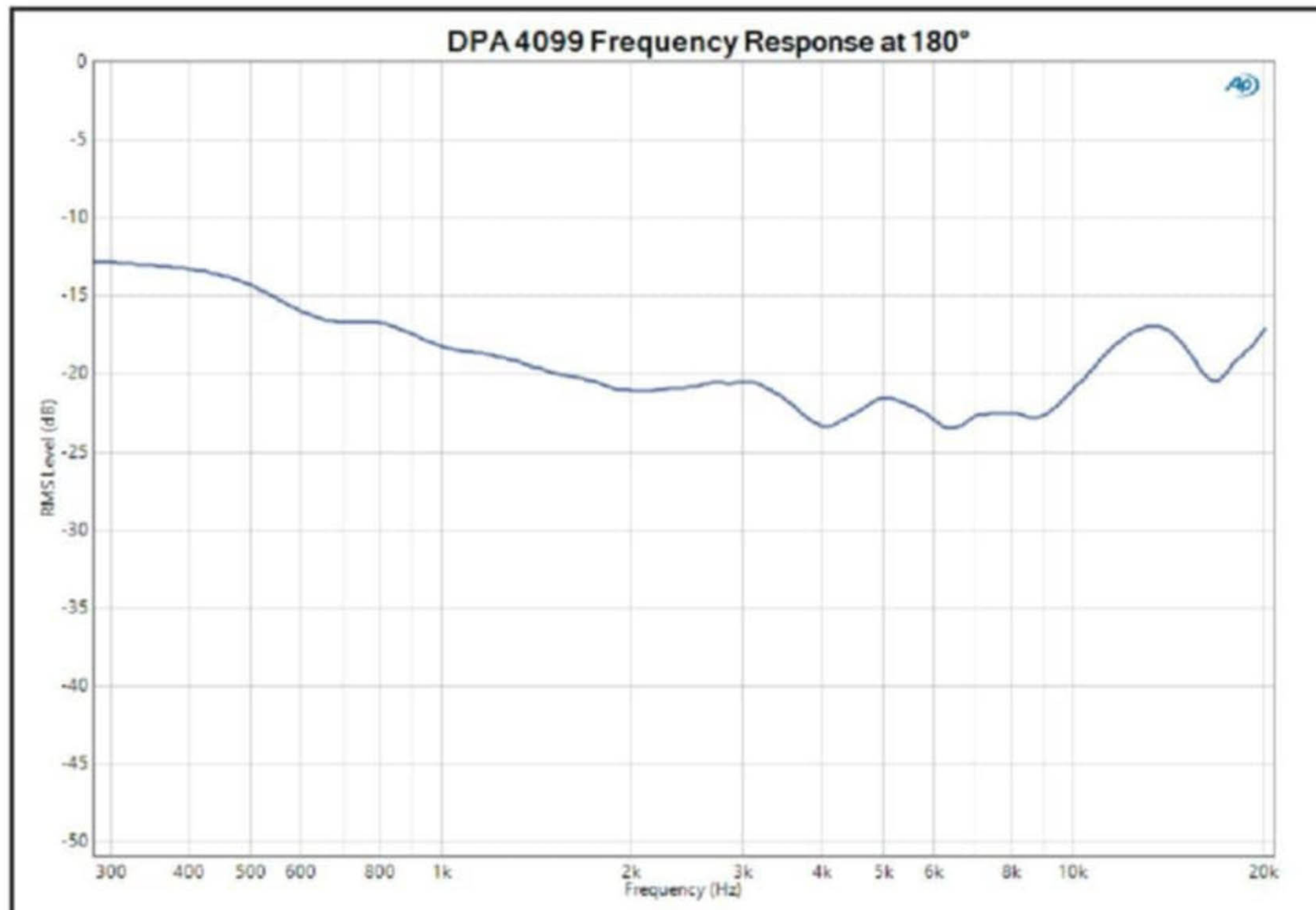


Figure 3. The ratio of the response at the rear of the microphone to the on axis response shows a nearly 20 dB rejection, which can be very effective in removing room sound in close-miking, as well as minimizing feedback.

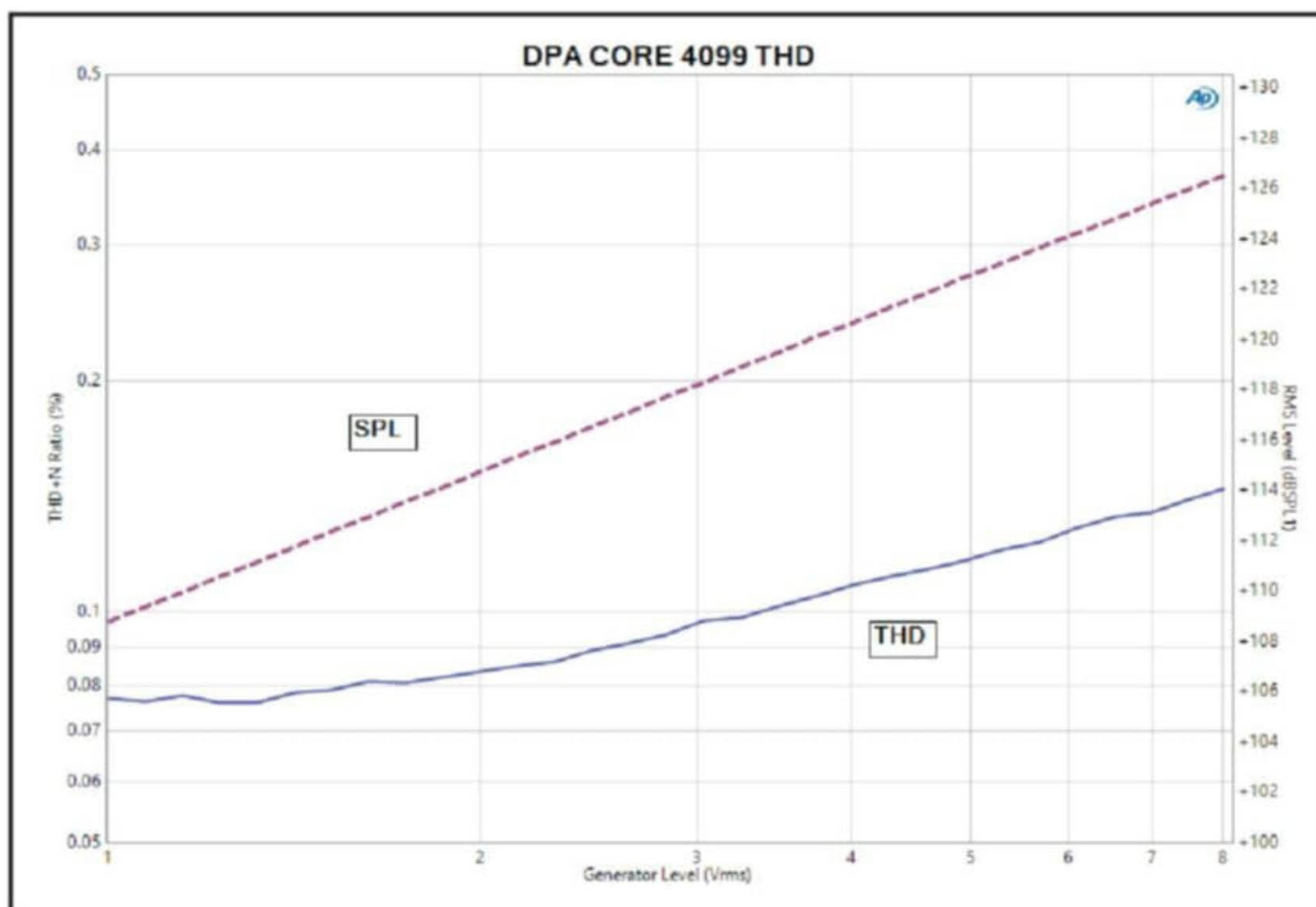


Figure 4. The distortion performance at 200 Hz when swept from 105 to 127 dB SPL is impressively low, which correlates to the subjective impression of an ultra-clean sound.

I liked the tonality that the 4099 captured, but the mechanical noise pickup from the keys was something that would take some work in the mix to minimize, certainly starting with a high-pass at 150 Hz or so. If the mic had been placed anywhere further down the flute, the key noise would have been even higher. The upside is, of course, that Slavin could bob his head around freely without getting unwanted volume and tone wobble, and it was easy to position the mic to minimize breath noises. A link to the raw recording is in Resources. We also had Slavin try the mic with his flute in a live performance, and in that application it worked superbly (see **Photo 3**).

Next up was a pennywhistle, and here, we had a lot less success. The band of the Universal clip would not get tight enough, and the mic kept slipping while Slavin was playing. We did not get a satisfactory recording for that instrument, but to be fair, this isn't one listed by DPA for that model clip. This is an instrument better served with a stand-mounted mic.

The next instrument that gave us a problem was Slavin's mandolin. The rounded back did not allow the clamp portion of the clip to find a secure purchase, and it slipped off several times. With a more conventional flatter back on the instrument (this mandolin was an Ovation), this would not have been an issue, but if that's the style of instrument you play, stand mounting is mandatory.

We did a recording that way, with the microphone about 5 to 6 cm from one of the mandolin f-holes, and the presentation was clean, bright, and incisive. Maybe a bit too much so (the treble rise of the 4099 makes the sound more than a bit jangly and fierce, and the pick noise was too dominant), but EQ can tame that while retaining the ultra-clean sound quality. Again, a link to the raw track is listed in Resources, and as before, you can play with EQ and compression to see what kind of sounds you can get in a mix.

Finally, we used the guitar clip to attach the 4099 to Slavin's Taylor six string single cutaway acoustic guitar. This time, the attachment was solid and we

got a really nice rendition of the guitar tone. A link to the raw track is also in Resources, and there would be very little processing in a mix that I would want to do. For close-miking an acoustic guitar, the 4099 would be one of my top choices.

Wrap-Up

As with any instrument microphone, there are pluses and minuses. On the plus side, the low distortion and extended frequency response made for ultra-clean and dynamic tracks. The tight hypercardioid pattern was great for suppressing unwanted room sound in the recordings and for minimizing feedback in live performance situations. The 4099 is light, unobtrusive, and versatile for mounting and positioning, and allows musicians a lot more freedom of movement than when they have to play to a fixed microphone.

On the negative side, the shock-mounting is better than nothing, but is inadequate to significantly reduce mechanical shock from instrument fingering. Much of that can be fixed by the recording engineer or by judicious use of filters in live performance. For conventional instrument shapes, the clamping systems from DPA work very well. But if your instrument is



Photo 3: The 4099 worked superbly as a live performance pickup for the flute, allowing multi-instrumentalist LJ Slavin to move around at will and reject feedback from his monitor speaker. (Photo courtesy of Cynthia Wenslow)

unusual in size or shape (e.g., the pennywhistle and the Ovation mandolin that we used in recording) you may have to resort to stand mounting, which loses some of the advantages of instrument mounting (e.g., the freedom of the musician's movement.)

Depending on the chosen accessories, the price of the 4099 is between \$600 to \$650. For that cost, you get a lot of versatility and some very interesting performance-oriented technology, as well as a microphone that will withstand the rigors of daily use and give many years of reliable service.

Project Files

To download test recordings, visit <http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

Flute, Guitar, and Mandolin Recordings: <https://syclotron.com/?p=2903>

LJ Slavin: <http://soundsforreal.com/bio.html>

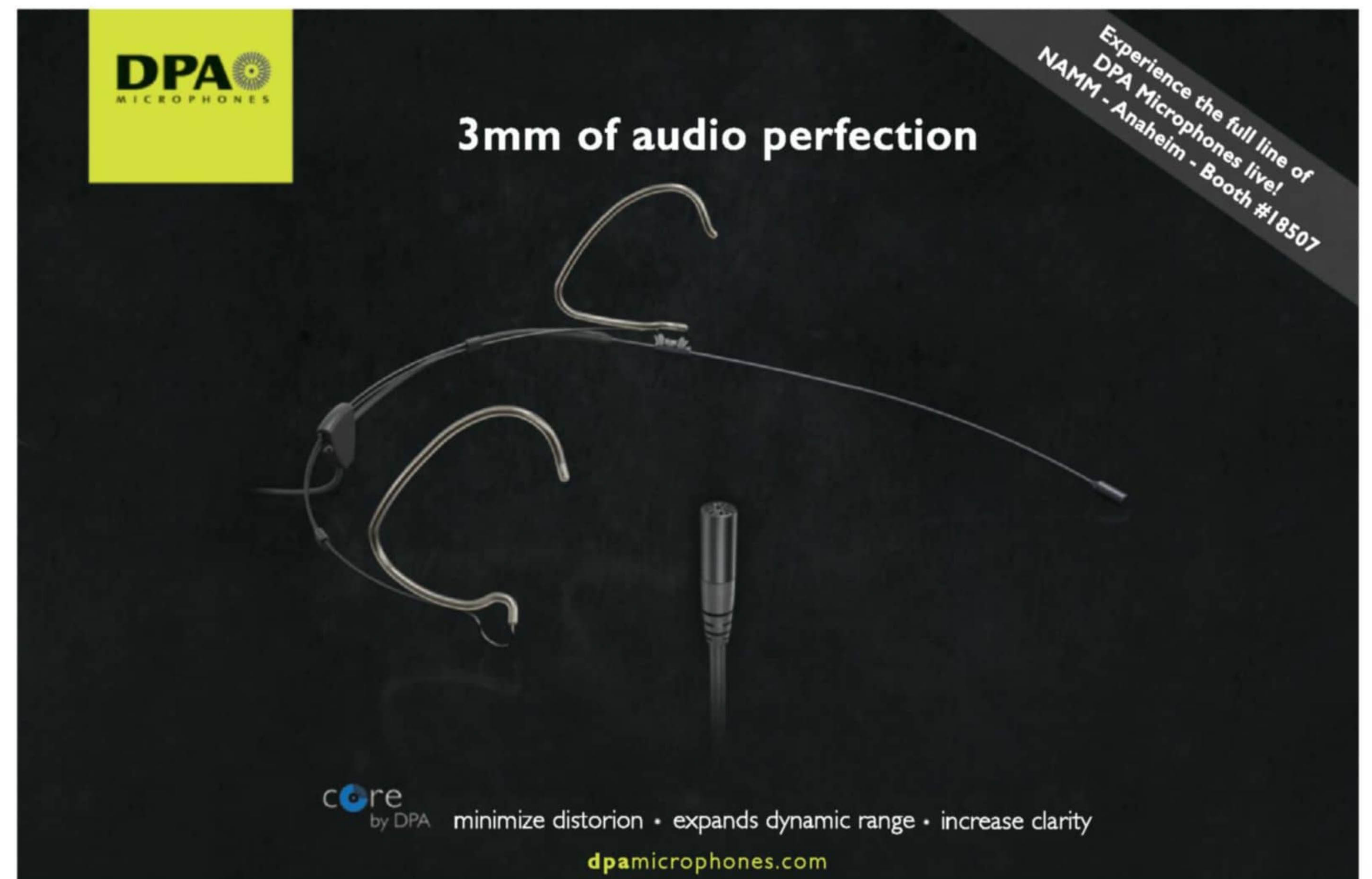
Resources

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"The amazing construction of a miniature microphone capsule," https://youtu.be/MG2eYTsBH_o

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1MORE Triple Driver Over Ear Headphones

By
Stuart Yaniger
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Stuart Yaniger puts 1MORE's Triple Driver Over Ear headphones through a series of rigorous tests. Intrigued by the design that uses a 40 mm graphene-coated dynamic driver, a ceramic tweeter, and what the company calls a bass reflector, Stuart investigated and shares what he found.

1MORE is a China-based company specializing in providing high-quality (both perceived and actual) headphones in a variety of geographic and application market segments. 1MORE has established a reputation as having smart engineers who can deliver performance well above the average for the price. So even though I'm not really a headphone guy,



Photo 2: The TDOEs fold up neatly into a provided hard-shell carrying case.

I jumped at the chance to review the 1MORE Triple Driver Over Ear headphones (see **Photo 1**). To save on typing, I will abbreviate the 1MORE headphones to TDOE for the rest of this article.

The TDOEs remind me a bit of that old joke about the Holy Roman Empire—it was neither holy, Roman, nor an empire. In this case, the headphones really aren't "over ears." They're more of an on-ears/over-ears hybrid. The earpads are small enough not to engulf the pinnae entirely, but large enough to form something of a seal around the perimeters of the pinnae. Likewise, there are only two drivers—the third isn't actually a driver since there's no electrical signal presented to it. Instead, it is what 1MORE calls a bass reflector (and isn't actually a reflector). More about that shortly.

Company Background

A little background on the company: 1MORE was founded by former executives at Foxconn, the Chinese electronics behemoth. Because of their experience with subcontracting design and assembly houses, they were able to leverage that knowledge to form a high-quality supply chain. The result has been a line of headphone products that have been extremely successful in the Chinese domestic market, and now are experiencing a rapid expansion into other geographic areas. For North America, 1MORE has established a headquarters in



Photo 1: The 1MORE Triple Driver Over Ear headphones (TDOE) are an interesting entry into the dedicated audio market segment.

San Diego, CA, and its marketing strategy appears to be relying on influencers and viral marketing, rather than the traditional heavy advertising campaigns.

1MORE heavily promotes the idea that its headphone designs are "tuned" by Luca Bignardi, an experienced MIDI programmer and mixing engineer. The description of the process is a bit vague and more analogy than substance, but whether the process is equalization or "floobydust," this is certainly part of the marketing story needed for differentiation in a crowded market. You can read 1MORE's take on it at the link provided in Resources.

The Product

True to the marketing-driven approach, the packaging for the TDOEs is elaborate and has a luxurious feel. When the box is unfolded, the inner covers are decorated with vaguely da Vinci-ish sketches. Included in the package are a soft carrying bag and a hard-shell case for the headphones. The headphones fold up to fit in the case (see **Photo 2**), but the overall case size is a bit large for traveling, at least for placing in carry-on bags. Also included is a cable terminated with a 3.5 mm TRS plug on one end and a Y-split cable at the other, with connections to the headphones made separately by left and right channel plugs that go to each earpiece. As with the packaging, the cables and plugs feel solid and very high-end. A helpful touch was easily visible color-coding of the left and right channel connectors (see **Photo 3**).

The headphones themselves feel far more solid and well-constructed than you'd expect at their price point. There's very little plastic, with most of the construction being anodized aluminum and leather. The earcups pivot on two axes for fit and comfort, and there's no discernable play—can I say "solid and luxurious" again? Despite the solid feel, the TDOEs are relatively light (about 10 oz or 300 g). The headband and ear pads are extremely soft and compliant. One more nice touch for those of us aging Baby Boomers who usually need to squint to make out channel designations is that the inside of the earcups have very large printed R and L designators that are about the size of the leading E on an eye chart (see **Photo 4**).

Although physical comfort is even more subjective than sound quality, I have to say that unlike most headphones, I found the TDOEs to be extremely comfortable for my head and ears, and I was able to wear them for hours on end without fatigue or ear-sweat.

More controversial is the styling, which apparently was overseen by an automobile enthusiast. The outer part of the earcups have transparent polycarbonate covers to form a sound



Photo 3: The Y-cord for the TDOE plugs into color-coded jacks on the headphones.



Photo 4: Left and right channels can be spotted at a glance.



Photo 5: The styling of the headphone cups is love-it-or-hate-it, resembling sporty automobile wheels. The "bass reflector" can be seen through the transparent covers.



Figure 1: This rendering shows an exploded view of the 1MORE Triple Driver Over Ears (TDOE) headphones. (Image courtesy of 1MORE)

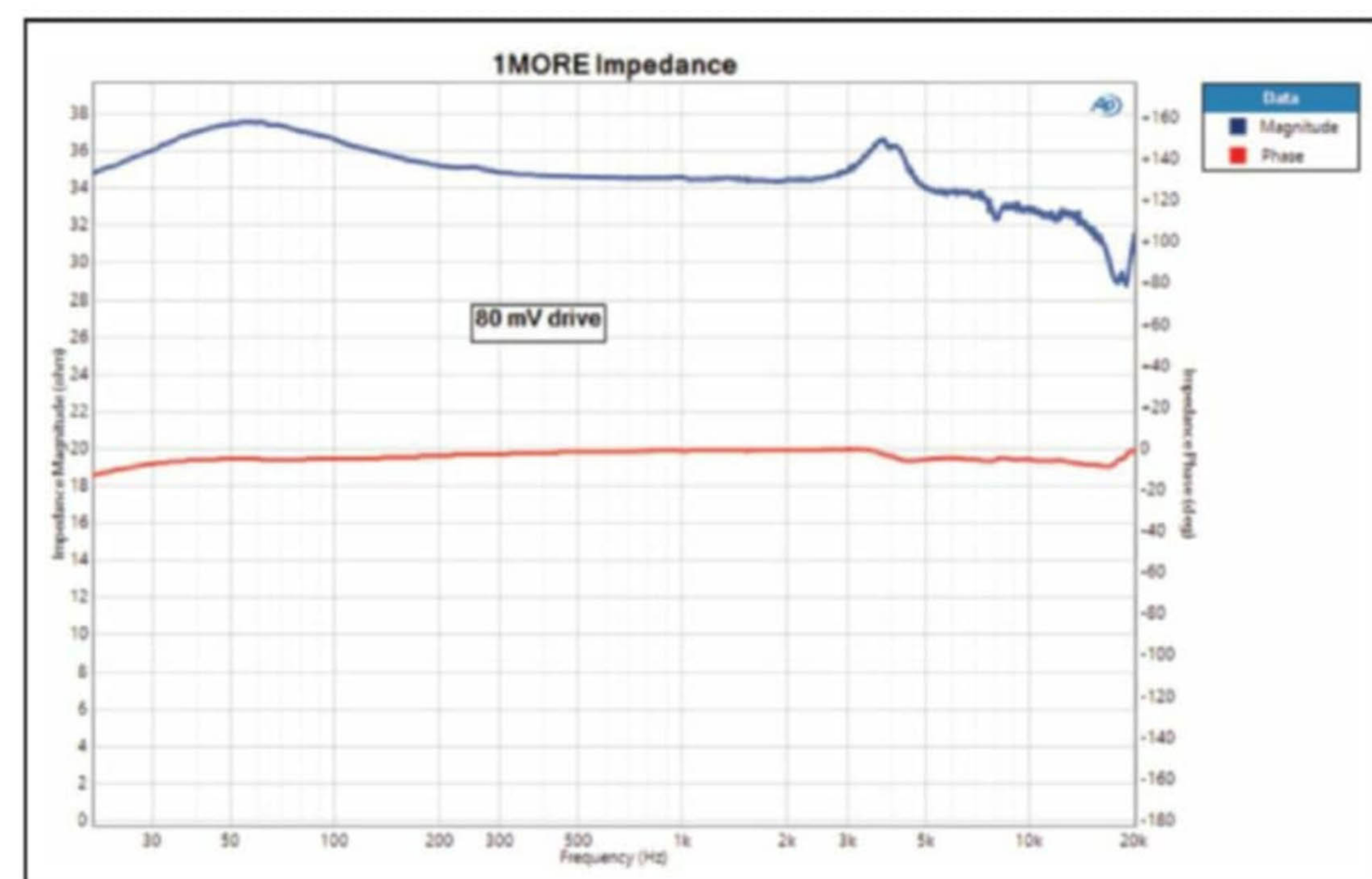


Figure 2: The impedance versus frequency plot for the TDOEs shows that these headphones are an easy load for amplifiers.

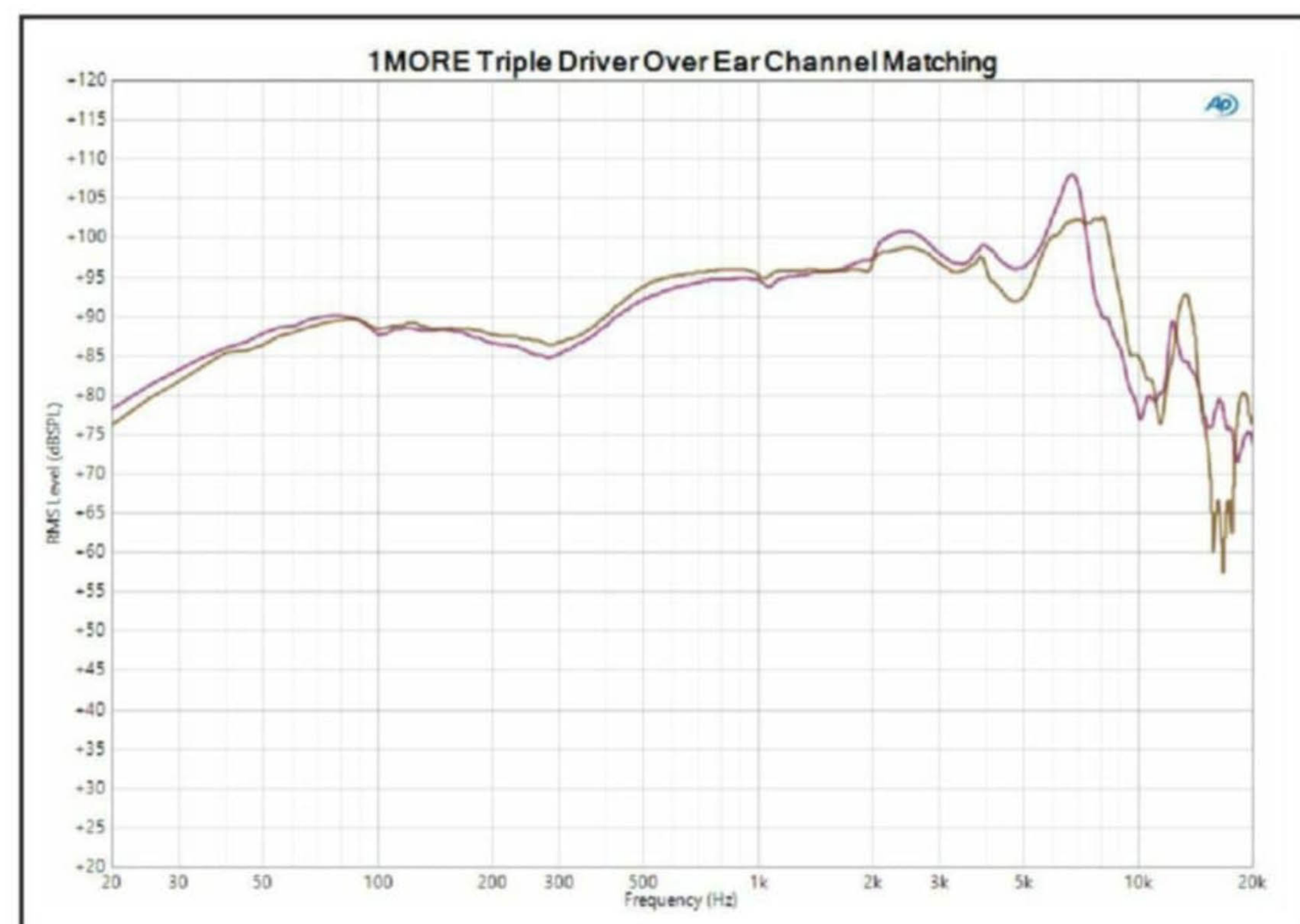


Figure 3: This is the plot of frequency response for the left and right channels using the Audio Precision AECM206 headphone test fixture.

chamber with the so-called bass reflectors (which are visible through the polycarbonate), but the transparent covers have a molded plastic piece over them, which is designed to resemble the spokes of an automobile wheel (see **Photo 5**). As far as I can tell, this is a purely decorative feature that has no acoustical function, but it does provide a nice place for dirt to settle. Compared to the functional aspects of the mechanical design, this styling gimmick is a minor annoyance. However, if you're a gearhead, you might find it attractive. In any case, although the spoked feature gives the headphones the appearance of being open-backed, they actually are not, thanks to the polycarbonate covers.

One thing to note—the TDOEs are intended for music listening. They do not include a microphone or remote volume controls. This will be no surprise to serious headphone users, but I have noted complaints about this lack from various online reviews.

An exploded view of the TDOE's construction is shown in **Figure 1**. The heart of these headphones is a 40 mm diameter driver with a diaphragm made from polyester that has been vapor-coated with a layer of graphene, then thermoformed. The graphene, which is a two-dimensional crystal form of carbon, is intended to stiffen the diaphragm with a minimal increase in moving mass. 1MORE does not disclose the thickness of the coating, nor any supporting data to show that the graphene has a significant (or unique) effect. Nonetheless, its marketing heavily promotes the use of graphene in these drivers.

There are rumors on various websites that 1MORE makes a version of the TDOE using a titanium layer instead of the graphene for the European market. If true, I'd love to see the comparative performance. It's unfortunate that "graphene" has been a major buzzword in tech products, replacing the last fashion, "nano." And that's why I'm curious and a bit frustrated at 1MORE for not being forthcoming. If the use of a graphene coating is a real technological plus rather than a marketing gimmick, I'd think the company would be eager to show data demonstrating the superiority of the approach.

At the high-frequency end, the main driver crosses over to a ceramic piezoelectric driver. 1MORE would not provide any information on the crossover point or the slopes. In Internet searching, I've seen a variety of confident answers to this, no two of which are the same. In any case, because I was reluctant to take the headphones apart to check this for myself, like the actual effects of the graphene coating, it will remain a mystery for the moment.

At the low-frequency end, 1MORE touts its "Bass

Reflector," which strikes me (again) as a marketing term rather than technical. Of course, as I previously mentioned, it is in no way a driver since it's a passive element. Given the dimensions of headphones and the drivers relative to the wavelengths of bass sounds, "reflector" is also an inapt term.

When I asked about the Bass Reflector's function, 1MORE refused to comment beyond some vague marketing-speak (I'm sensing a pattern here!). But with a bit of examination, I think we can reasonably hypothesize. The Bass Reflector looks like a passive radiator, with a flat rigid diaphragm surrounded by a compliant elastomer suspension. This forms an enclosed acoustical cavity between the Bass Reflector and the polycarbonate covers, this volume being behind the main 40 mm driver. This cavity will have a resonance, which depends on the cavity volume and compliance, the mass of the diaphragm, and the compliance of the suspension. So the back of the 40 mm driver is loaded by this resonator, which may serve to increase the effective rear volume and enhance the bass output below the 40 mm driver's normal resonance. The potential downside to this is that the asymmetrical load between the front and the back of the 40 mm driver might be expected to exhibit increased second-order harmonic distortion.

Performance

Let's get to the crux of the muffin. I performed several sets of measurements using an Audio Precision APx525 audio analyzer, the APx1701 transducer interface, and the AECM206 headphone test fixture (see Resources). A few measurements were repeated using the miniDSP EARS headphone test fixture (to be reviewed in the March 2019 *audioXpress* issue) in conjunction with the APx525. Listening at home, work, and a hotel was mostly done using a CEntrance DACportable DAC/headphone amplifier. My listening while hiking used a decidedly lower tech iPhone 8S with a standard Lightning-to-3.5 mm adapter module.

Figure 2 shows the TDOE's impedance magnitude and phase versus frequency. 1MORE rates the impedance at 32 Ω , which seems very reasonable. Note that the magnitude curve is relatively flat, and that the phase angle stays below $\pm 10^\circ$, which is a very easy load to drive with just about any headphone amp. The sensitivity was likewise high, with 63 mV (0.12 mW) yielding a sound pressure level of 95 dB at 1 kHz. My iPhone, not exactly a powerhouse, had no difficulty driving the TDOE cleanly to much higher levels than I am comfortable.

Figure 3 and **Figure 4** show the frequency response measured with the AECM206 and miniDSP EARS test fixtures, respectively, with the former also showing

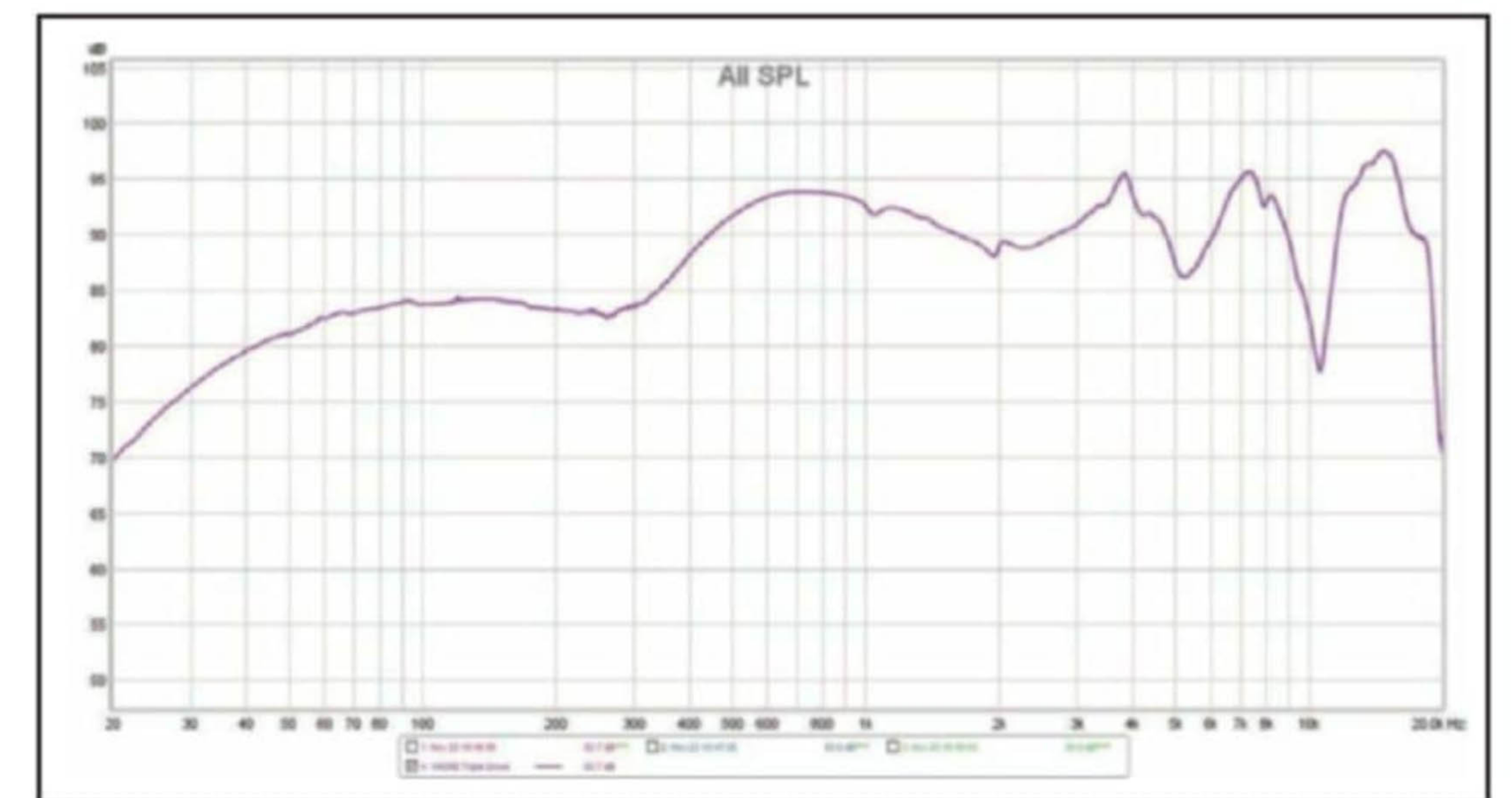


Figure 4: This is the plot of the averaged frequency response using the miniDSP EARS headphone test fixture for comparison.

the matching between channels. I'm presenting results from both setups because test fixtures do show significant variations in their measurements due to different dimensions, different sizes and hardnesses of the artificial pinnae, and different artificial ear canal geometries. So with both of these plots, we can look at the commonalities and attribute those to the actual performance of the headphones.

The most interesting feature is the shelf in the bass response. This is totally unexpected because subjectively, the TDOEs have bass that is... not shy. Not boomy or overripe, but not at all reticent and a bit tipped up. The texture and definition of the bass also struck me as very good, if not the very best I've heard. One example that stood out to me in subjective listening was the Steven Wilson remix and remaster of the classic Jethro Tull album *Benefit*. The kick drum and bass stood out individually with no hint of muddiness or boom. It really gave me a sense of the improvement with this album edition.

What could explain the disconnect between

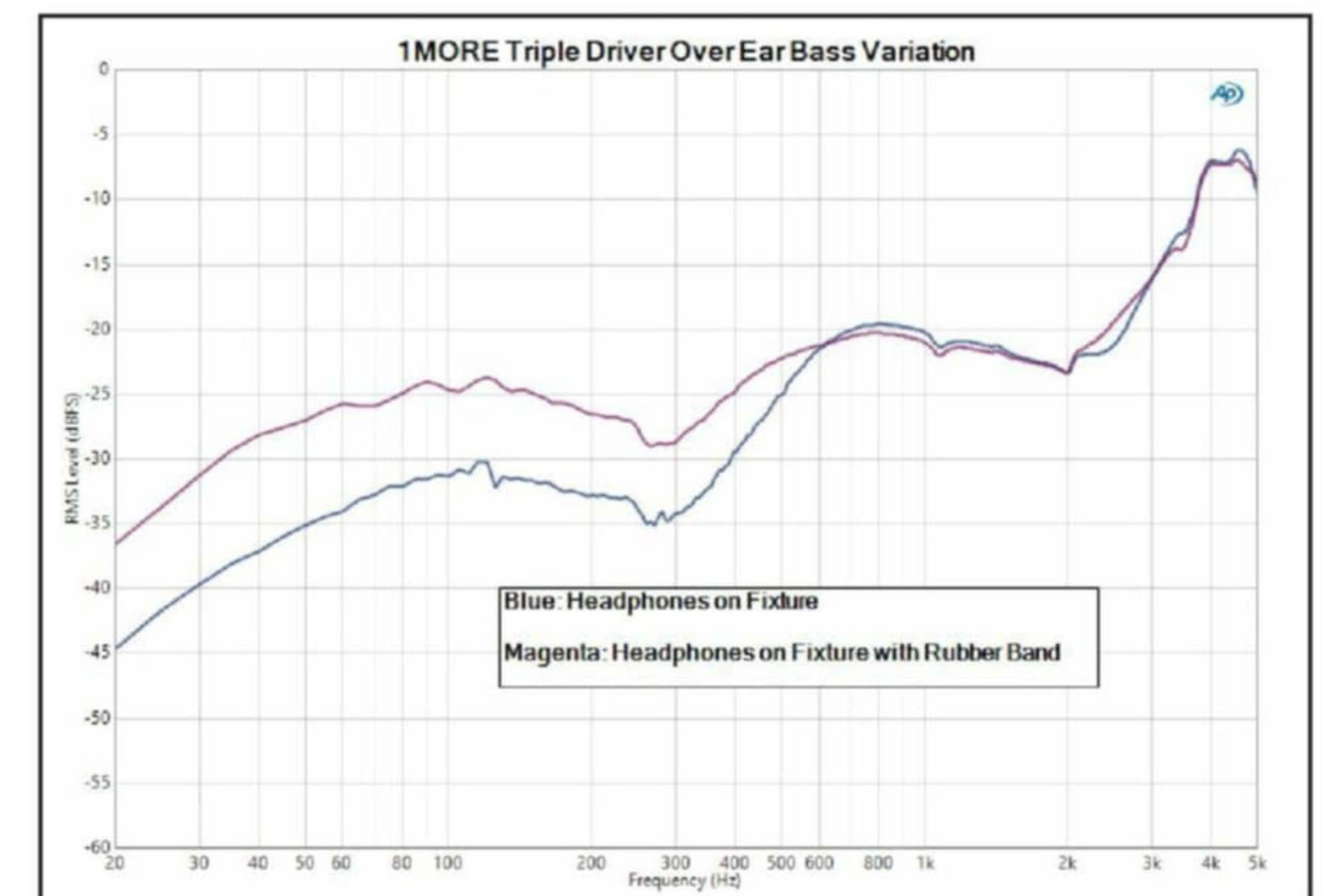


Figure 5: Applying compression to the headphones improves the seal to the test jig (in this case, the miniDSP EARS) and gives a more realistic bass measurement.

measurements and perception? I will venture that the root cause is the seal of the headphones to the artificial ears. I mentioned in my reviews of both test fixtures that the pinnae do not bend and compress in the same way as actual human ears, and this is clearly going to be critical with the sort of hybrid over-ear/on-ear seal that the TDOEs depend on to achieve their good bass response. To test this

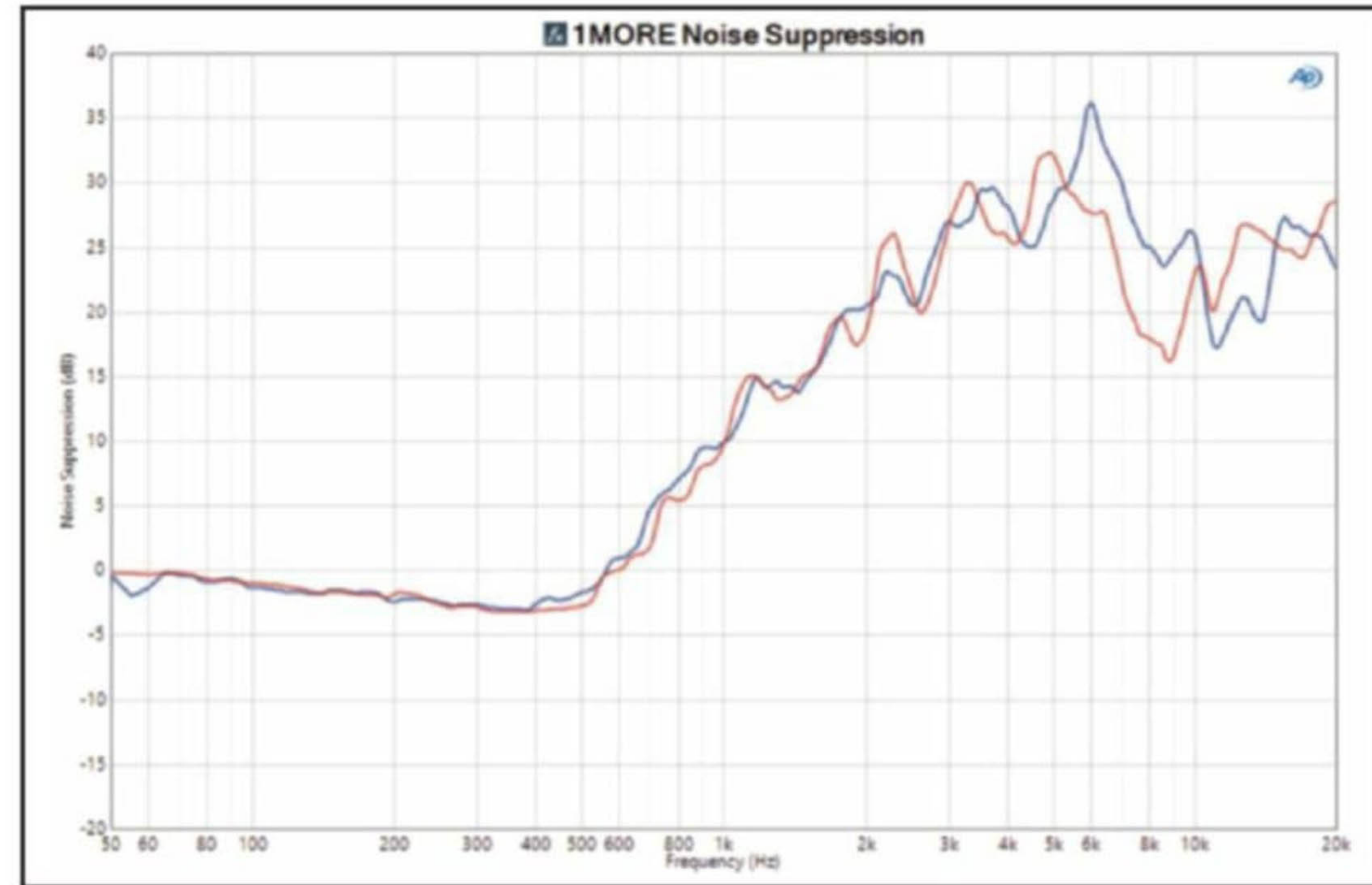


Figure 6: The TDOEs provide impressive acoustic isolation from ambient noise at mid and high frequencies. These data were acquired with the AECM206 fixture.

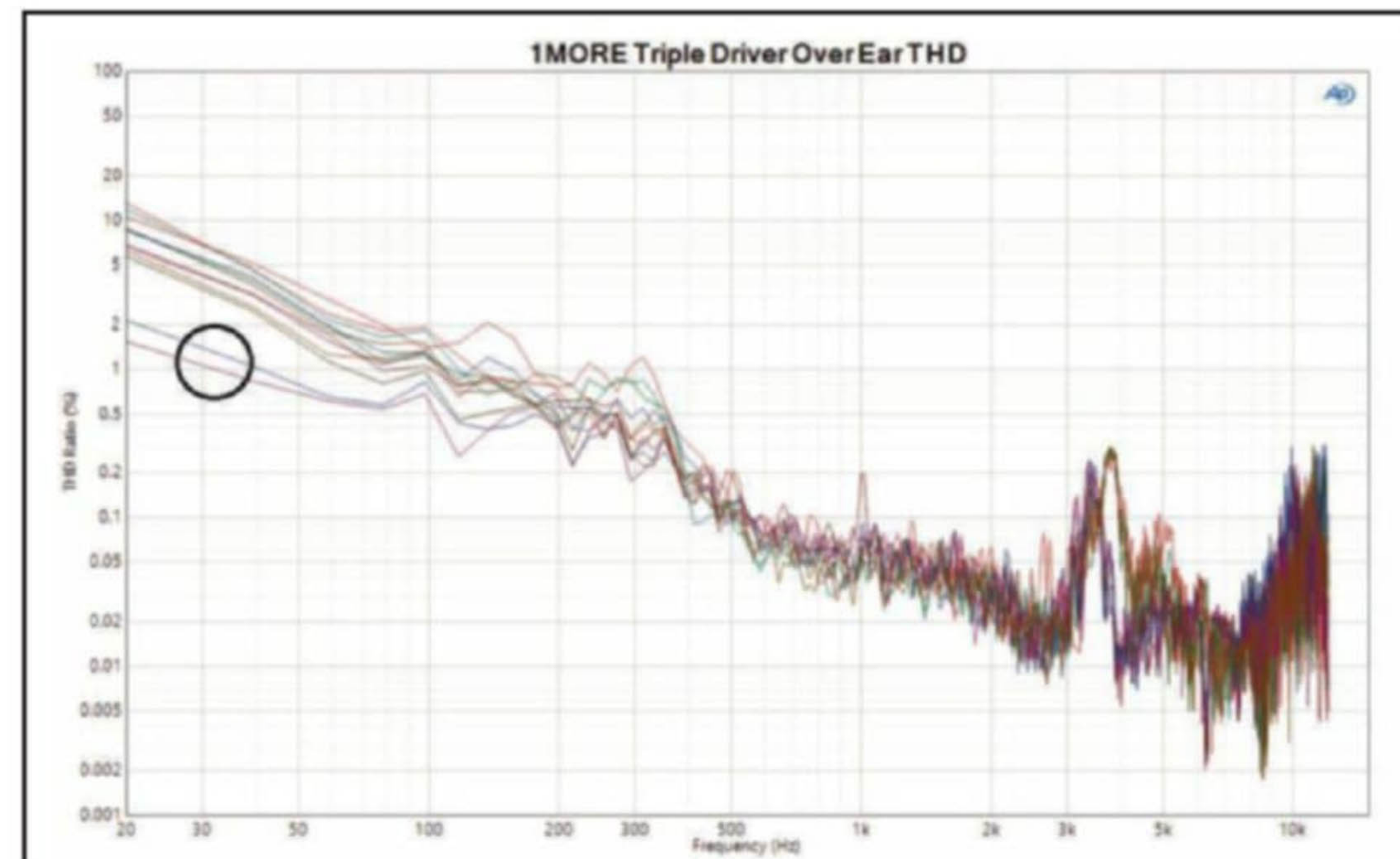


Figure 7: The THD versus frequency taken at 90 dB SPL with the AECM206 test fixture is commendably low in the midrange and treble, other than a glitch in the upper midrange. The circle indicates sweeps taken with the headphones compressed against the test fixture.

About the Author

Stuart Yaniger has been designing and building audio equipment for over half a century, and currently works as a technical director for a consumer products 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.

hypothesis, I measured the TDOEs on the miniDSP EARS fixture, then remeasured with a rubber band put around the headphones and fixture to artificially compress the earpads and increase the seal. The results are shown in **Figure 5** and the 10 dB boost in that bass shelf clearly demonstrates that the issue is the fit to the test fixtures, not with the response of the TDOEs themselves.

What does correlate with my subjective perceptions is the deep notch at about 10 kHz. It doesn't make the TDOEs sound dull, but it does serve to take the edges off recordings and gives a sound that is easy and non-fatiguing to listen to without at all sacrificing clarity or definition—which these headphones have in spades. Ditto the slight unevenness in the midrange. This does not come across as a recessed or muffled tonality, but more of a relaxed and easy quality. Close-miked and compressed vocals in particular can be annoying (and are more often than not the norm in commercial recordings), but the TDOEs tended to smooth them over. A good example is *Matt the Electrician Is Alive*, which can be almost unlistenable with a lot of headphones and speakers, but had its ear-laser quality noticeably reined in with the TDOEs.

Another subjective perception was that of the suppression of external noise. There's no active noise cancellation, but slipping the TDOEs on immediately made things around me significantly quieter. In this sense, the passive noise-blocking of the TDOEs was much more like over-ear than on-ear headphones, and the very soft and compliant earpads worked as intended. **Figure 6** shows the passive noise reduction achieved by the TDOEs. This measurement was taken by placing the AECM206 headphone test fixture about a meter from a pink noise source and measuring the response with and without the headphones in place. The response with the headphones was then ratioed to the noise with the bare test fixture to eliminate the influence of any response variations in the source. As can be seen, the effectiveness increases significantly in the critical midrange and treble, and 25 to 35 dB of passive noise reduction is achieved in the regions where hearing is most sensitive.

Another Observation

I like to listen to music when I'm out walking or hiking. Unfortunately, living in the Chicago, IL, area, the experience is often marred by excessive noise from the ever-present wind. My usual over-ear headphones, which normally do a decent job of acoustic isolation, become almost useless, seemingly amplifying the wind noise. The 1MORE TDOEs did a terrific job of reducing that noise to

nearly unnoticeable levels. Impressive indeed, and since they arrived, my old headphones have been relegated to indoor use only.

Distortion performance is shown in **Figure 7**, which was taken with the AECM206 fixture and at 90 dB level at 1 kHz. Each curve is a single measurement after repositioning. Because of the seal issue mentioned previously, the rise in distortion in the bass is likely an artefact of the test, and indeed, when I lightly squeezed the headphone cups to the test fixture with my hands, the bottom two curves (indicated by a circle) were generated, showing dramatically lower bass distortion.

An interesting feature of the distortion curve is the spike in distortion to 0.2 to 0.3% between 3 and 4 kHz, with a slightly different peak frequency and amplitude between left and right channels. To make sure that this curious feature was due to the headphones and not the measurement, I repeated it with the miniDSP EARS and using REW software. The same feature was present, and harmonic analysis showed that it is principally second order. It's likely not a coincidence that it coincides with the small impedance peak seen in Figure 2, so there's more than a good chance that this is either a diaphragm resonance or a hint about crossover point. Since piezo tweeters are rarely crossed over that low, and there's some inconsistency between channels, my bet would be on the former. I can speculate that whatever causes this resonance may well be responsible for the rather smooth and relaxing midrange.

Wrap-Up

I admit that I'm not particularly a headphone enthusiast. I've gotten to spend time with a lot of different high-end models, but I'm a loudspeaker enthusiast first. Most of my headphone use is either when I'm traveling, during hikes, and occasionally for mastering my recordings. Although I'm certainly aware that the sound I got from the TDOEs doesn't have the neutrality or ultimate definition of the best headphones I've used, and is a bit too colored for mastering use, I found myself using these headphones for listening for enjoyment much more so than any of the other headphones that have passed through here. For dedicated headphone geeks, the TDOEs might not be their first choice, but for someone more interested in headphones that are easy to drive, easy to wear, and easy to listen to, they're a great deal at the \$250 asking (and under-\$200 street) price. My only real regret is that the interesting technology and careful engineering that undoubtedly went into achieving these goals is overwhelmed by the marketing, but I can't fault the company for recognizing that its target market is broader than techno-nerds like me.

Resources

1MORE Triple Diver Over Ear headphones
<https://usa.1more.com/collections/headphones/products/tripleoverear>
<https://usa.1more.com/blogs/reviews/a-little-light-on-luca-bignardi-and-tuning>

O. Masciarotte and S. Yaniger, "CEntrance DACportable High-Resolution USB DAC and Headphone Amplifier," *audioXpress*, April 2017, <https://www.audioxpress.com/article/fresh-from-the-bench-centrance-dacportable-high-resolution-usb-dac-and-headphone-amplifier>.

S. Yaniger, "Audio Precision AECM206 Headphone Test Fixture," *audioXpress*, April 2018.

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Noise Isolation for Home Studios

Richard Honeycutt shares some of the challenges he encountered when he helped a client convert a guest bedroom into a home studio. And he offers a few solutions for readers who are thinking of repurposing a room of their own.

By
Richard Honeycutt
(United States)

Photo 1: Resilient material can be installed under wood flooring to improve IIC.
(Photo courtesy of Acoustical Surfaces, Inc.)

Often, when the word “acoustics” is mentioned, people think of the effects of a room on the quality of sound in the room: echoes, reverberation, boomy or brittle quality, speech intelligibility, etc. As important as these considerations are, an equally vital aspect of acoustics concerns control of noise—both internally generated noise (e.g., from HVAC units) and externally generated noise.



Figure 1: The area outlined in red is the part of the home that will contain the home studio.

A client recently contacted me to help him with his plans to repurpose a room on the first floor of his house from a guest bedroom to a home studio. He understood some of the challenges involved, but planning this project gave him the opportunity to learn a few new things.

Figure 1 shows a section view of a home similar to that of my client. The actual floor plan consisted of two rows of rooms, and the section view only shows one. The section is cut as if the wall nearest the viewer were removed; the other row of rooms is located “further into the page.” The plan shows the downstairs dining room (boxed in red) in the location actually occupied by the guest bedroom in my client’s home.

HVAC Noise

As previously mentioned, noise can be categorized by the location of the source: internal or external. Internal noise can be generated by machinery inside the room or introduced into the room by ducts and vents. This includes noise from HVAC or any sort of motor or fan inside the room,

plus noise created in other rooms and carried into the studio through HVAC ducts or through the air or structural elements.

Often a home’s HVAC system consists of a heat pump that heats or cools the air in a plenum, from which ducts are routed to the various rooms in the home. If more than one room is served by branches of a single duct, noise will pass easily between those rooms. Unless the noise is fairly intense or the ducts are short, noise from one room that passes through a duct into the plenum, and then through another duct to the studio is likely to be well enough attenuated that it is not a problem.

Figure 2 shows an HVAC duct layout that is somewhat similar to that on the first floor of my client’s home. Room 3 is the future studio. Note that there is no direct path from another room through any of the ducts into the studio. Sound from, say, room 4 would have to pass through the small ducts from that ceiling to the blue and green supply trunk that behaves acoustically more or less as a plenum would, and then back through another small duct to get to the studio. The brown air return duct does not terminate in the studio. If room 4 had been chosen as the studio, there could be a problem with noise ingress from room 5, as noise could easily pass through the pair of small ducts from one room to another.

High-velocity air passing through vents in the studio can generate unacceptable levels of noise. If this situation exists in a home, the only economically feasible option for quieting a studio may be to close off the HVAC supply to the studio and provide heating or cooling by opening the doors to adjacent rooms between takes. This solution can also be applied to studios that share HVAC ducts with other rooms that are noisy.

Noise Transmitted from Rooms Above

Referring back to Figure 1, you can see that the room most directly above the studio is an open space serving more or less as a hall. Had the studio location been below a bedroom, noise isolation would have been easier, since foot traffic—which causes impact noise—in bedrooms is limited for much of the time. However, in halls, there is likely to be considerable foot traffic at most hours. Footfalls create noise with strong low-frequency content. This noise is easily conducted through the building structure into rooms below. My client’s upstairs hall had a hardwood floor that he wanted to keep. Carpeting the floor would have greatly reduced the “heel-strike” noise created by people walking in the hall. An expensive option would have

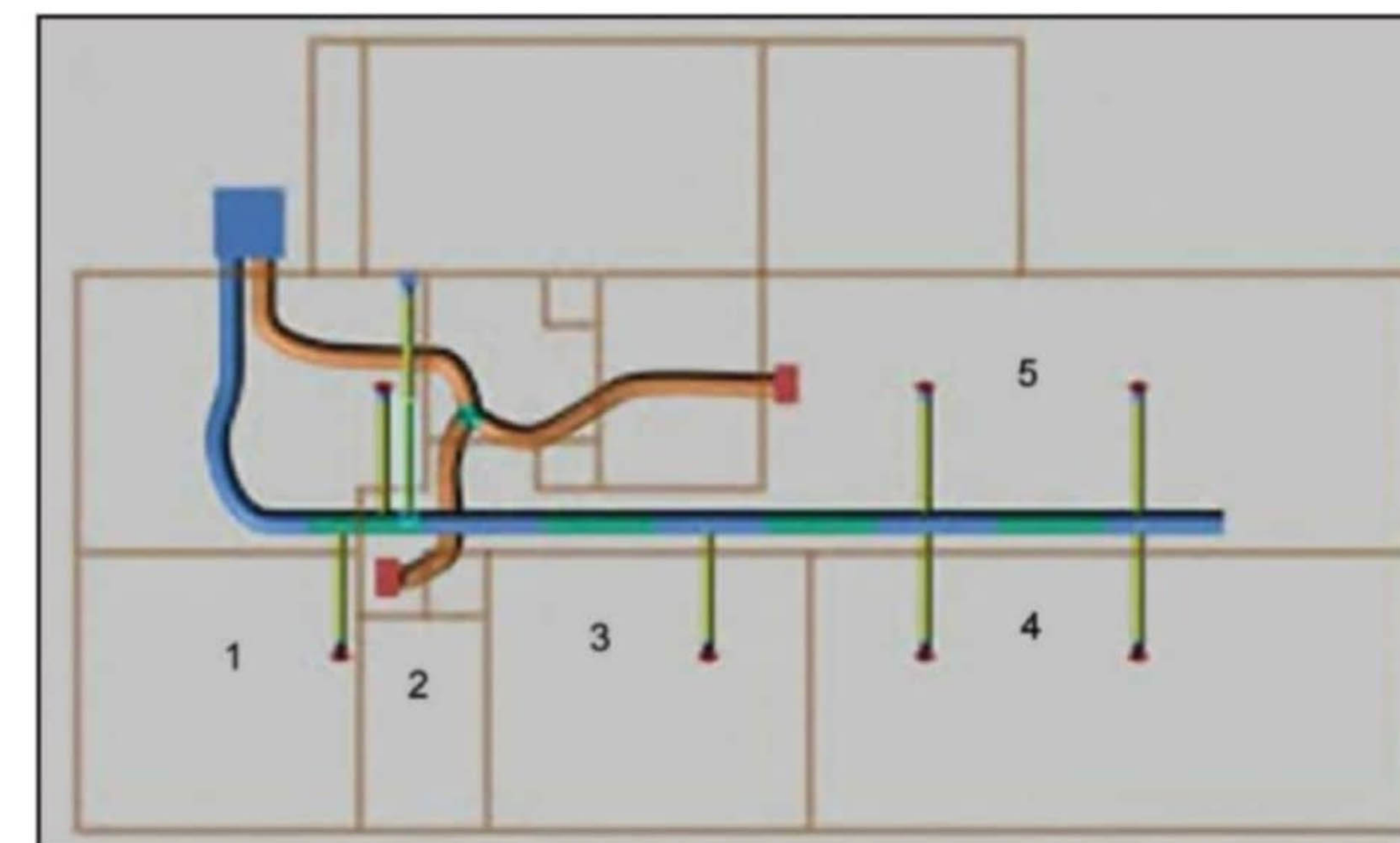


Figure 2: This HVAC layout is similar to that found in my client’s home.

been to re-floor the hall, using a resilient acoustical floor underlay such as Quiet-Floor NP, as illustrated in **Photo 1**. The method my client chose to reduce floor-to-ceiling noise transmission was to install a ceiling of gypsum tiles in the studio, with the supporting grid mounted using resilient hangers (see **Photo 2**), and with fiberglass batts in the cavity between the original ceiling and the suspended one. Since he had planned to use a combination of absorptive and reflective ceiling material in the studio anyway, the suspended gypsum tiles did not constitute a major cost addition. I did warn him not to use fiberglass ceiling tiles, but to use mineral fiber or stone wool ones instead, because the low density of the fiberglass tiles keeps them from providing significant noise isolation. Mineral fiber and stone wool tiles are more massive, and, therefore, provide better sound insulation.



Photo 2: This resilient hanger can be used for a suspended ceiling.

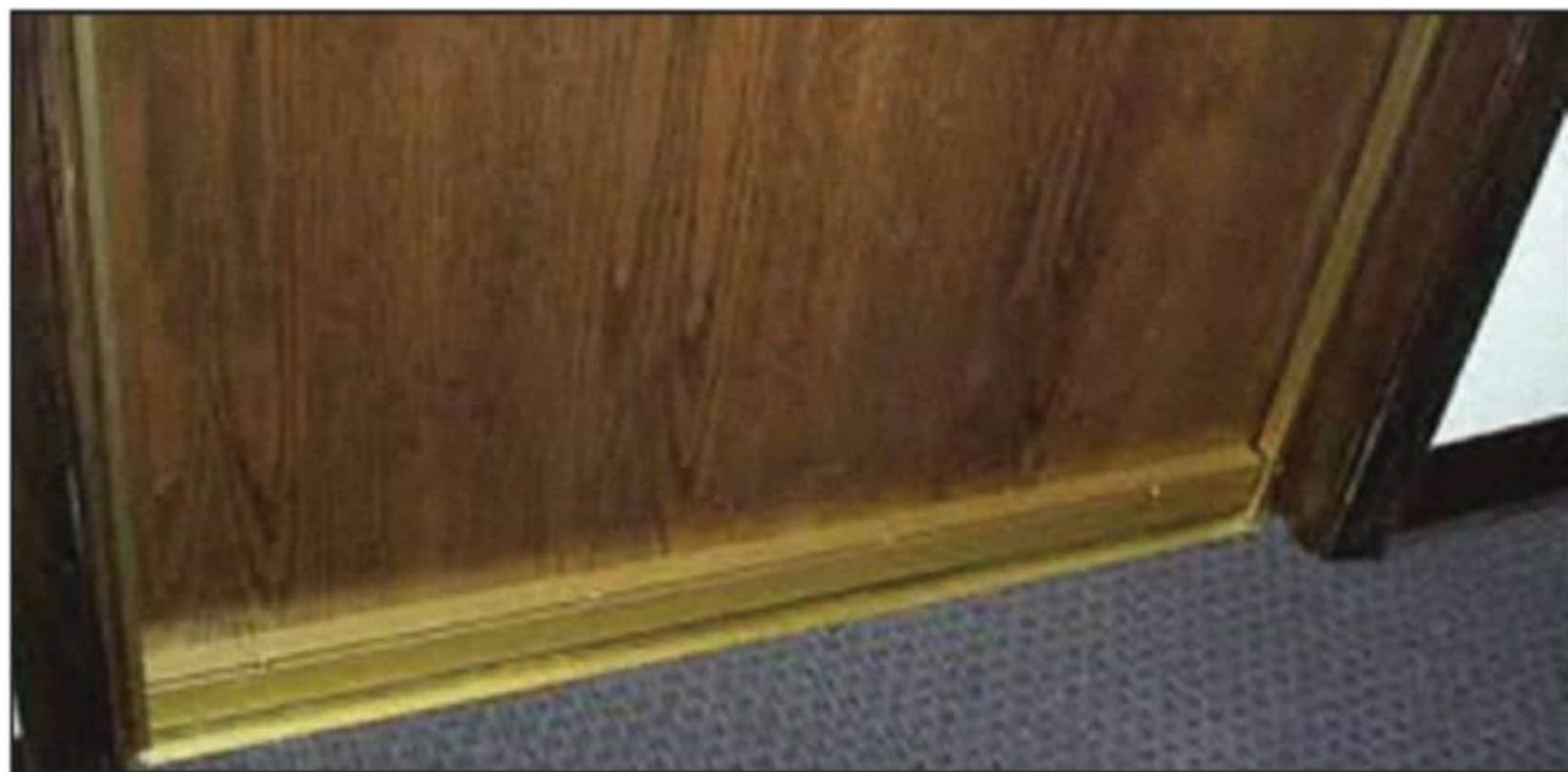


Photo 3: In this instance, we were able to use an acoustical door-bottom seal.

Since impact noise conducted from the upstairs floor can radiate from walls in the lower room, for further noise isolation it would be beneficial either to add a layer of resiliently mounted gypsum, perhaps using resilient adhesive such as Green Glue, (www.greengluecompany.com) to the walls downstairs or to replace the downstairs walls with Quiet Rock (www.quietrock.com) mounted on Genie Clips (<http://pliteq.com/products/genieclip.php>). Adding further ceiling treatment beyond that described in the previous paragraph would not help as much as this wall modification.

Airborne Noise from Adjacent Rooms

Airborne noise from adjacent rooms can enter through either of two paths: gaps around doors or conducting room boundaries. Most recording and broadcast studios use acoustical doors, or doors having excellent seals at the top, sides, and bottom (see **Photo 3**). Acoustical doors can do an excellent job, but are usually expensive. A heavy solid-core door with good seals is often effective enough for a home studio.

The other path begins with airborne noise in one room vibrating the boundaries (walls, floor, and/or ceiling) of that room. The noise is then conducted through the building structure into the boundaries of the studio, from which it can re-radiate. This sort of noise can be reduced by adding mass to the walls, perhaps in conjunction with resilient materials. An example is the use of Quiet Rock or layers of gypsum board laminated together with Green Glue. If the noise path involves the floor, a floating floor may be needed in the studio. This consists of a massive (wood, concrete, or ceramic tile) floor mounted over a resilient underlay or supported by resilient mounts. My client was fortunate enough that he had no airborne noise issues requiring more than door seals to remedy.

Structure-Borne Noise

In commercial buildings, noise conduction through shared walls or floors can be a problem. For instance, a dance studio using bass-heavy music can cause the walls and floor to vibrate. This vibration can be conducted into other spaces—even spaces not immediately adjacent—and be re-radiated into a sensitive space. Such structural noise transmission in homes is not often an issue, unless a playroom, home workshop, or a similar area likely to have substantial impact noise or heavy bass is part of the building. Where such problems exist, mitigation may require vibration isolation of the walls and/or floors of both the room containing the noise source and the noise-sensitive room.

My client had no such problems. However, the refrigerator in the adjoining kitchenette (shown in Figure 1) conducted noise into the floor and into the wall between the kitchenette and the studio. Since the problem was vibration and low-frequency noise from the refrigerator, the best solution was to mount the refrigerator on resilient vibration mounts. This broke the conductive path for the noise.

Plumbing Noise

Probably the most obtrusive noise for a studio is plumbing noise. Toilets flushing or sinks draining do not enhance quiet passages in recorded music! Referring once again to Figure 1, you can see that my client has a bathroom almost directly above the studio. Reducing toilet noise depends upon whether the waste pipe is plastic or cast iron. In either case, a plumber would need to isolate the pipe-mounting straps with 1/2" felt.

If the pipes are plastic, wrapping them with mass-loaded vinyl (<https://acousticalsolutions.com/product/audioseal-duct-pipe-lag/>) will help significantly. It is best to apply the vinyl over fiberglass pipe wrap. Cast-iron pipes are so massive that they do not need this treatment.

To isolate the toilet, you need to lift it and put a resilient gasket under it. A quarter-inch thick, soft-rubber like gasket is recommended. Trim around the outside edge of the toilet. There are similar looking products made of hard plastic that will not work. To complete the toilet isolation, you can apply rubber or sorbothane grommets around the floor bolts where the nuts go on top to isolate bolts from toilet. If the toilet itself is noisy, there are relatively quiet toilets for less than \$200. My client was able to achieve satisfactory noise isolation by using the resiliently mounted suspended ceiling.

External Noise

In rooms with concrete slab floors, noise conduction through the slab can be a problem. The sources of such noise include heavy trucks, trains, and other devices that create high levels of vibration in the ground. Noise entry through a slab almost always requires floating floors, sometimes made of separate concrete slabs mounted above the original floor on vibration mounts.

Even using such extreme measures, slab-conducted noise poses a difficult problem. I once had to make some measurements in an anechoic chamber located about two blocks from a train track. Despite having a floating floor, the chamber had a 4-Hz vibration that showed up in my measurements whenever a train passed. My client's floor was framed above an air space about 6" above the ground, so slab-conducted noise was no issue for him.

Recording studios and other noise-sensitive installations located in downtown areas often have issues with traffic noise. For locations subject to frequent airplane flyovers, there may be aircraft noise with which to contend. In some residential

neighborhoods, lawn-mowing noise is a problem. All surface-level external noise sources require good noise isolation from the walls, and especially the external windows and doors. Aircraft noise requires attention to the roof construction. Since my client is located on a quiet cul-de-sac, surface-level external noise sources are not an issue. The occasional noise from aircraft flyover is adequately blocked by the asphalt shingle roof, second-floor gypsum ceiling, and the floor-ceiling assembly between the second and first floors. Thus, external noise was not an issue for this project.

As I complete this article, I have just communicated with a city government client who wants to convert a 1912-vintage building into a performing arts center. With other noise-generating businesses immediately adjacent, and two train tracks within 100' of the property, even though they plan to completely remodel the interior of the building, I had to advise them against putting a noise-sensitive venue in this location. Fortunately, they arranged for advice on acoustics before investing in what would almost certainly have been an unsuccessful project!

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An Interview with David Scheirman

By
Shannon Becker
(United States)

SHANNON BECKER: Tell us a little about your background.

DAVID SCHEIRMAN: I grew up in the midwestern United States. I was involved in a church music program and was fortunate that our choral program director was President of the American Choristers' Guild. This exposed me to world-class visiting expert performers, and also gave me the opportunity to attend Westminster Choir College in Princeton, NJ, as a teenager. At the same time I was involved in a school orchestral program, learning to play the acoustic bass. This led to lessons with the principal (first chair) bassist for the Oklahoma City Symphony. At the same time I was keenly interested in electronic circuits, chemistry, amateur rocketry and aviation, and a broad range of other hobby interests that required basic learning about scientific experiments and such.

SHANNON: How did you become interested in audio electronics?

DAVID: When I saw the Beatles perform on the Ed Sullivan TV show in 1964, I realized I already knew how to play the bass that Paul McCartney had; I just needed to transition to an electric one. I asked my

parents for a starter bass guitar and music-store amplifier. From then on, at-home experiments with electronic gadgets, 45 RPM rock-and-roll records, and my bass guitar performances with a local garage band doing parties and dances helped me learn about simple public address systems. This led to a lifelong interest in sound reinforcement.

SHANNON: How have your interests changed since you starting working in the audio field?

DAVID: Moving from live performances to technical support (as a concert sound technician, system operator, and live event sound designer) to product development was a natural transition. I was always quite interested in understanding how to improve audio equipment, how to achieve higher quality reproduction of a live performance for the audience, and perhaps most importantly, how to use the scientific method and apply test and measurement techniques and processes to take things out of the realm of personal opinions, into objective data assembly and fact-based decisions. That of course led to a requirement for other, different educational programs. Studying topics such as electro-magnetics, international economics, and strategic marketing for technology products



David Scheirman is the Global Director, Concert & Rental Business, for Bose Professional and the 2017-2018 President of the Audio Engineering Society.

became both an interest and necessity. I've been fortunate to have learned from leading experts in the fields through programs at institutions like CalTech, Massachusetts Institute of Technology (M.I.T.), and Stanford University.

SHANNON: Can you please share with us some of the most memorable high-profile events on which you have worked?

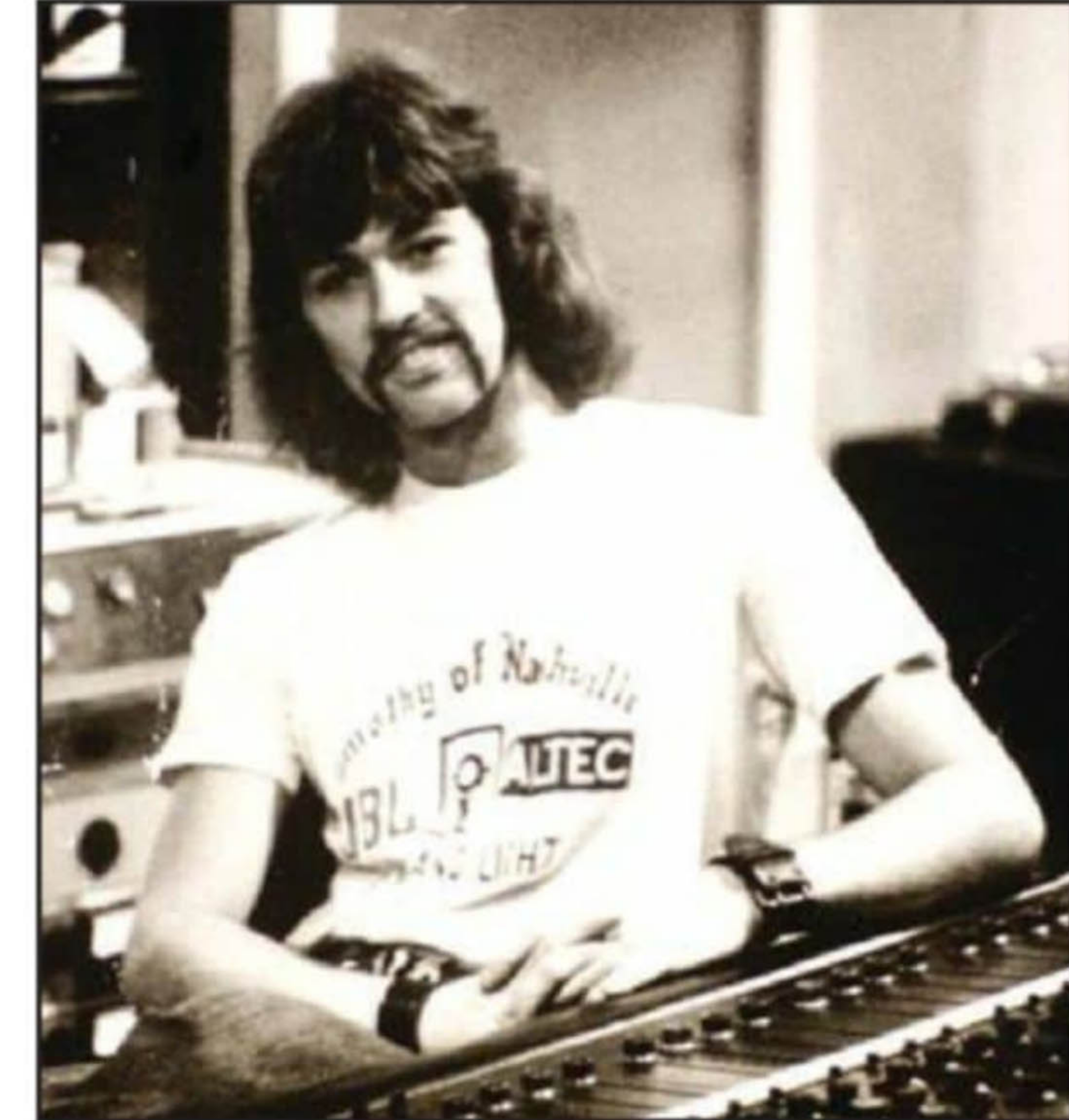
DAVID: As a concert technician and soundmixer, I was involved in more than 3,000 live performance events ranging in size from auditoriums to arenas to stadiums. These shows exposed me to some challenging sound reinforcement field applications around the world, from Africa to Asia to Europe to Latin America and Northern America. Participating with the United States' White House Communications staff on a Presidential Inauguration, working with the Vatican for very large scale audiences with the Pope, and being involved with audio for the Olympic Games were all stand-out events. Working with touring artists such as Linda Ronstadt, George Benson, Electric Light Orchestra, Al Jarreau, and many more was both challenging and rewarding.

SHANNON: I know you worked for JBL Professional and managed the tour sound market segment. How has the technology evolved over the years?

DAVID: Yes, I was Vice President, Tour Sound for JBL Professional and now am Global Director, Concert & Rental Business for Bose Professional. Audio technology for live entertainment sound reinforcement has seen a series of steady, incremental improvements since I first started stacking up simple speaker columns with tube amplifiers for high school dances 50 years ago.

The shift from analog to digital technology to improve user interfaces on system drive electronics and power amplification, and the adoption of advanced materials and technologies such as neodymium magnets and full-bandwidth line source speaker array designs, have dramatically changed the technical approaches taken for live sound in both fixed-installation and portable applications. These incremental evolutionary changes do take time. But sound systems have steadily gotten lighter, more powerful and able to operate in a more consistent manner, over time.

SHANNON: Can you tell us about your work as director of the global concert and rental business for Bose Corp.? What does your job entail?



David Scheirman is pictured at the Woodland Studio in Nashville, TN (1972).

DAVID: Working with the senior management team of the Professional Division, I am assisting the company in leveraging its broad portfolio of intellectual property, electronic and acoustic technologies, and its extremely competent research and development team to envision and develop next-generation sound reinforcement system products. This includes coordinating efforts of the marketing and product-line management team, the engineering and R&D group, and the sales and distribution-channel strategies in a global business development role. Bose Professional is a key part of Bose Corp., and with M.I.T. as an owner, we have unique access to best practices and product development resources that should



David Scheirman is pictured with a Harrison HM-5-FOH mixing console (1990).

As part of his work with Bose, David Scheirman is shown here in the Bose booth at the 144th Audio Engineering Society Convention in Milan, Italy.



help to provide some interesting new developments for the worldwide sound reinforcement industry in coming years.

SHANNON: What do you see as some of the biggest changes in the last few years?

DAVID: The development of reliable higher-powered amplifiers, the availability of component transducers with better power-handling capabilities, and the application of measurable acoustical principles to loudspeaker arrays have all dramatically improved available system products. Once upon a time, sound system deployers often had to design and build their own equipment. Today, there are many credible vendors providing useful tools these days.

SHANNON: What projects are you currently working on?



David Scheirman gives a lecture in 2016 at the AES Brazil Expo in 2016.

DAVID: At Bose, our first exploratory line array system for the performance-venue and live entertainment rental sound industry is called ShowMatch. A goal of this compact line array system is to provide increased SPL output capabilities and lower distortion in a roadworthy system package, when compared to other alternatives. It has a high ratio of high-frequency drivers to the low-frequency component transducers, and employs a crossover point about one octave lower than other systems of similar size and shape. Introducing the benefits of this new system to the worldwide rental and staging market is a current active project. ShowMatch is ideal for integrated A/V rental production companies and other rental businesses

SHANNON: In 2017, you were elected president of the Audio Engineering Society (AES). When and how did your first get involved with AES?

DAVID: I first joined the AES while living in Nashville, TN, in my early twenties. I've been a member for more than 40 years. I can honestly say that many of the most interesting, challenging and profitable sound reinforcement projects of my career have come about due to referrals to me from other AES members over time. The Society currently numbers nearly 13,000 members worldwide, and we welcome other professionals who share common goals and interests.

SHANNON: What are some of the challenges you faced as AES president?

DAVID: Coming from a corporate business-management background, I have worked with AES colleagues to examine some of the underlying assumptions about operational structure and our financial model. As just one example, our membership recently approved a Bylaws change to align our fiscal year with our elected officers' terms. It is best practice to create synergy between an organization's corporate goals and priorities, and its annual budget. In addition, in response to a recent resignation, I appointed an executive director search team. The Society worked with Stanton Chase, a global top-ten Forbes-listed executive recruiting firm to identify suitable candidates.

SHANNON: How would you like to see AES evolve?

DAVID: The Audio Engineering Society has invested a good deal of energy and resources in engaging the student community worldwide. I believe our Education and Membership Committees will

envision and execute on new ways to reach the growing community of entry-level audio enthusiasts worldwide. I hope to see the emergence of AES-sponsored learning tools to assist in career development for the next generation of audio professionals. I am certain that many of today's young audio enthusiasts will become a force to be reckoned with in the worldwide professional audio industry. I look forward to seeing the AES grow and change and evolve with the many new industry trends such as audio for gaming, virtual and augmented reality, "hearables," and other mobile audio applications that serve transportation systems and public spaces, and more.

SHANNON: One last question. Where do you see the audio industry being in 10 years?

DAVID: My personal observation is that the whole world wants better sound, and more of it. Whether you examine sound for live events, for commercial applications in public spaces, in the home or in our cars or just as individuals "on the move," the prospects for new, innovative audio devices and products and systems and experiences are bright.



David Scheirman served the Audio Engineering Society as its president, 2017-2018.

A decade from now I believe we'll see more tightly integrated software service bundles coupled with lighter-weight, more articulate, and more user-friendly audio systems. Component transducers will become lighter, smaller, and offer better fidelity. The "pro audio" industry has a lot to teach and offer the consumer electronics industry, and vice versa.

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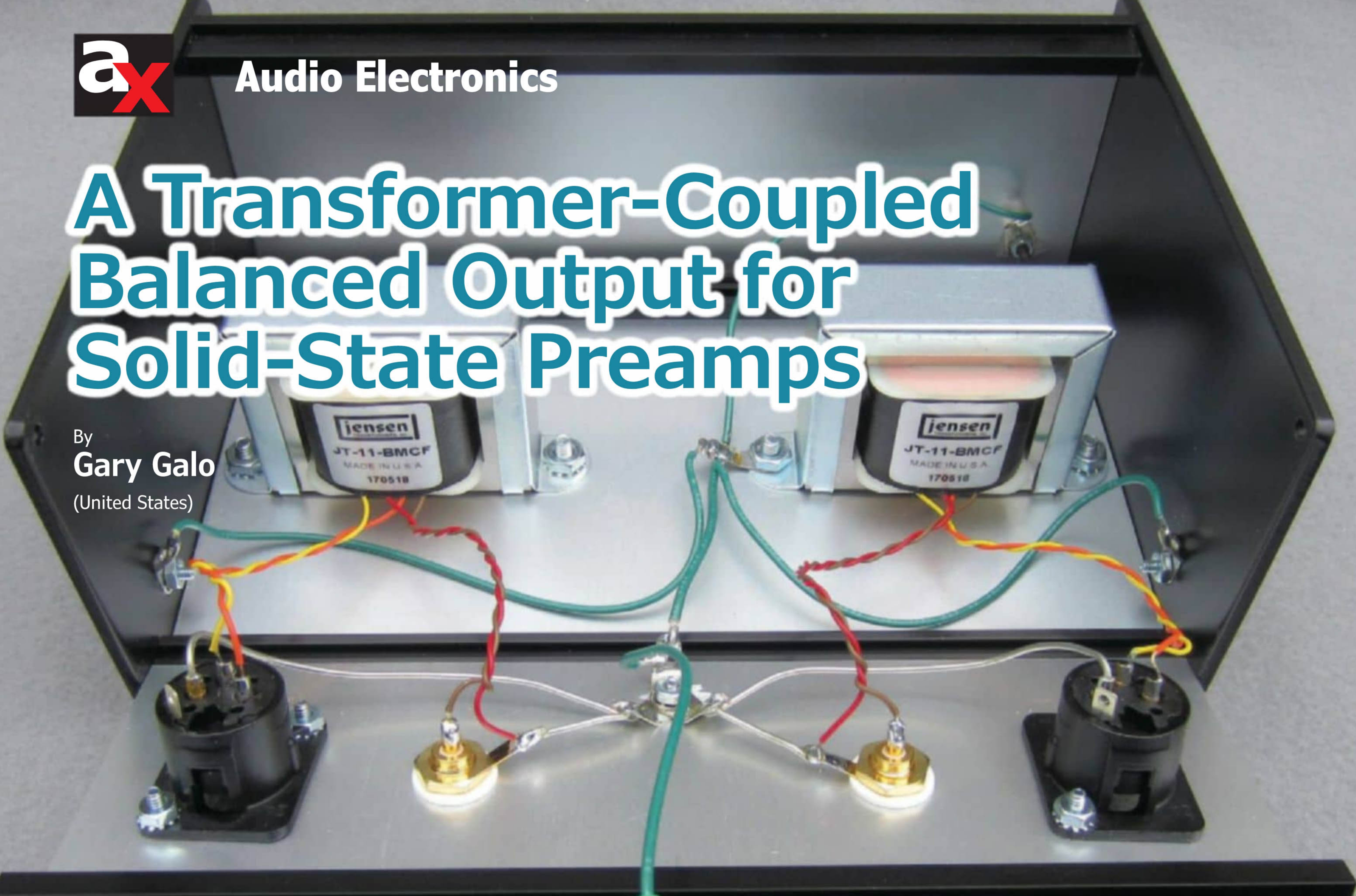
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A Transformer-Coupled Balanced Output for Solid-State Preamps

By
Gary Galo
(United States)



Our longtime contributor Gary Galo explores the differences between balanced interfaces and unbalanced interfaces. Then, he explains why he selected Jensen transformers for his own high-end system.

Balanced interfaces have long been standard practice in the pro-audio world, both for microphone-level and line-level signals. Before explaining the difference between unbalanced and balanced interfaces, let's clarify two often-misunderstood terms: phase and polarity. To understand the difference, consider the fact that musical waveforms are normally asymmetrical. The

Figure 1: The difference between phase and polarity is best illustrated using asymmetrical waveforms. The upper illustration shows a phase difference—waveform B lags waveform A by 180°. The two waveforms in the bottom illustration are of opposite polarity.



positive-going portion of the waveform is rarely a mirror image of the negative-going portion. If you view two continuous sine waves, or other test tones, on an oscilloscope, the symmetrical nature of the waveforms makes it impossible to distinguish between a phase difference and a polarity difference. The asymmetrical waveform in **Figure 1** makes this much easier to visualize. Phase refers to the time relationship between two audio signals. In the upper illustration shown in Figure 1, waveform B lags behind waveform A by 180°. Conversely, waveform A leads waveform B by 180°. This is a phase difference. In the lower illustration shown in Figure 1, waveform B is inverted in polarity relative to waveform A. This is what occurs when you reverse the leads to one of the loudspeakers in your stereo system. We normally refer to the two speakers as being out-of-phase, but this is technically incorrect—they're really of opposite polarity.

Balanced lines are preferred in professional audio applications because of their superior noise rejection. **Figures 2–5** illustrate the concepts behind unbalanced and balanced interfaces. They are not intended to be complete, working circuits. For the purposes of illustration, I've treated each interface as unity-gain. What's important here are the relative polarities of the audio signals and, especially, the noise. As additional reading, I recommend Jensen Transformers' Application Note AN-003, "Interconnection of Balanced and Unbalanced Equipment," by long-time company president Bill Whitlock. Whitlock retired from Jensen in 2014, but the new parent company, Radial Engineering, has retained him as a consultant. I'll refer to that document several times in this article, along with other correspondence I've had with Whitlock.

Figure 2 shows an unbalanced interface between an audio source and a load. The source could be a preamplifier or mixer, and the load might be a power amplifier or audio recorder. The cable connecting the two pieces of equipment consists of a single signal carrier surrounded by a shield. The audio source has a single-ended output, which feeds the audio along the center conductor to a single-ended input in the receiver.

Unfortunately, even the best shielding is never 100% effective in preventing the center conductor from picking up electromagnetic radiation. Indeed, the shield is often the source of the noise. In AN-003, Whitlock notes: "The inherent weakness of unbalanced interconnections is that the shield, which is also a signal conductor, is a path for power-line related currents that always flow between equipment grounds. The voltage drop across the resistance of the shield and connectors adds directly to the signal, producing the familiar hum and buzz."

Hum and other noise picked up by the center conductor will be amplified by the receiving equipment along with the audio fed from the source. The longer the cable, the worse the problem. This can be especially problematic with microphone-level signals, which are typically only a few millivolts. Unbalanced interfaces are completely unacceptable for professional, microphone-level interfacing, and are normally avoided even with line-level applications.

Figure 3 shows a balanced interface. The audio source feeds a pair of signal carriers from an active, differential output, which provides signals of opposite polarity to the two conductors. These opposite-polarity signals are fed to an active, differential input in the receiving equipment. The differential input receiver has both non-inverting and inverting inputs. When the opposite-polarity

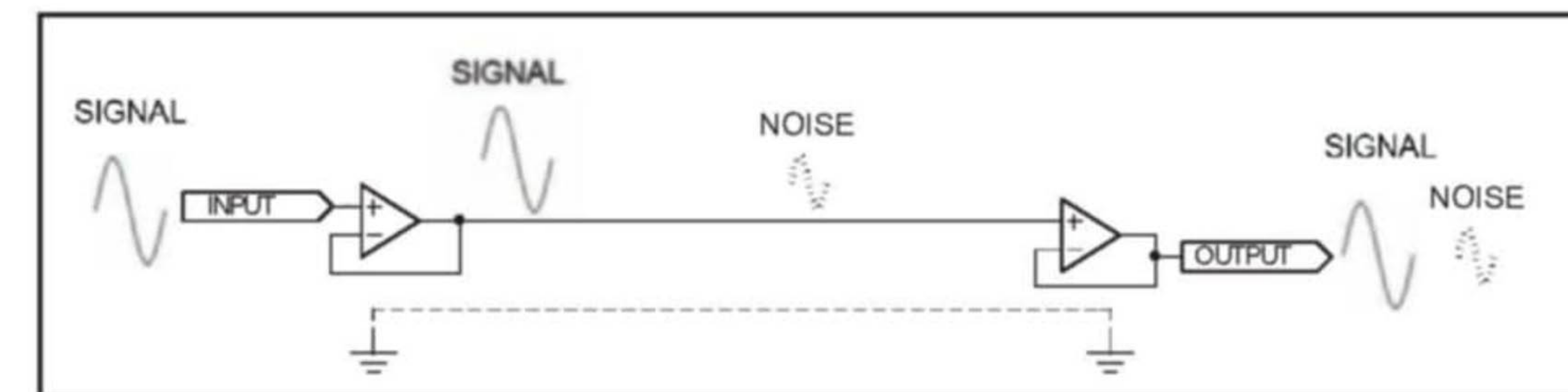


Figure 2: An unbalanced interface has a single signal carrier and a shield. Any noise picked up by the signal carrier will be passed to the receiving equipment along with the audio signal.

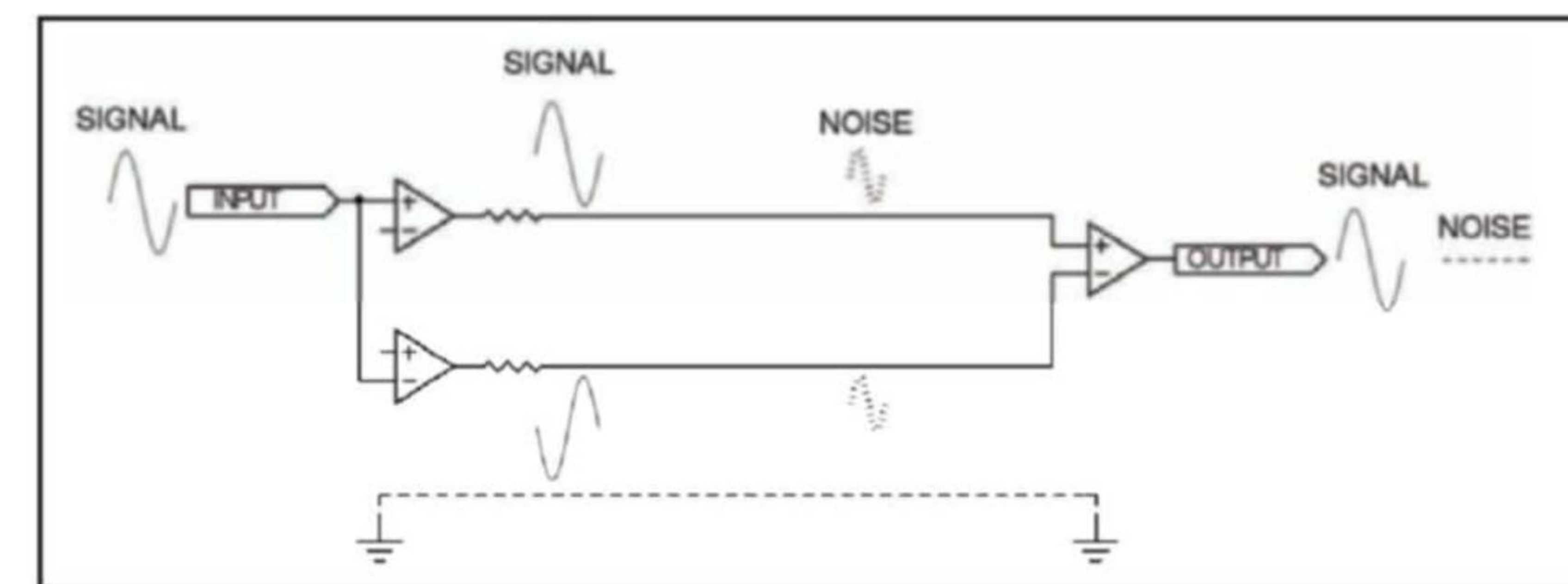


Figure 3: A balanced interface usually feeds audio signals of opposite polarity to a pair of conductors. Noise picked up by the cable will be the same polarity in the two conductors, and will cancel when connected to the differential input in the receiving equipment. Good common-mode rejection is dependent upon matched impedances in the two audio lines, and not signal symmetry. A balanced interface can be made with discreet components, IC op-amps, or various IC line drivers and receivers designed for that purpose.

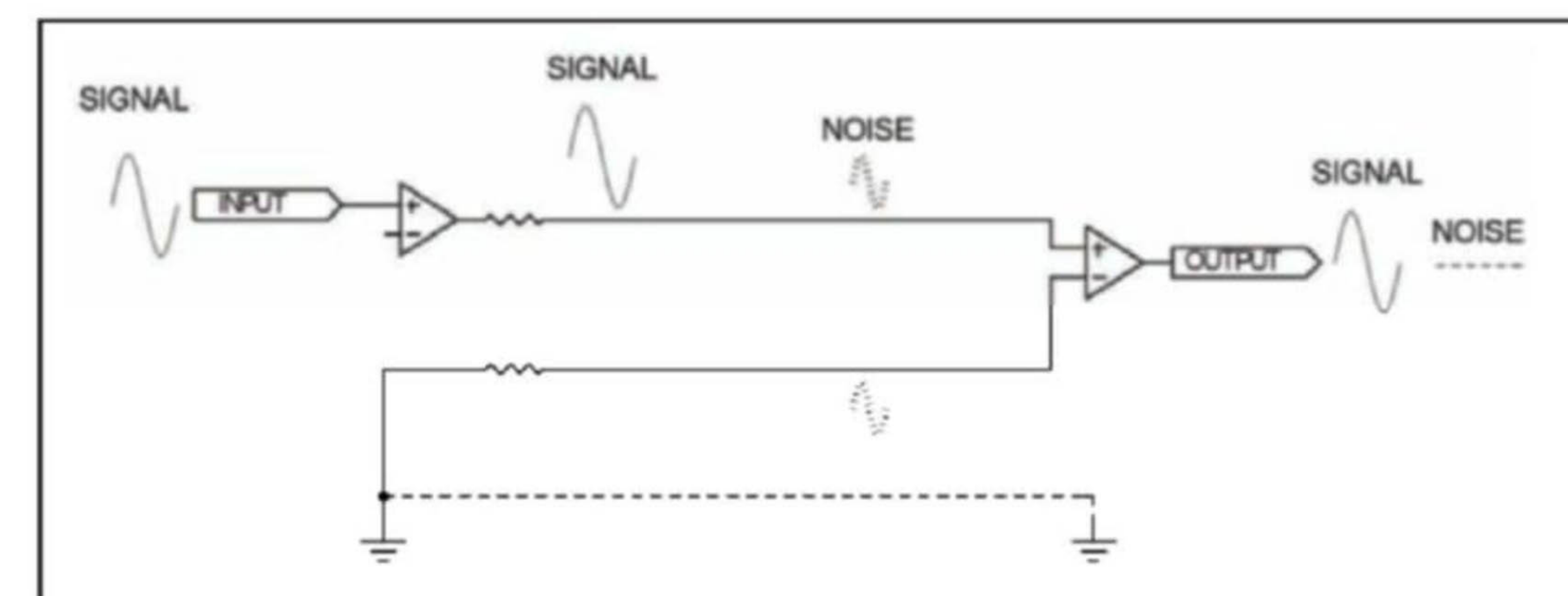


Figure 4: A true balanced line can be made with only one signal carrier, but still requires two conductors. Noise will still cancel as long as the impedances of the two conductors are matched. This scheme has been used in applications where professional operating levels are not essential, and is often incorrectly called a quasi-balanced or pseudo-balanced line.

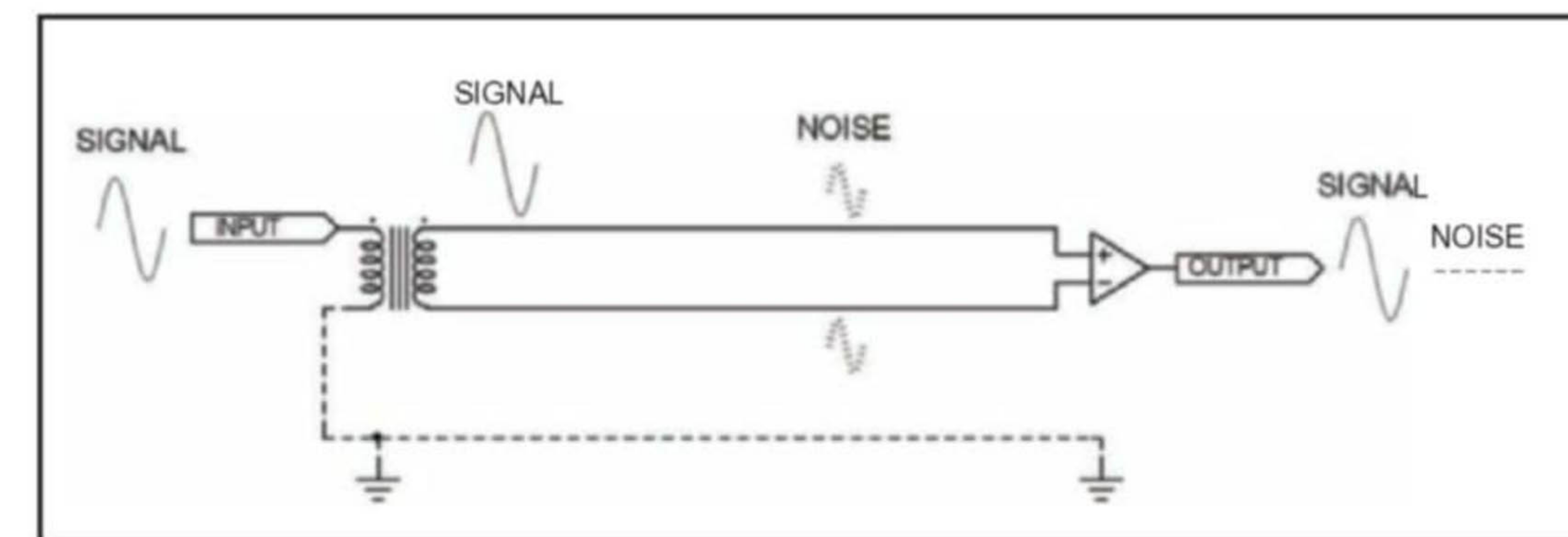


Figure 5: In a balanced interface driven by a line-output transformer, signal is present on only one end of the transformer secondary. This is due to the capacitive coupling between the windings of the transformer, which holds the "low" end of the secondary at the same signal potential as the corresponding end of the primary winding. This is still a true balanced interface due to the matched impedances in the two lines. Noise will still cancel when the two conductors are connected to the differential input in the receiving equipment.

signals are fed to the differential input, they combine as though they are the same polarity. Like the unbalanced interface, the two signal carriers are surrounded by a shield. Noise picked up along the way will be the same polarity in each conductor, and is known as a common-mode signal. Since these same-polarity noise components are fed to the differential input, they combine as though they are of opposite polarity, resulting in cancellation of the noise. The ability of a balanced interface to cancel noise is known as common-mode rejection. Active balanced interfaces can be designed around discrete components, IC op-amps, and specialized line-driver and line-receiver chips.

In 2005, Whitlock gave a superb presentation on grounding and system interfacing at the New York convention of the Audio Engineering Society. I wrote a detailed report on his presentation for the January 2007 issue of *audioXpress*. One point Whitlock drove home was that in order to qualify as a balanced interface, the two conductors must be matched in impedance relative to ground. The more closely the impedances are matched, the better the common-mode rejection. But, he also noted that the two conductors need not both carry the audio signal.

Balanced Lines

Figure 4 shows an active, balanced interface where only one conductor carries the audio signal. At the source component, the second conductor is grounded through a resistor. The resistor value is selected to ensure that the source impedance of this conductor is the same as that of the active device driving the signal carrier. Normally, this resistor will be the same value as the resistor in series with the output of the active device. Noise picked up by the two conductors will cancel at the differential input, as before. The only disadvantage of having just one signal carrier is a 6 dB reduction in combined signal level, so they're normally not used where professional operating levels are required. However, they'll be fine in most home audio systems. This type of interface has been used by at least one manufacturer of pro-audio mixers for auxiliary sends, and even main outputs on their least expensive models. It's sometimes called a "quasi-balanced" or "pseudo-balanced" interface, but those terms are misleading, if not downright incorrect. As Whitlock points out in AN-003, this is a true balanced output, since it provides correct impedance matching of the two conductors, which is all that's necessary for noise rejection.

A balanced interface can also be made using transformers. Figure 5 shows a balanced interface using a line output transformer feeding an active,

differential input receiver. Intuitively, one would think that a line output transformer would feed symmetrical audio signals to the two conductors, equal in level but of opposite polarity. However, this isn't the case. The primary winding of a line output transformer is normally driven by applying audio signal to the "high" side of the primary. The "low" side is normally connected to signal ground. If you put your oscilloscope probe on the "high" side of the transformer secondary—the output—you'll see the audio signal at full level. But, if you connect your probe to the low side, you'll see nothing. The transformer's "high" and "low" outputs appear to be floating, so how can this be?

I posed this question to Whitlock, who provided a most illuminating answer (note that Jensen color codes the "high" sides of its JT-11-BMCF transformer primary and secondary windings brown and orange, and the "low" sides red and yellow, respectively). Whitlock said, "Because it's an output type transformer, there is a relatively large capacitance between primary and secondary, 36 nF for the Jensen JT-11-BMCF. Since this capacitance is distributed uniformly across the windings, a first-order approximate equivalent circuit would see about 18 nF at each end (between BRN and ORG and between RED and YEL, for example). Therefore, with high-impedance balanced loads, the voltages at ORG and YEL will essentially be identical to those at BRN and RED—it effectively operates as a pair of coupling capacitors. Only when the load impedances (from each end to ground) are quite low (300 Ω each, for example), will the secondary voltages become symmetrical (which is not the same as balanced, contrary to widely-held belief)."

Whitlock went on to clarify the requirement for a balanced interface: "In a practical interface, this behavior is of no consequence—noise rejection in an interface does not arise from signal symmetry. When we wrote the 'informative annex' to the IEC standard 60268-3 test for common-mode rejection, we emphasized this point. What makes a balanced interface 'balanced' is not symmetrical signals, it's equal ('balanced') impedances from each leg to ground. Symmetry of signal on the two legs has advantages (more differential signal level is possible, and radiated crosstalk from an unshielded cable is reduced), but they have nothing to do with noise rejection ... and that's the reason we use balanced interfaces in the first place."

Whitlock also explained the differences between output and input transformers, which are widely misunderstood: "The most striking difference is due to the inclusion of a Faraday shield (a layer of copper foil between primary and secondary windings) in

input types. When this shield is grounded (to the signal reference ground for the secondary), capacitive coupling between the primary and the secondary is virtually eliminated. This ultra-low capacitance is absolutely essential to achieving high common-mode rejection ratio (CMRR) for an input transformer.

"But there's a catch: because primary and secondary are now physically separated, some of the magnetic field generated by the primary isn't captured by the secondary—this looseness in coupling is measured as 'leakage' inductance (several mH or more is typical). In an equivalent circuit model, this inductance is effectively in series between primary and secondary. This wouldn't necessarily be a problem unless the load on the secondary has either a low resistance or a high capacitance, or both. Therefore, input transformers will exhibit severe high-frequency response losses if loaded with a high capacitance (over a few hundred pF, generally). This makes them eminently unsuited for use as an output transformer, where they may be required to drive a hundred feet of cable!

"On the other hand, output transformers are wound to put each turn of the primary in the closest possible proximity to corresponding secondary turns (multi-filar construction). This makes magnetic coupling very tight and 'leakage' inductance extremely low (often just a few μ H). This makes them suitable for driving long runs of cable with little or no loss of high-frequency bandwidth. But it's this intimacy of primary and secondary that creates high capacitance between primary and secondary, and the side-effect you noted with signal symmetry," Whitlock concluded.

In Figure 5, the audio signal appears on only one conductor, just as it does in the actively-driven example in Figure 4. The transformer secondary winding provides matched impedances to each leg of the balanced line. As long as the input impedances of the differential input receiver are also matched, noise will cancel just as it does in Figure 3 and Figure 4. At 60 Hz, a high-performance transformer can have a common-mode rejection ratio in excess of 100 dB. A line-input transformer could be substituted for the active, differential input. High-performance line-output transformers are capable working at professional operating levels, given sufficient input signal from the active source.

Why Balanced?

Most consumer audio equipment is connected with unbalanced lines, using single-conductor shielded cables and RCA connectors. The short cable runs encountered in a domestic environment hardly ever cause audible noise, so audio manufacturers have traditionally found balanced lines unnecessary. In recent years, balanced connections on stereo preamps and power amplifiers have become more common. Most modern audio power amplifiers with balanced inputs have true differential input stages, so optimum performance is not dependent on signal symmetry at the balanced input. The issue is noise. Modern audio power amplifiers have lower inherent noise levels than their predecessors, and will perform best if they are fed from a balanced source.

Benchmark is an example of a manufacturer that offers only balanced XLR inputs on its superb AHB2 power amplifier. This ultra-quiet amplifier has a signal-to-noise ratio of >130 dB across

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the audio spectrum, and an input CMRR of 80 dB up to 1 kHz, and 65 dB at 20 kHz. The limiting factor, as far as noise is concerned, will be the source equipment and the interface between that equipment and the AHB2. Benchmark strongly recommends a balanced interface in order to make the most of this amplifier's exceptional noise performance.

Whitlock offered these observations on signal symmetry, noise and related issues in power amplifiers: "For 'most' power amplifiers (the ones designed by engineers who understand that a balanced signal is not necessarily symmetrical), the answer is that it doesn't matter. However, there are quite a number of power amplifiers whose design is 'balanced' (as if it were a feature) in that the signals on pins 2 and 3 of the input XLR feed two identical, single-ended amplifiers that drive the push-pull output tube grids (every one of these miscreants I've seen is a vacuum-tube design). Most importantly, there is no differential amplifier to eliminate the common-mode input voltage. If such amplifiers are fed by a balanced line with no signal on, say, pin 3, one output tube fights with the other. Worse yet, if there is common-mode voltage at the XLR, both output grids see it as in-phase ... which causes abnormally high peak plate currents since the primary of the output transformer is essentially a short circuit when driven with the same signal on both sides of its center-tap. The result is very high and awful-sounding distortion. The cure should be a redesign by the manufacturer, but it can be cured by an input transformer having a center-tapped secondary (Jensen makes an ISO-MAX for this) inserted at the amplifier input. It not only removes the common-mode signal, but produces the phase inversion that the designer apparently assumed was a required characteristic of 'balanced' signals. This myth ruins designs more often than you might think ... I've seen it on some very expensive gear!"

Back in the days when we played only analog sources, our audio systems were largely free of digital noise. But, digital players, digital servers, and outboard digital-to-analog converters (DACs), can radiate digital noise that will be picked up by analog interface cabling. Switching-mode power supplies, which are becoming more prevalent in professional and consumer audio equipment, can also radiate noise, and sometimes put noise back out on the power cables that feed them. Noise from digital equipment and switching power supplies can begin at audible frequencies, and often extends into the radio-frequency spectrum. It's normally low enough in level to be inaudible as noise from

your loudspeakers. But, high-frequency noise can contaminate audio signals, resulting in a loss of detail and even grungy sound. Balanced interfaces can help alleviate these problems.

Belden engineer Steve Lampen has written an interesting article on the rejection of EMI and RFI by balanced lines, which is available on Belden's website (see Resources). Lampen describes a situation where the RFI from a large antenna farm for San Francisco, CA-area TV transmitters rendered a nearby audio system unusable. Converting the equipment interfaces to balanced lines, using "expensive transformers," solved the problem—"clean as a whistle," he notes.

In 2016, Paul McGowan, CEO of PS Audio, offered a series on balanced interfaces in his daily e-mailing "Paul's Posts." This multi-part series ran between November 20 and December 4 of that year. I've provided the link to the first part in the "Resources" section—at the end of each section you can click on the arrow to advance to the next one. McGowan began by noting that, like most of us, he had long used unbalanced interconnects in his system even when balanced inputs and outputs were available on the equipment he manufactured. After trying balanced cables, he became convinced that "they just sound better." McGowan offers his own, always-interesting perspectives on this issue.

I have my Audio Concepts Sapphire III/Sub-1 loudspeaker systems passively bi-amped with a pair of AHB2 amps. When I ran my Benchmark power amps from an unbalanced source, I never heard any audible noise from my loudspeakers—no hum, buzz or other anomalies. My system was audibly silent under no-signal conditions. Having used them both ways, I totally agree that they sound better when operated in the balanced mode.

Balanced Output Choices

Active interfaces using integrated circuits (ICs) can be built for very low cost, and have become ubiquitous in professional audio equipment. These include circuits designed around IC op-amps, and specialized line-driver and line-receiver chips. If you're designing a preamplifier from scratch, or have the ability to modify one internally, there are many options for adding a balanced line driver to your preamp. Walt Jung offered several balanced line driver topologies in the two applications handbooks published by Analog Devices in 1993 and 2005. Fortunately, these invaluable volumes are now available as downloads on its website (see Resources).

Several manufacturers of audio ICs have made balanced line driver and receiver chips. These include

Analog Devices' SSM series, and Texas Instruments' (TI) DRV and INA series, along with THAT Corp.'s OutSmarts 1606 and 1646 balanced line drivers and InGenius 1200-series of balanced line receivers. I wrote two articles on THAT's chips for *audioXpress*—those articles are still available (see Resources). TI offers two fully-differential audio op-amps, the LME49724 from its National Semiconductor line, and the more recent OPA1632, from its Burr-Brown line. The OPA1632 has impressive specs, including a slew rate of 50 V/μSec, a gain-bandwidth product of 180 MHz, a THD of 0.000022%, and noise of 1.3 nV/√Hz. OPPO Digital uses the OPA1632 for the balanced outputs on its UDP-205 digital player (see *audioXpress*, November 2017). On paper, at least, the OPA1632 may be the best single-IC solution for active, high-performance balanced outputs.

The above choices all provide signal symmetry in addition to impedance-matched balanced interfacing. If professional operating levels are not required, signal symmetry probably isn't either. In these cases, the topology shown in Figure 4 will work just fine. Whitlock highly recommended this option in our correspondence, and describes its implementation in AN-003. Note that the source impedance provided to XLR pin 3 must precisely duplicate the preamp's active output.

In addition to the output series resistor, some preamps may have an output coupling capacitor terminated with a load resistor. This should be duplicated for the other leg of the balanced line, as shown in the illustration on page 3 of AN-003. The added resistor and other components, if needed, should be mounted as close to their counterparts at the line stage output as possible, and connected to signal ground close to the active circuitry. I would avoid the temptation to connect them to any convenient chassis or signal ground on or near the rear panel. PC trace or other ground lead inductance and resistance between the added components and the active circuitry's signal ground will change the impedance, unbalancing the line.

Transformer Option

There will be instances where it's not possible to mount XLR connectors, additional passive components, or an active balanced line driver, in an existing preamp chassis, due to lack of space. It's certainly possible to house active circuitry in an external enclosure, but this will require building a power supply, which may mean another power transformer, rectifiers, regulators, and AC line cord. If you're lucky, you may be able to tap suitable power supply voltages in your preamp, but I would still recommend local regulation in the external

enclosure. If you're using the method shown in Figure 4, I don't recommend housing the extra components in an external enclosure, for the reasons cited above.

If this sounds a bit cumbersome, there's a much simpler solution that can offer outstanding performance: a line-output transformer. A high-quality line-output transformer can offer performance rivalling active circuitry, but there are disadvantages, namely physical size, weight, and cost. High-performance audio transformers aren't cheap, and you definitely get what you pay for. My goal was to build an outboard box that could be connected to the unbalanced outputs of any solid-state stereo preamp with a sufficiently-low output impedance, using a very short interconnect. After consultation with Dave Hill, General Manager at Jensen Transformers, I decided on the company's very-best line-output transformer, the JT-11-BMCF mentioned earlier in this article.

Physically, the JT-11-BMCF is Jensen's largest output transformer. It has an 80% nickel alloy lamination core, which provides very low distortion at low frequencies. It will also drive 600 Ω loads, though this should not be of any concern in the modern audio world.

As Whitlock pointed out in his 2005 seminar, 600 Ω impedances are a holdover from early telephone practice. Modern audio circuits are designed with low output impedances and high input impedances, making performance with 600 Ω loads irrelevant. Nonetheless, manufacturers still feel the need to make sure their products will perform into 600 Ω loads, despite the fact that this ceased to be an issue decades ago.

Grounding and Capacitive Loading

Before beginning construction, I had several questions for Hill. The first concerned grounding. Should signal ground from the unbalanced input be tied to pin 1 on the balanced output? Or, should the input RCA jacks' shields be connected only to the lower end of the transformer primary, and not tied

Parts List

(2) Jensen JT-11-BMCF Line-output transformers (Jensen Transformers, Inc.; Markertek)

(1) Connectronics CTX-MC-6A Aluminum Project Cabinet (Markertek)

(2) Neutrik NC3MD-L-1-B, 3-pin male XLR connectors (Markertek)

(2) D.H. Labs CM-R1 Chassis-mount RCA Jacks (D.H. Labs; Parts Connexion)

Miscellaneous: Machine screws, nuts, lock washers, and ground lugs

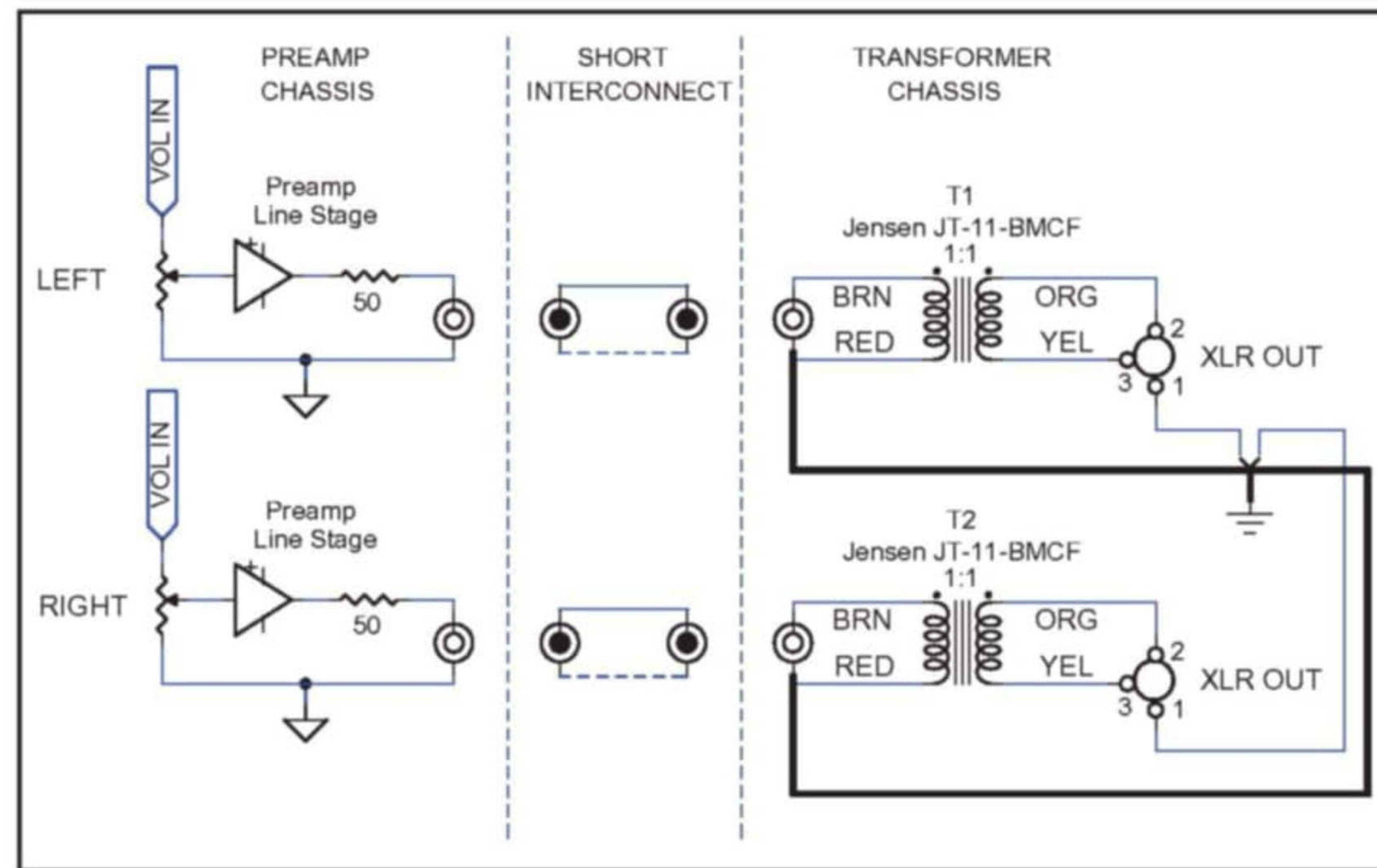


Figure 6: This is the complete circuit showing the unbalanced source, short interconnect, and the external unbalanced-to-balanced transformer. The JT-11-BMCF is Jensen's finest line-output transformer.

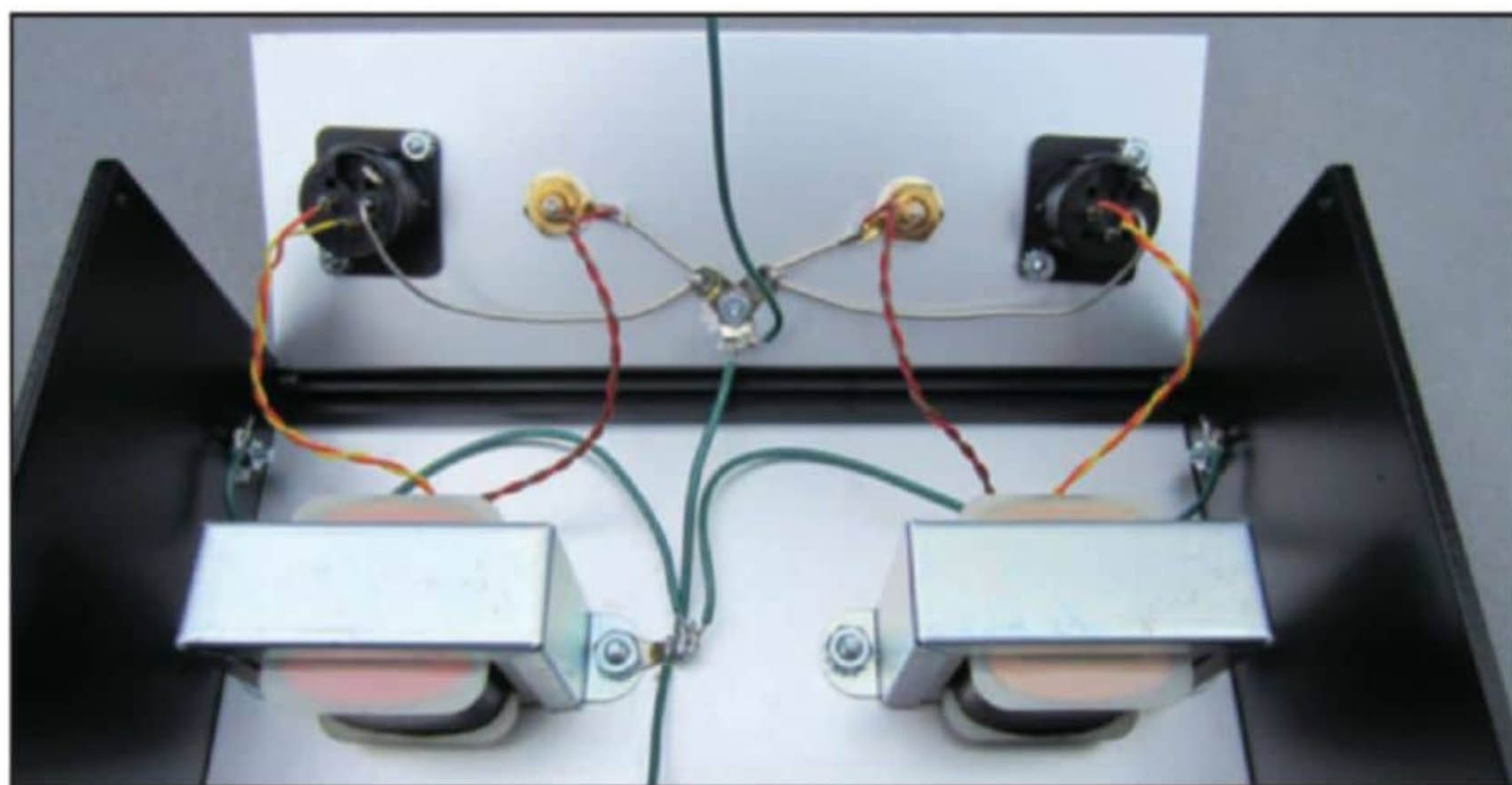


Photo 1: This is an inside view of the Connectronics CTX-MC-6A enclosure housing the two Jensen JT-11-BMCF line-output transformers. The star ground is in the center of the rear panel, along with the D.H. Labs RCA jacks and Neutrik XLR connectors.

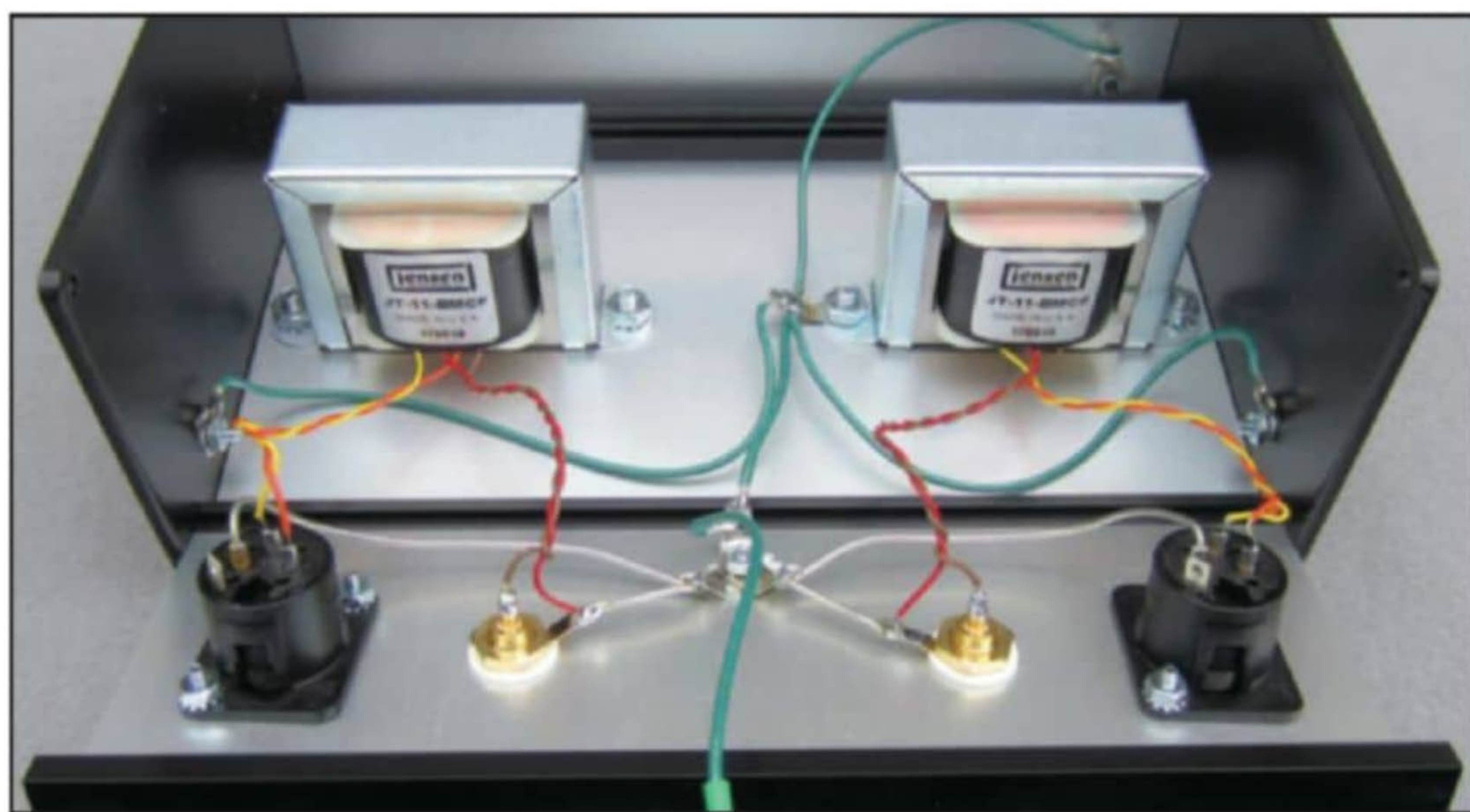


Photo 2: Here's another inside view of the enclosure. Connecting the top, bottom, and sides of the enclosure ensures that all pieces are at ground potential. The anodizing should be scraped off wherever a ground lug is used.

to chassis ground and pin 1 of the output XLRs? I was thinking of connecting the I/O grounds through a "Ground Lift" switch that would allow it to be used either way.

Hill responded as follows: "You bring up some important questions regarding grounding when using transformers. The rules are different when you are mounting a transformer in an external box than they are when you install them directly inside the equipment. The balanced-output cable's shield that connects to Pin 1 of the external box's output XLR's should also be connected through that box back to the chassis of the source equipment. To do this, the Pin 1 connection inside the box should carry over to the input RCA's sleeve connection, so that Pin 1 has a hard connection all the way through to the source equipment's chassis. If you are using RCA connectors that are isolated from the chassis, then you should also connect the metal box to the input RCA's sleeve, so the box itself gets grounded back to the source equipment's chassis. In your application, a ground lift switch would not be necessary."

Most audio preamplifiers protect the line stage from the effects of capacitive loading, which can cause oscillation in audio circuitry, particularly IC op-amps. A low-value resistor in series with the preamp's output is usually sufficient. Jensen makes a parallel L/R network for this purpose, the JT-OLI-3.

The JT-OLI-3 is a 39 Ω resistor with a coil of enamel-coated wire wound around the resistor body to form the parallel inductor. The impedance of the network is 0.3 Ω across the audible spectrum, but rises to around 40 Ω in the RF region. In its datasheet, Jensen recommends using two of these networks with the line-output transformer—one in series with the signal input to the primary winding, and the other in series with the ground side of the primary.

I asked Hill if this network was necessary for my project and he assured me that it wasn't. The Jensen isolation network was designed to protect output op-amps from the effects of very long cable runs, which can be several hundred feet in professional applications. With the cable runs encountered in most home installations, the preamp's single internal series output resistor is all that's necessary.

In high-end, solid-state designs this output resistor may be anywhere from 20 to 100 Ω , but in cost-effective products I've seen values as high as 475 Ω . In my own preamp, this resistor is 50 Ω . I asked him about the effect of higher source impedances on the performance of these transformers.

Hill said, "The resistor value does have a very minor effect on the low frequency response, but only down so low we're not able to even measure it. The 80% nickel alloy laminations in the JT-11-BMCF are the least sensitive to source impedance, when compared to 50% nickel alloy laminations. Our standard recommendation for the source impedance driving our 80% nickel alloy transformers is to keep it under 200 Ω . With 50% nickel alloy, we recommend keeping it under 100 Ω . With both types, the lower the source impedance, the lower the measurable THD will be at 20 Hz."

Although high-performance solid-state preamps generally have output impedances that are well within the range recommended by Jensen, there are also tube preamps that meet these requirements. Some tube preamps have output impedances higher than 200 Ω . Check the manufacturer's specifications on your particular preamp to be sure.

I also asked Hill about the physical mounting of the transformers in a stereo application, to minimize crosstalk. He notes: "Our engineer said that mounting them end to end would be best. In cases where they have to be mounted side by side, we recommend mounting one at a 90° angle to the



Photo 3: This is the rear view of the enclosure, showing the unbalanced RCA inputs and the balanced XLR outputs.

other, if space permits, to minimize the chance of crosstalk between the two.

Crosstalk becomes a concern when the transformers are being driven hard enough to approach saturation. 1% THD at 20 Hz is where we spec our transformers to have their saturation point. On the JT-11-BMCF that point is +27 dBu (with a 0 Ω source). With a 50 Ω source, that point moves down a bit to around +25 dBu. So your preamp's output section would have to be able to produce a signal level above +20 dBu at 20 Hz to be able to

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drive the output transformers hard enough towards saturation.” It’s reassuring to consider that the line stages of most consumer preamplifiers will never even approach the level required to saturate these transformers.

Hill added, “Also, we recommend that you tightly twist the brown and red primary wires together, and the orange and yellow secondary wires together, trimming them to the length you need without leaving a lot of excess, and then untwist

the ends only enough to make your connections. Don’t twist the brown/red & orange/yellow bundles together; leave them as separate twists. Twisting the two bundles together will capacitively couple the primary and secondary windings together at high frequencies, which will defeat the isolation properties of the transformer.”

For those who might want to fit a pair of these transformers into an existing preamp chassis, Hill offered the following advice: “The more important concern when mounting our Jensen output transformers into a piece of gear is to try and keep them physically away from the internal power supply section to avoid magnetic hum pickup.”

Construction

For this project, I used one of the Connectronics MC-series aluminum project boxes, readily available from Markertek. The CTX-MC-6A is exactly the size I needed—it’s 8” wide, allowing enough physical spacing between the transformers to minimize any crosstalk. **Figure 6** shows a schematic diagram of the complete system. A typical unbalanced solid-state preamp line stage is shown on the right, with an output impedance of 50 Ω . Short, unbalanced interconnects in the center connect the preamp’s outputs to the external transformer chassis.

My interconnects are 12” long, which I made myself. A half meter is usually the shortest you can buy ready-made, which should be fine. The input RCA grounds and pin 1 of the output XLR connectors are all tied to a single, star ground on the rear panel. I recommend using RCA jacks that are insulated from the chassis. If you allow the RCA jacks to contact the chassis, you’ll defeat the star ground.

Photo 1 and **Photo 2** show inside views of the transformer box. The side pieces of the Connectronics enclosures are painted; the remaining four pieces are anodized. In order to ensure that the entire enclosure is at ground potential, I recommend a ground lug on each piece, with wires tying the pieces together. Use the star ground point to connect the rear panel to the other pieces.

Be sure to scrape or sand the paint and anodizing off of each piece, where the ground lug is mounted. I used a push-on connector to connect the ground lug on the top plate, which allows easy removal and installation. I prefer D.H. Labs CM-R1 chassis-mount RCA jacks and Neutrik NC3MD-L-1-B, 3-pin male XLR connectors, but builders can certainly choose their personal favorites. **Photo 3** shows the rear panel with the RCA inputs and XLR outputs.

Performance

Figure 7 shows the frequency response graph published in the Jensen datasheet. The response is ruler-flat from 20 Hz to 20 kHz. On the low end, the -3 dB point is 0.15 Hz. The top end shows a -3 dB point just below 200 kHz, followed by a rise above that frequency. This transformer is an ultra-

wide bandwidth design, with Jensen specifying the ultimate -3 dB points at 15 MHz. **Figure 8** shows the response I measured, from 10 Hz to 1 MHz, with both 50 and 200 Ω source impedances. With a 50 Ω source Z, the response is -0.006 dB at 10 Hz. With the source impedance at 200 Ω , this drops to -0.077 dB, a negligible difference. From 20 Hz to

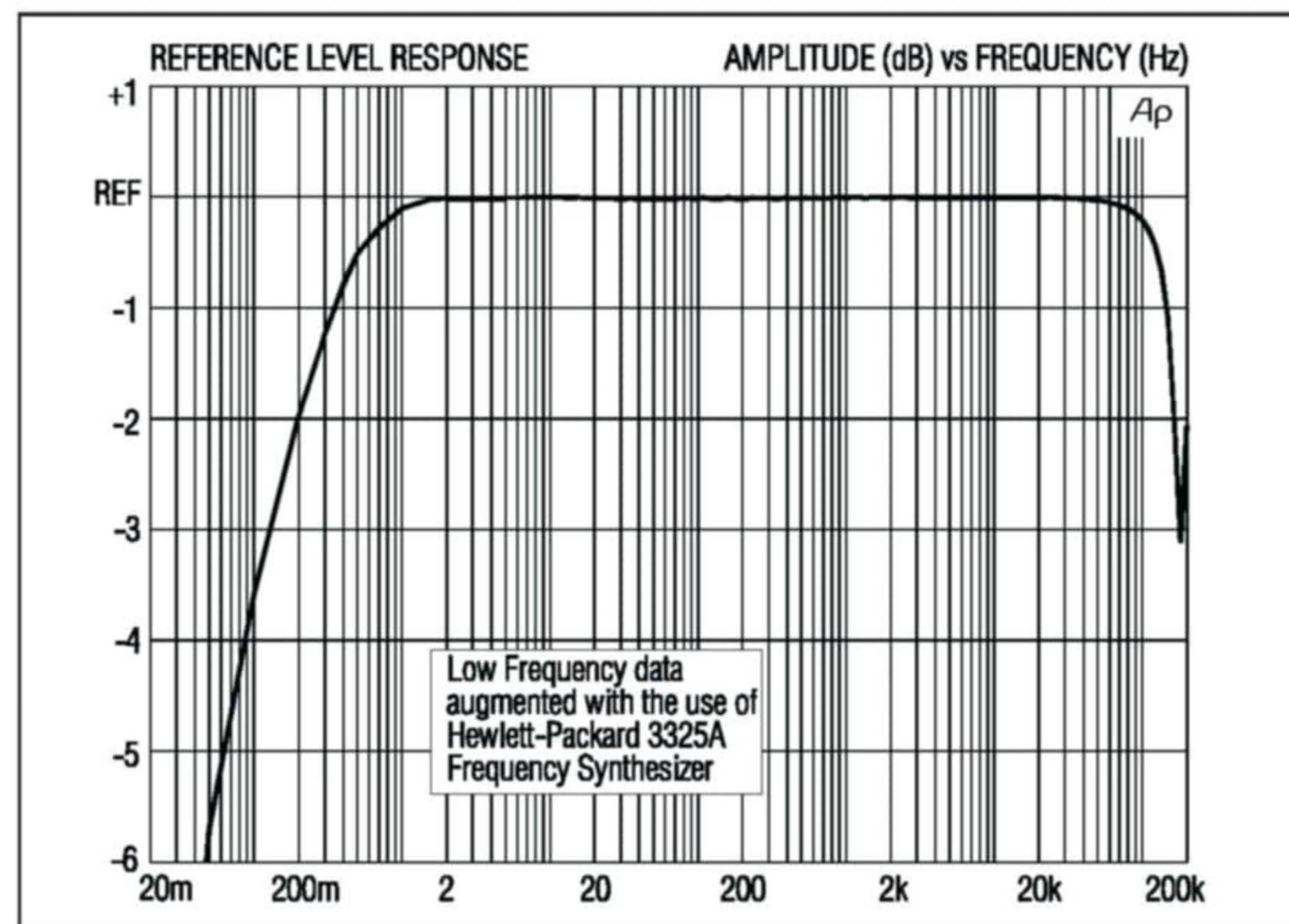


Figure 7: This is Jensen’s frequency-response graph for the JT-11-BMCF line-output transformer. Jensen specifies ultimate -3 dB points of 0.15 Hz and 15 MHz. (Image courtesy of Jensen Transformers, Inc.)

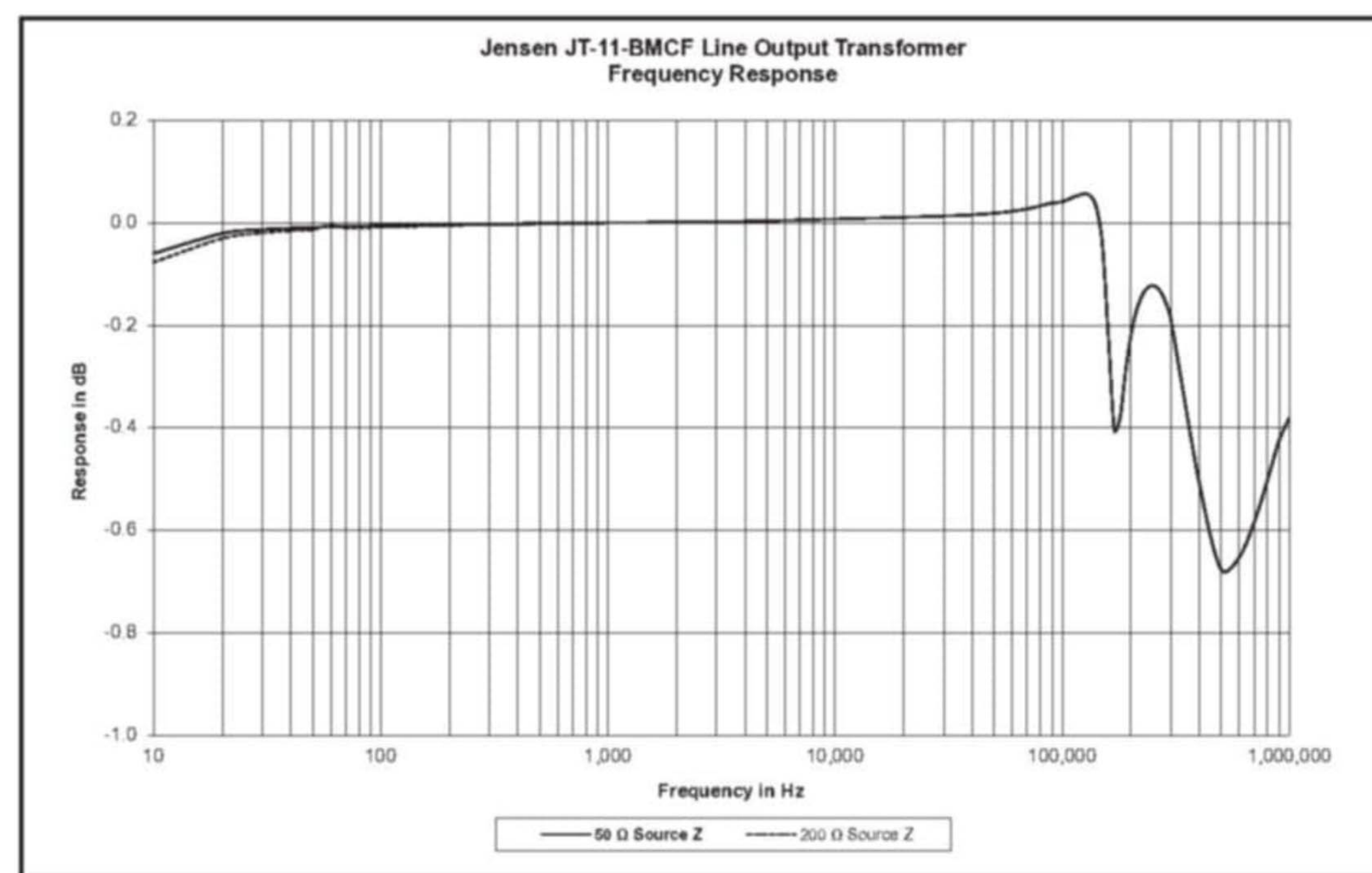


Figure 8: The solid line is the measured response of the JT-11-BMCF transformer with a source impedance of 50 Ω , plotted from 10 Hz to 1 MHz. From 20 Hz to 20 kHz the response is flat ± 0.016 dB. The dashed line shows the response below 1 kHz with a source impedance of 200 Ω . The additional roll-off at 20 Hz is only 0.07 dB.

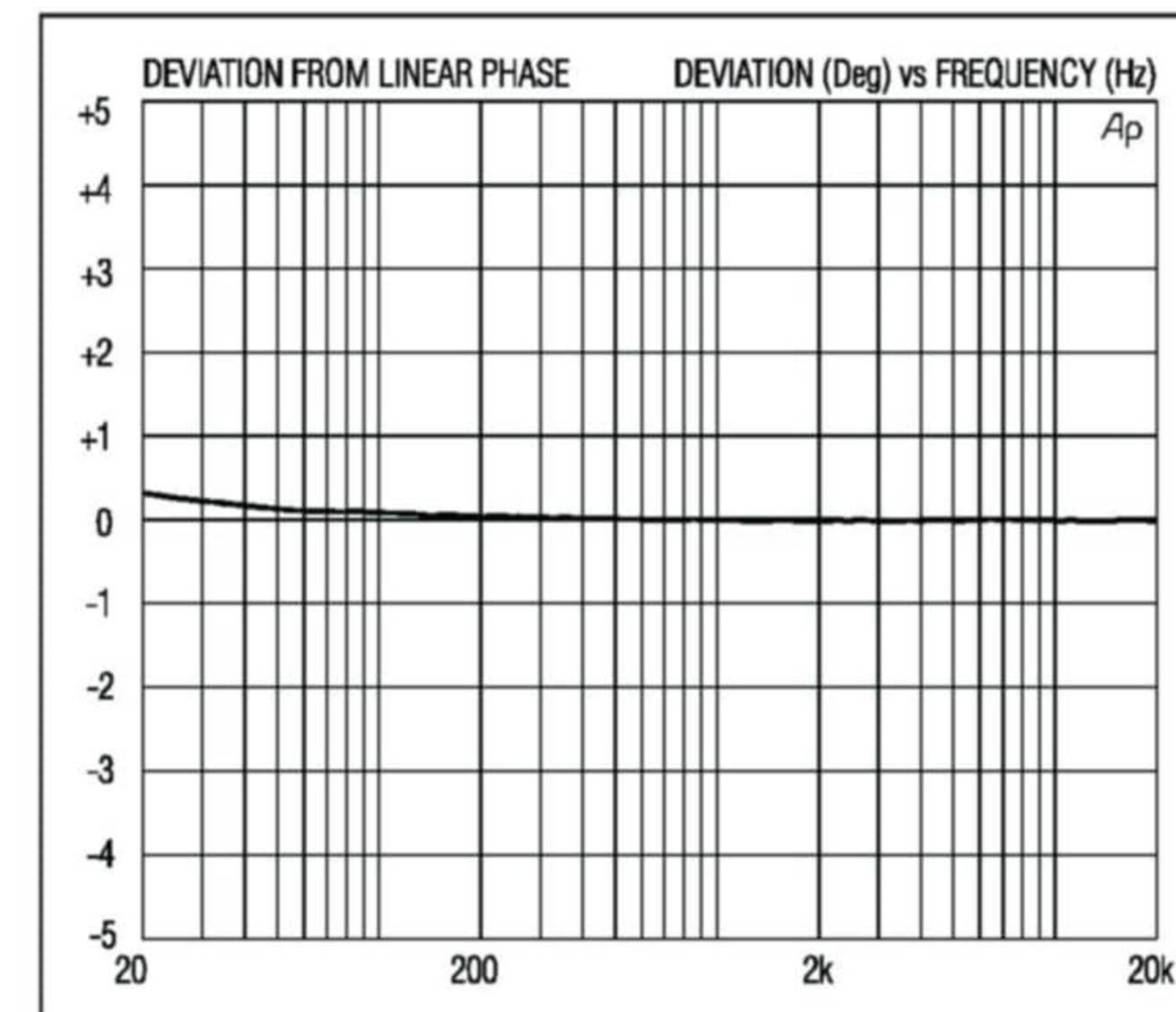


Figure 9: Extremely low deviation from linear phase is a hallmark of Jensen’s audio transformers. The JT-11 BMCF is virtually flawless to 20 kHz, and has only a slight deviation of around 0.3° at 20 Hz. (Image courtesy of Jensen Transformers, Inc.)

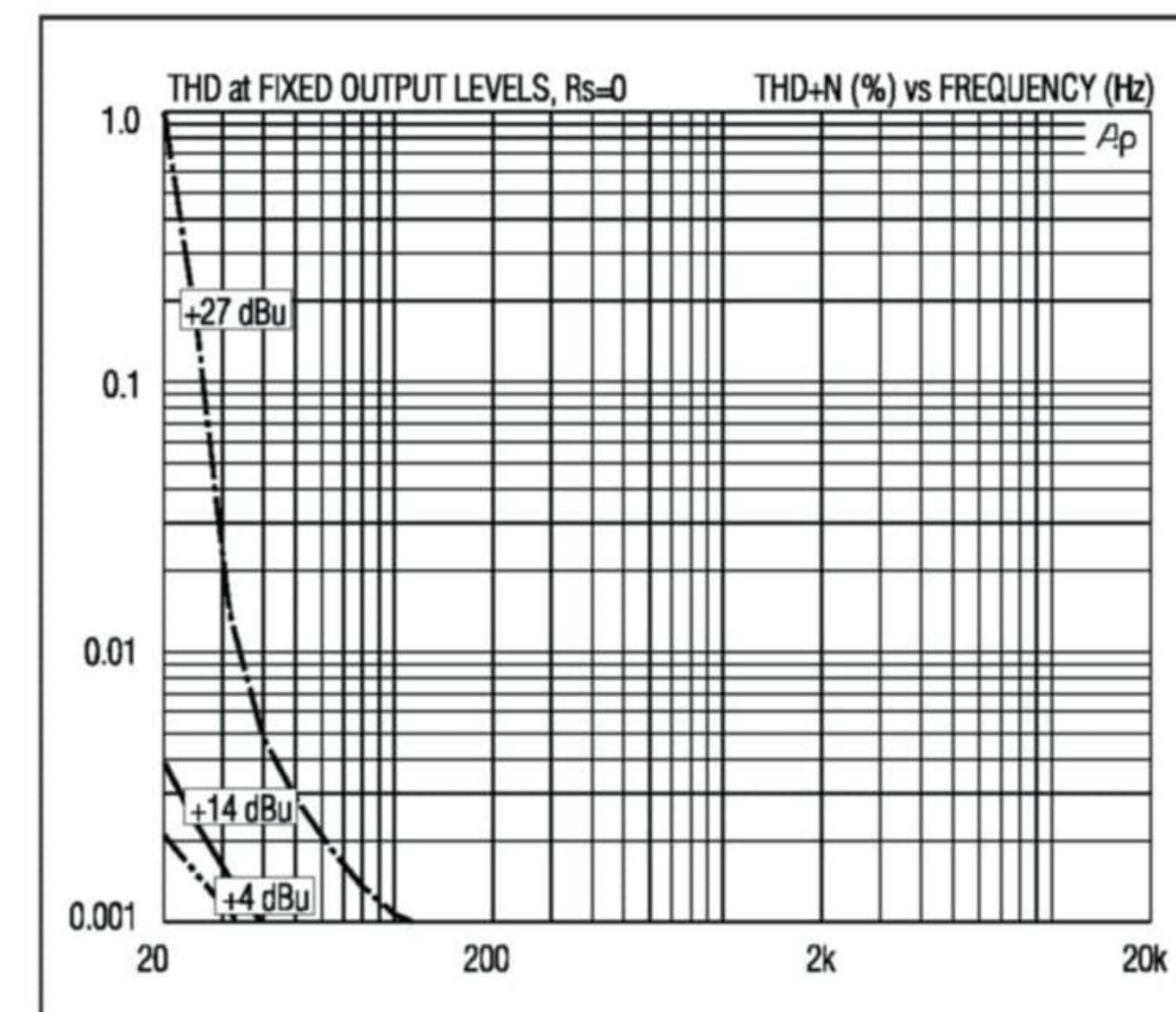


Figure 11: This graph shows total harmonic distortion (THD) of the Jensen transformer at levels of +4, +14, and +27 dBu. THD is below 0.001% across most of the audio band. (Image courtesy of Jensen Transformers, Inc.)

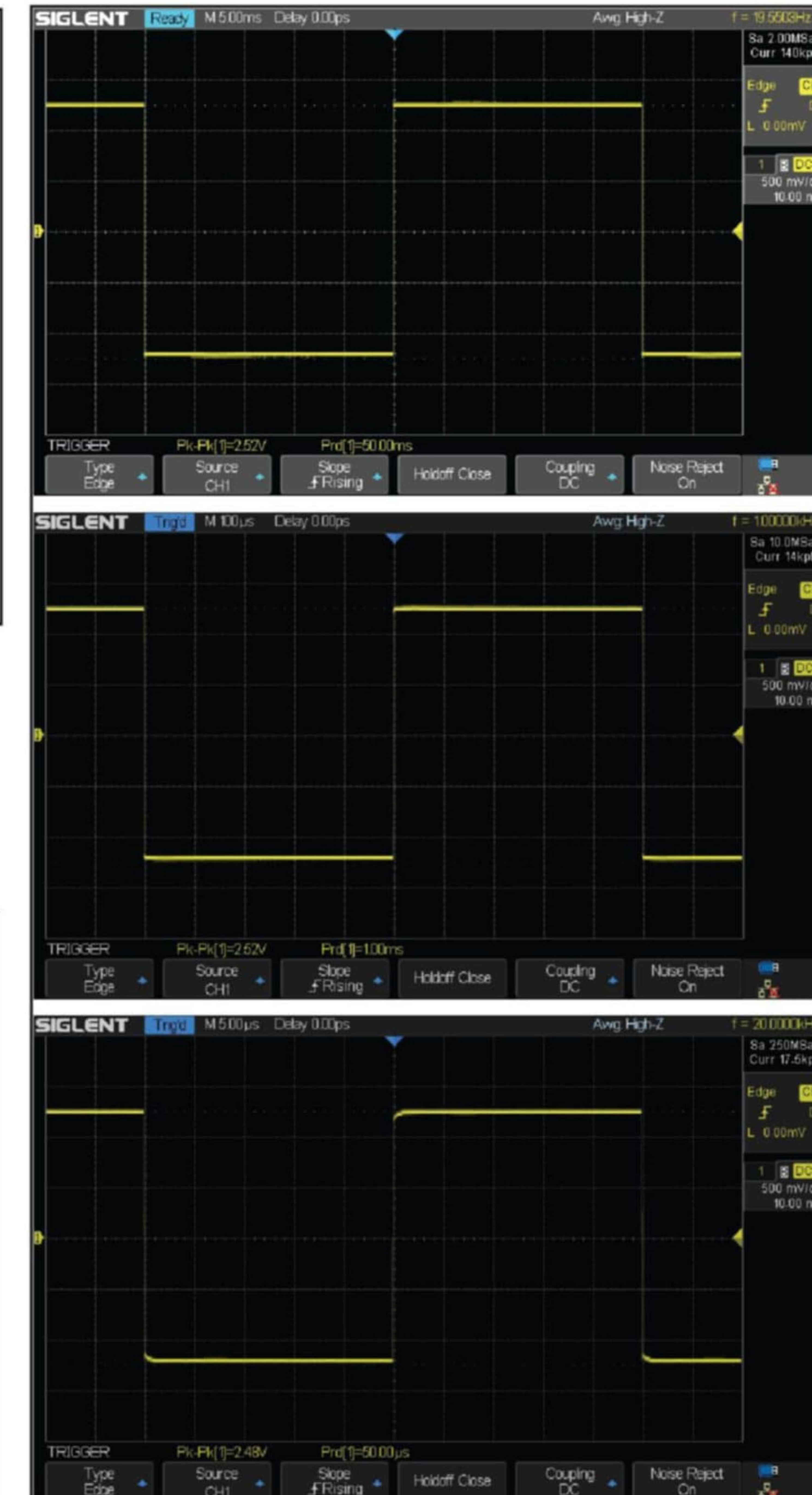


Figure 10: This shows the square wave response of the Jensen JT-11-BMCF transformer at 20 Hz (top), 1 kHz (middle), and 20 kHz (bottom), all with a 50 Ω source impedance and a 10 k Ω load, at a level of +4 dBu. The excellent performance reflects the transformer’s wide bandwidth and linear phase response.

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“System Applications Guide, 1993,” Analog Devices, www.analog.com/en/education/education-library/system-applications-guide.html.

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About the Author

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

20 kHz the response is flat ± 0.016 dB. A slight high-frequency rise peaks at +0.06 dB at 130 kHz before rolling off. The discrepancy between Jensen’s graph and mine above 100 kHz is due to the different source impedances used in the measurements. Jensen used the isolation network discussed earlier, which has a reactive component at very high frequencies. I used a generator with a 50 Ω resistor in series with the signal output.

Linear phase response has always been a hallmark of Jensen’s transformers. **Figure 9** shows the deviation from linear phase for the JT-11-BMCF transformer. The deviation is 0° over most of the audio spectrum, and has only a slight rise of +0.3° at 20 Hz. Square wave performance shows the benefits of linear phase response and wide bandwidth.

Figure 10 shows the square wave response I measured at 20 Hz (top), 1 kHz (center) and 20 kHz (bottom), with a 50 Ω source impedance and a 10 k Ω load, at a level of +4 dBu. This is virtually flawless performance. You’d be hard-pressed to do better with an active circuit. Finally, **Figure 11** is Jensen’s plot of total harmonic distortion (THD), at professional operating levels of +4, +14, and +27 dBu (0 dB = 0.775 VRMS). THD is well below 0.001% above 100 Hz, rising at the lowest frequencies. Operating levels with consumer audio equipment will typically be lower than +4 dBu, where the THD is 0.002% at 20 Hz.

How does it sound? Very little, in my view. The Jensen JT-11-BMCF transformers are extremely transparent. The sound is clean, quiet, and detailed, with excellent extension and definition on both ends of the audio spectrum. As I noted earlier, the only real disadvantage is cost. But, these transformers will sonically outperform most single-chip balanced line drivers, including the THAT 1600 series. I have not had a chance to directly compare them to the TI OPA1632 fully-differential op-amps, but will probably do so in the future.

Readers who own power amps with balanced inputs, but who’ve been unable to take advantage of them, will find balanced interfaces a very worthwhile sonic upgrade. After living with my own preamp driving my Benchmark AHB2 amps in balanced mode, I could never go back to an unbalanced interface. If you have a high-end system saddled with an unbalanced preamp-to-power amp connection, the Jensen transformers offer an excellent solution to the problem.

Author’s Note: I am grateful to Bill Whitlock and Dave Hill for the invaluable information they provided during the preparation of this article.

The “Must Have” reference for loudspeaker engineering professionals.

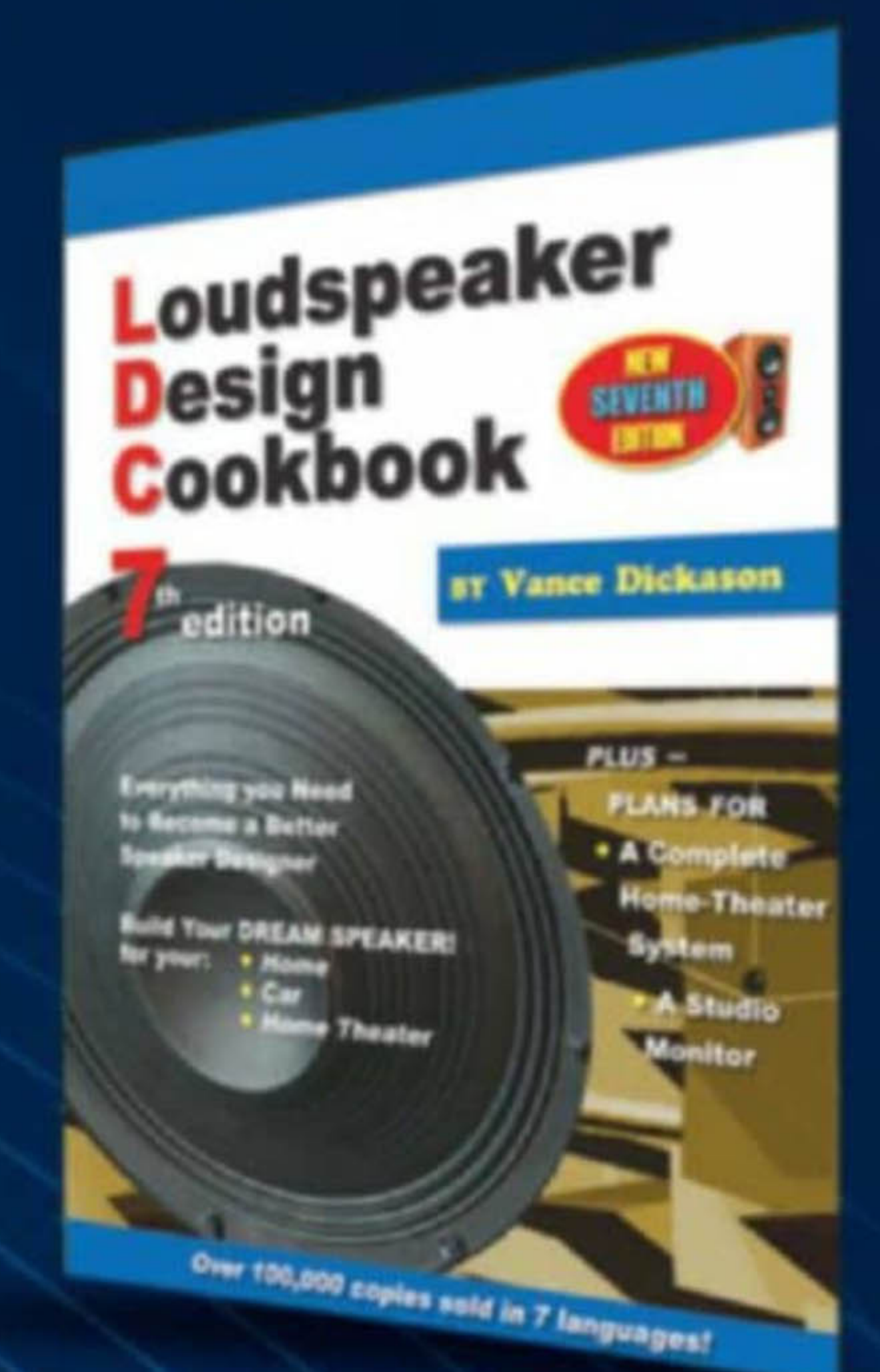
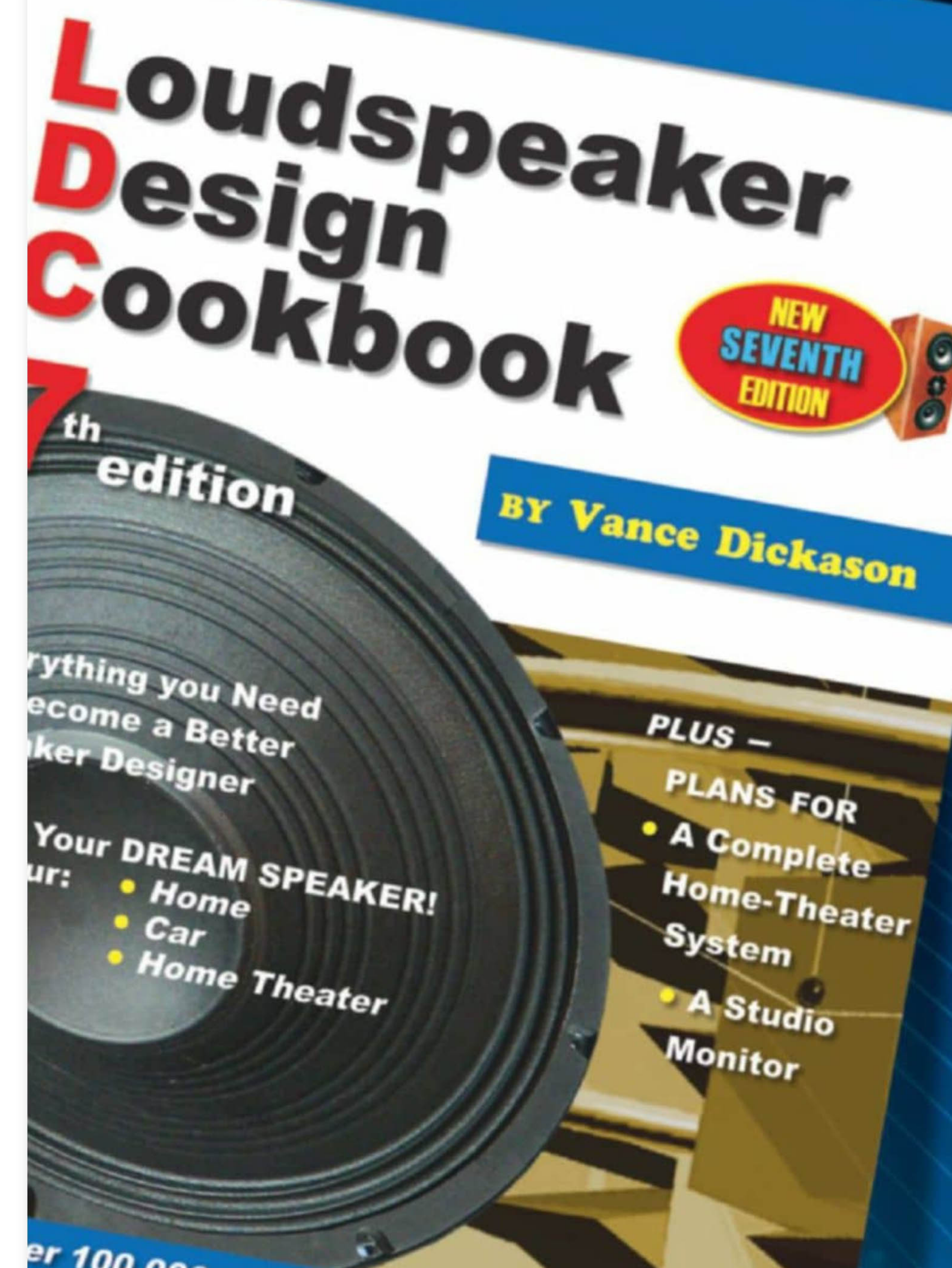
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An Experimental Cutterhead for Phonograph Disc Recording

By
Jim Wood
(United States)

There are a few of us among the readership of *audioXpress* who remember how it was in the days of “Hi-Fi” and even earlier. Nowadays, the public takes audio recording for granted... downloading music from the Web, ripping tunes to iPods, or capturing a live concert with a cell phone. Even a child can do it. Before there were MP3s, CDs, or even tape recorders, radio and 78 RPM records brought music into our homes, and unless you played a musical instrument, that was it. In this article, Jim shares details about the lost art of cutting phonograph records.

Home recording was virtually unheard of until the 1940s. My parents purchased a portable home recorder during that era to capture my first gurgles and, later, words. It was a combination radio, phonograph, and “record-maker,” much like the Meissner unit shown in **Photo 1**. This was not an inexpensive machine, selling for close to \$1,000 in today’s dollars. It cut a groove in a lacquer-coated cardboard or aluminum disc with a playing time of up to 4 minutes, and boasted audio response clear out to 6 kHz.

A decade later, consumer-grade tape recorders began to appear on the mass market. Tape machines offered a longer recording time, far better audio quality, and much easier operation. This essentially closed the chapter on home disc recording. The comparatively few home disc recorders in circulation went to thrift stores or to the local dump. By the 1970s, inexpensive and high-quality stereo tape recorders from overseas allowed audiophiles to commit the first playing of a new LP to tape, or to

copy friends’ albums “...for personal use at home,” of course.

Despite the arguable fact that today’s digital recording and delivery systems offer virtually flawless audio performance, there are many who insist that vinyl LP phonograph records sound better. Ongoing rumors of a “vinyl comeback” abound, whispered to the raised eyebrows of questioning audio purists. But whether this seeming step backward has any foundation in fact, or is simply nostalgia for how music used to sound to us, there is a growing cult following... not only of record collectors, but of those who want to make their own discs.

The Internet “Lathe Trolls” (www.lathetrolls.com) is home to a surprising number of tinkerers, experimenters, and even audio professionals, all sharing a fascination with disc recording. Members either DIY their recording equipment or grab whatever pops up from time to time on eBay or in junk shops. Available machines range in capability



Photo 1: Suitcase recorder

between the home “suitcase” recorder (see Photo 1) and vintage professional gear, once the mainstay of radio stations and recording studios. These old machines from the 1940s and 1950s are strictly monaural.

You can find a good many YouTube videos with demonstrations of homebrew disc recorders. Construction techniques vary from the use of scrap lumber and kitchen utensils to beautifully machined and 3D-printed parts. Results from these amateur efforts range from “Hey, I think I can almost recognize that tune!” to truly spectacular audio quality.

The Challenge

While many “trolls” remain content with cutting monaural records, the adventurous long for full-dimension stereo. And therein lies a problem. The electromechanical disc recording head or “cutterhead” remains the weakest link, and presents the ultimate challenge in making quality stereo phonograph records. Cutterheads are complex, precision electromechanical devices. Even an old mono head can fetch \$200 or more on eBay. A stereo head, if you could find one, would sell for thousands.

The project described here was undertaken with only a rudimentary understanding of mechano-electro-acoustical physics, and a senior tinkerer’s skill with home-garage tools—notably, a band saw, a drill press, and a belt sander, plus the usual assortment of files, screwdrivers, and a standby hammer.

The Design

To properly credit stereophonic recording from its inception would require a historical treatise harking back to non-electronic recording practices from the turn of the last century and would include myriad citations and credits. For the purposes of adhering to what became the world standard for stereo records, we’ll simply acknowledge the “45/45 system,” invented by England’s Alan Blumlein in 1931 and perfected by the US firm, Westrex (an offshoot of Bell Labs), in the late 1950s. Most readers will already understand how stereo records work, and the Internet provides a wealth of information on the evolution and mechanics of the “stereodisc” for those wanting a refresher course.

Figure 1 is a drawing taken from the Westrex patent that illustrates its fundamental cutterhead concept. The long-expired patent gives nearly enough detail for a clever and daring soul to copy the Westrex head. But with the limited skills and materials at hand, it seemed a better idea to use common tools and simple off-the-shelf component parts.

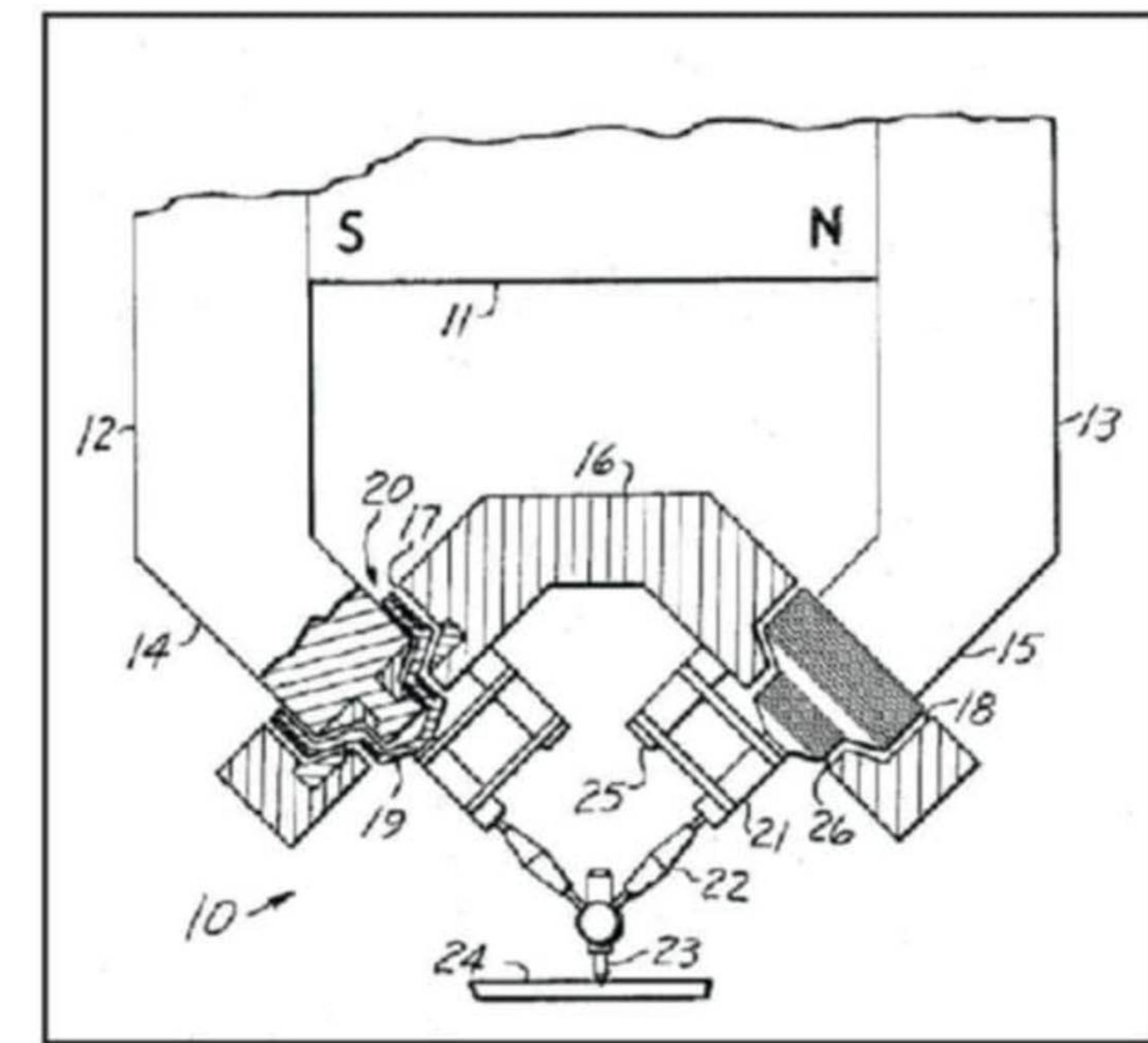


Figure 1: Westrex patent drawing

Photo 2 shows a device simply called an “audio transducer” or “exciter.” These are available from multiple sources on the Web and come in various shapes and sizes, all intended to “...turn your walls into a giant, wide-range loudspeaker!” Yeah, right. This particular unit measures about an inch in diameter and an inch tall, and can handle about 3 W of audio.

Transforming a pair of these transducers into a stereo cutterhead was pretty much a seat-of-the-pants effort, an exercise of fits and starts that lasted a couple of weeks and produced a trail of aluminum shavings from the garage to the house. **Photo 3**

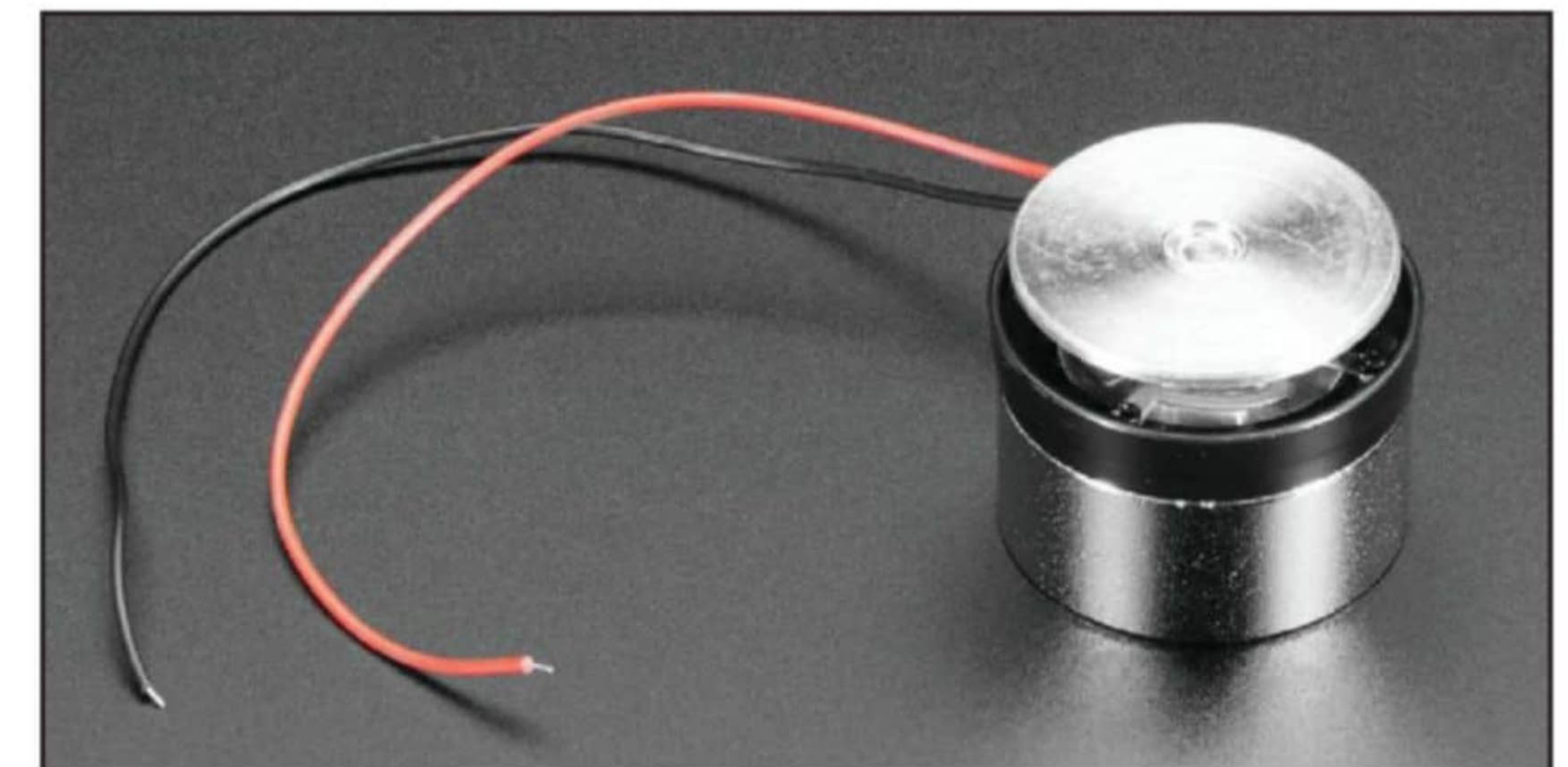


Photo 2: Audio transducer



Photo 3: The cutterhead

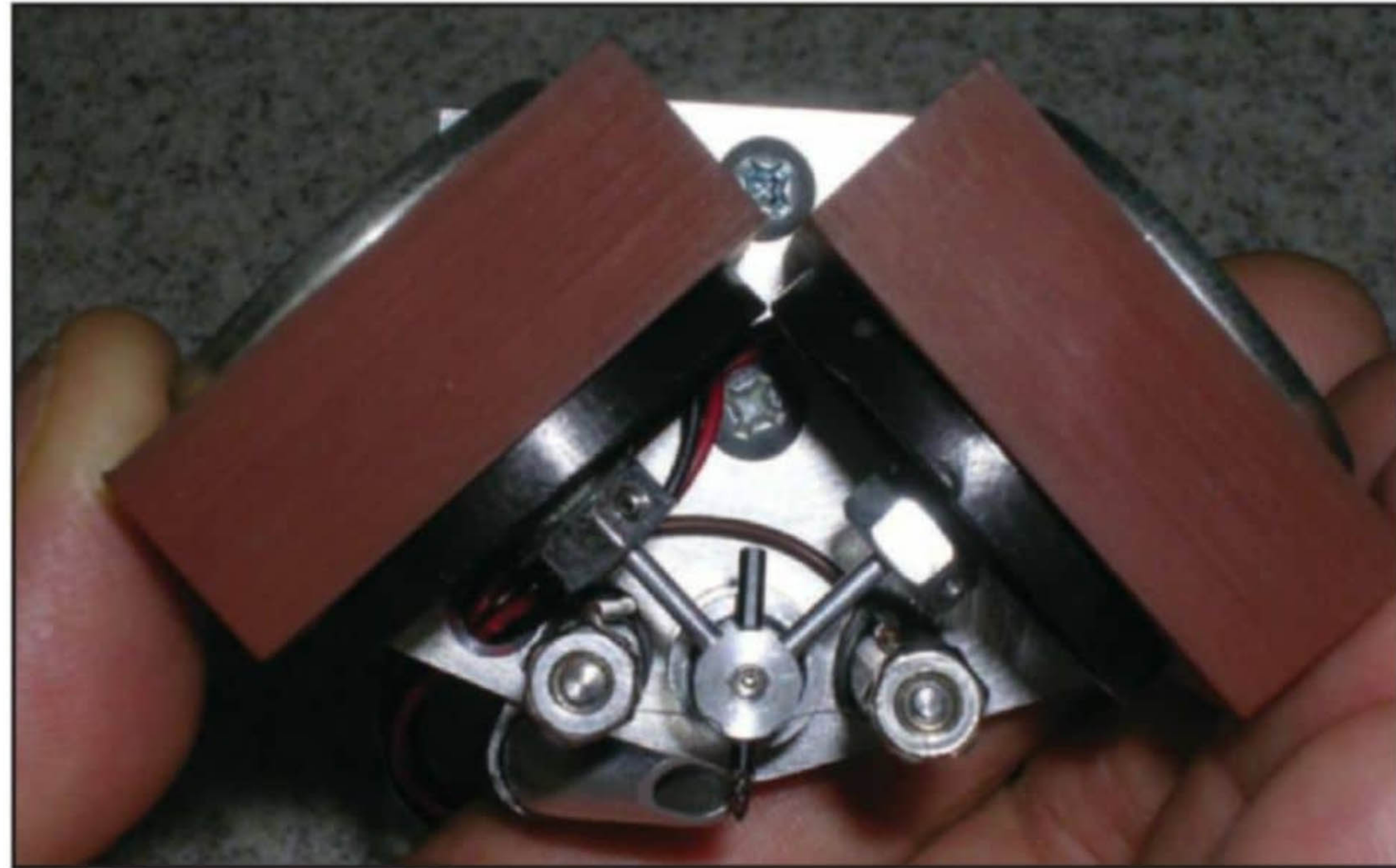


Photo 4: Top side of the cutterhead

and **Photo 4** show the finished head, construction details follow.

The transducers are secured in blocks of 1/2" phenolic material. Phenolic is a strong laminate of canvas and resin that can be worked almost like wood. The blocks were drilled with a 30 mm Forstner bit and are held to the aluminum mounting plate with screws through the plate into tapped holes in the phenolic.

The transducers have a lightweight aluminum voice coil terminating in a 5 mm threaded stud. Normally, this stud screws into the mounting flange shown atop the transducer as shown in Photo 2. For the purpose here, the flange was replaced with a 5 mm nut, drilled, and tapped for a setscrew to grip the pushrods that transmit vibration to the stylus holder. The pushrods are 1/16" hard aluminum alloy welding rod press-



Photo 5: Cutterhead on the lathe

fit into the stylus holder. They extend into holes drilled into the threaded stud and are clamped by the setscrews.

The stylus holder is a short piece of 1/4" aluminum rod. The hole for the cutting stylus was drilled 18° off vertical. The cutterhead is mounted with an 18° forward tilt, which falls in the range of the industry-designated "vertical tracking angle." These complementary angles assure that the stylus is perpendicular to the record surface. This is very much the same technique used with stereo phono cartridges for playback.

A 1/16" rod runs from the back of the stylus holder through the center of a rubber damper line encased in an aluminum tube. The damper is a length of Buna-N rubber O-ring material, and allows the stylus to move in the X/Y plane, but not the Z (fore and aft). The rubber is supposed to damp any mechanical resonance, but we'll see about that later. A couple of carbon fiber standoffs secure the stylus heater wire, and a vacuum nozzle sucks away the "chip" or "swarf" as the groove is cut.

The First Test Cut

Photo 5 shows the cutterhead mounted on a 1950s-vintage Rek-O-Kut lathe. This lathe is typical of the lower-end, workhorse professional units that would have been found in radio stations and recording studios before magnetic tape replaced discs. Because records could be handled quickly and cued easily, discs remained the preference for station promos and commercial spots until tape cartridge machines assumed this role in the 1960s.

The overhead mechanism of the Rek-O-Kut can be up-ended for easy access to the stylus. **Photo 6** displays the head in this position, exposing the vacuum line and wiring. **Photo 7** shows the head cutting a test track. **Photo 8** shows the disc during playback.

The first test recording was recognizable as music, but I've heard far more faithful sound reproduction coupling two tin cans with a string. With the optimistic attitude that anything can be fixed, the first step toward cutterhead redemption involved the recording of wideband pink noise, which was then replayed on a calibrated reproducer into a PC-based Real-Time Analyzer (RTA) to see just how bad the overall response was. **Figure 2** provides the terrible news: a 40 dB amplitude response spread that would ideally be ruler-flat!

BFEQ?

From the outset, nearly all cutterheads designed for high-quality disc mastering have utilized motional feedback to overcome mechanical resonances and

help secure a flat frequency response. But this cutterhead doesn't lend itself well to the additional coils and magnet assemblies that can be seen in the Westrex illustration. Anyway, motional feedback is complex and, if not done scientifically, can cause the system to break into nasty oscillation. Thus, BFEQ was about the only alternative. What does that stand for, you ask? Brute Force Equalization!

Figure 2 displays pronounced, very sharp resonance peaks at 800 Hz, 1.6 kHz, and 11.1 kHz. There's also a dip at 1.2 kHz, along with some more gradual rolloffs at both the bottom and the top. Trying to correct these deviations with simple analog equalizers could be a nightmare, but modern DSP techniques allow quick, precise, and repeatable corrections that can be easily fine-tuned in real time.

Analog Devices' "SigmaStudio"

Analog Devices Corp. offers a clever utility that runs on any PC. The company calls it SigmaStudio, I call it DSP for Dummies. I heartily recommend this program, which is useful for a vast number of audio signal processing tasks. You don't have to know how to write code, you simply pull ready-made audio building blocks from a menu tree and wire them together on your computer screen, just like an analog block diagram. Instead of having to clip out and solder in different component values, you have instant slider-adjustment control over all parameters that you can hear while experimenting. The menu offers equalizers, filters, crossovers, mixers, noise generators, dynamics processors, audio delays, and more. The list seems endless and is constantly growing. The well-organized and friendly users' group is an invaluable aid when getting started with SigmaStudio, and members gladly share their DSP creations and hacks. Once your schematic is complete, strike a key and the program compiles the code and blasts it into a DSP chip on the attached development board.

The DSP chips themselves are not all that expensive. The entry-level one used here, which is more than adequate for this project, costs about \$10. Other chips in the family that can handle more instructions for fancier tricks are about twice that price. The development platform circuit board runs about \$200, including the chip. It runs off a USB port and has 1/8" mini-TRS jacks for stereo audio in and out. But as useful and versatile as the Sigma development system is, the ultimate aim is to incorporate the DSP chip on your own board. For that you need some memory and a basic controller chip to load the DSP chip at boot-up. Find out more at www.analog.com. Drill down under Products until you see SigmaDSP Audio Processors.

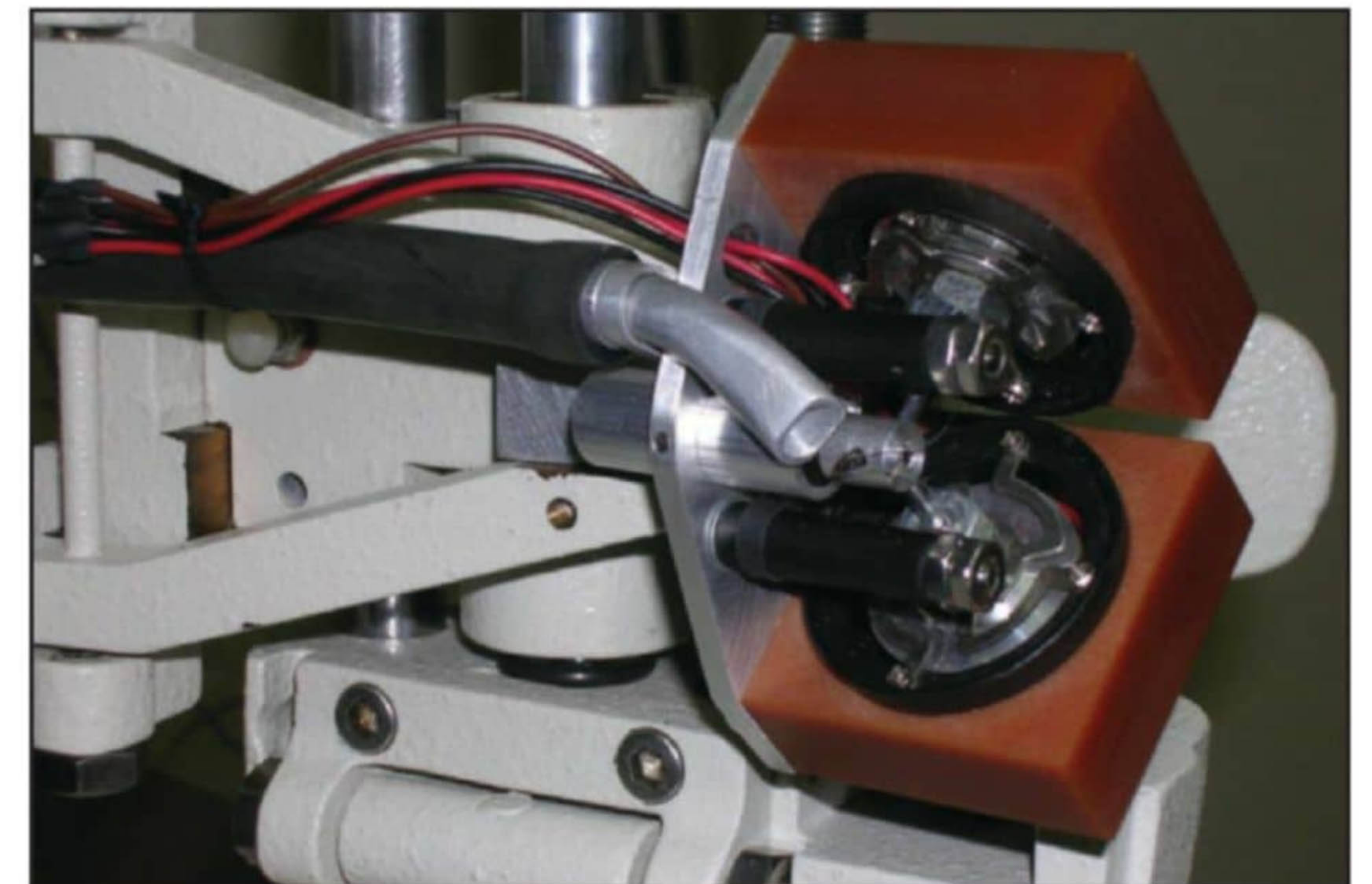


Photo 6: Lathe up-ended



Photo 7: Cutting a test track



Photo 8: The test playback

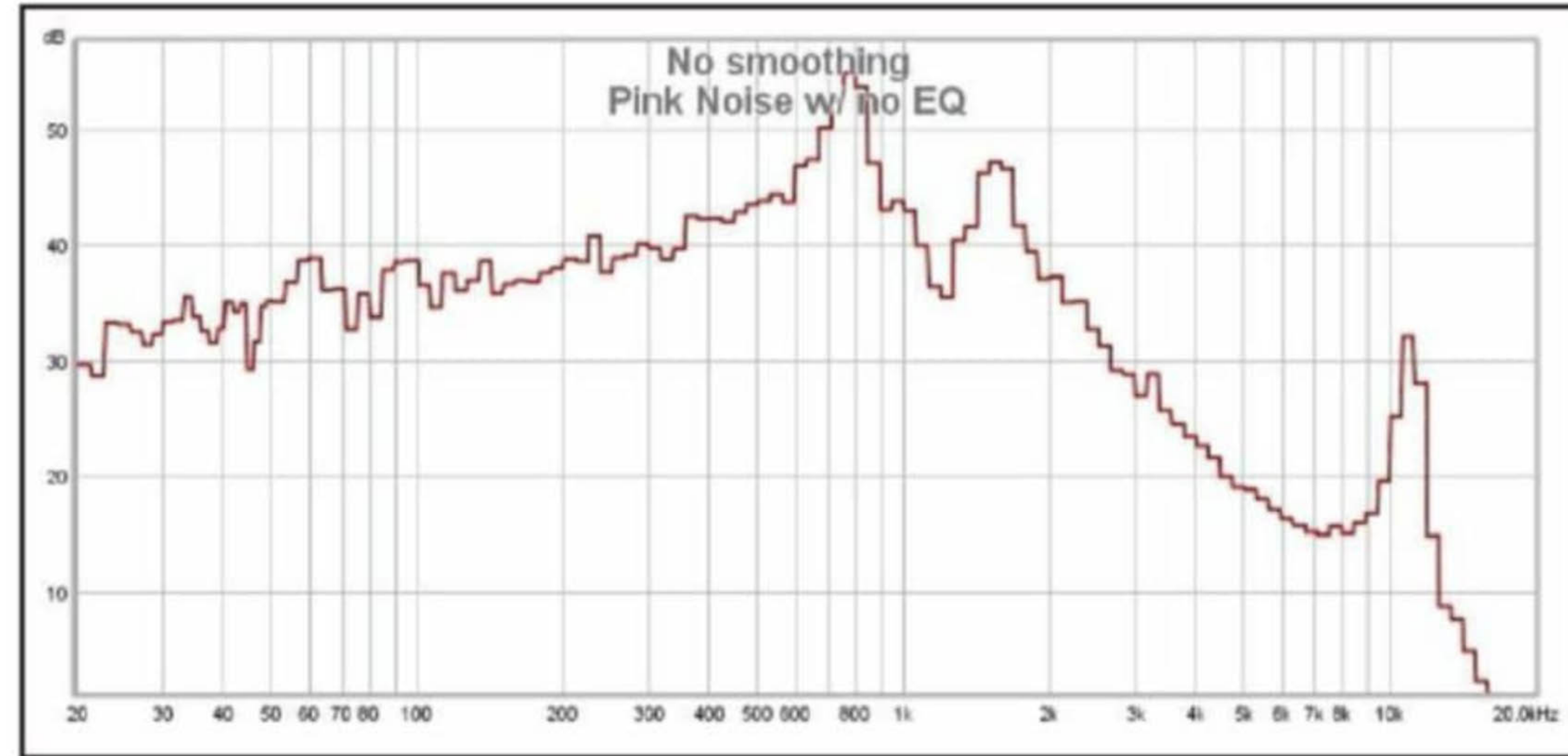


Figure 2: RTA response of raw pink noise

Photo 9 shows a screenshot of the SigmaStudio software as it was configured for this project. It uses seven cascaded parametric equalizers to correct cutterhead response irregularities. The center frequency, the “Q” (shape) and the gain (at the peak or dip) can be independently set for each section. These parameters were carefully tweaked to flatten the playback of the recorded pink noise while monitoring the RTA plot during playback in real time. Sigma’s Simulated Response graph pretty much mirrors the raw pink noise playback curve of Figure 2.

When this equalizer was relocated to the recording signal path, the cutterhead response irregularities were nicely compensated, just as I had hoped. Playback was within 2 dB from 30 Hz to 12 kHz. Music sounded much better, but there were still some issues.

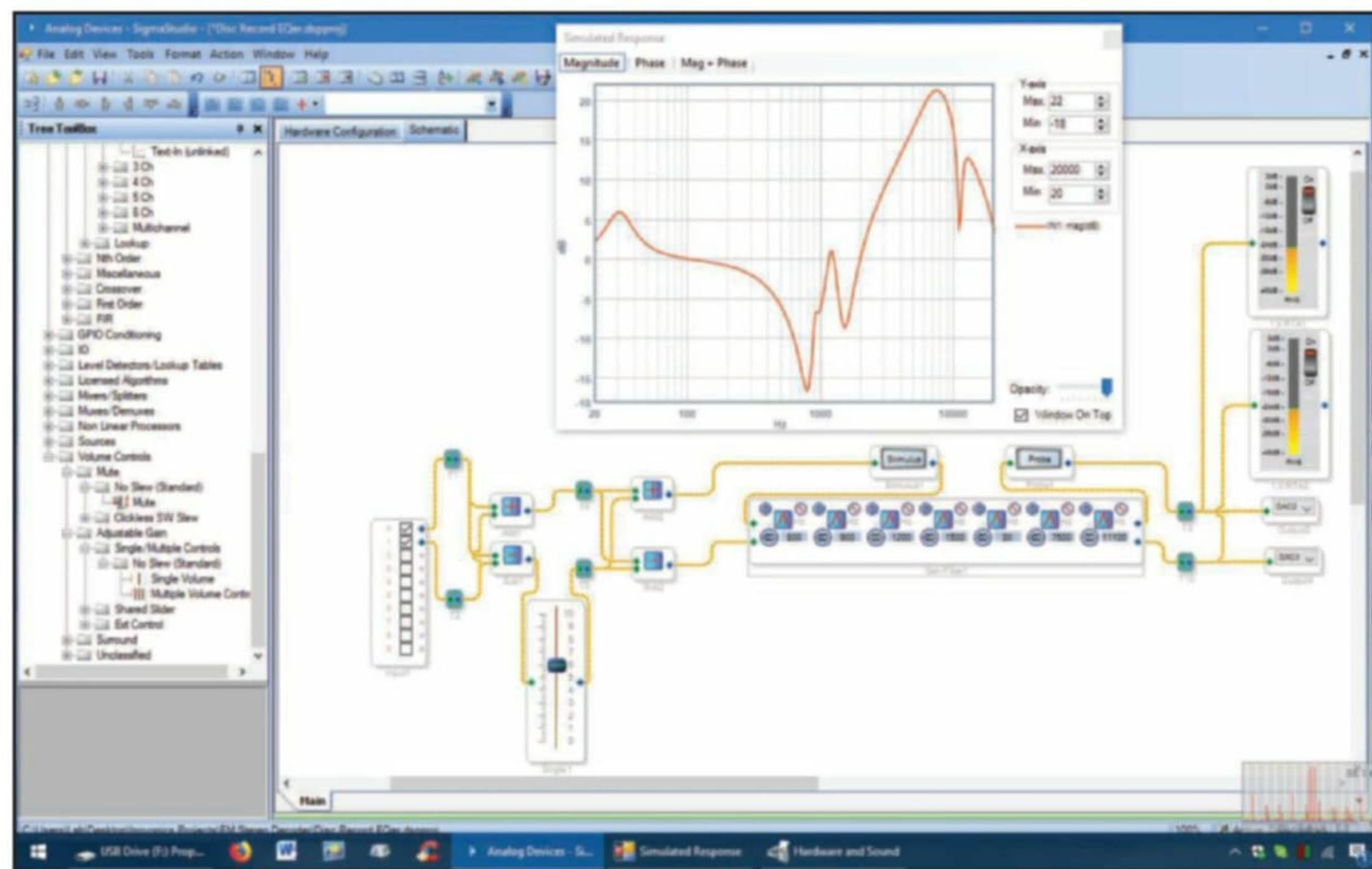


Photo 9: A screenshot of the SigmaStudio screen

Other Considerations

Using bodacious amounts of EQ to flatten audio response helps only to a point. The mechanical resonances are still there, and they cause a certain amount of ringing or “holdover” at the mechanical resonant points. This results in some smearing of music components at those frequencies, even though the measured response is not all that bad.

Another noted shortcoming was an apparent loss in stereo separation. The disc did not sound as wide as the source recording, and the stereo image was not nearly as well defined. This is probably due to the stylus holder and its pushrods being very rigid. Note that the Westrex drawing includes flexible coupling linkages, the No. 22 items detailed in Figure 1.

Sound sources centered in the stereo image are the monaural program components... pop music vocals, for instance. Center-stage sounds create only lateral groove modulation, like an old mono LP record. Unlike loudspeakers in a room, the transducers in the cutterhead are driven out of phase, so as one transducer is pushing the other is pulling. This causes the stylus to rotate (twist back and forth) to modulate the groove laterally.

Out-of-phase program components, the stereo, and ambience elements, cause the transducers to push and pull in unison. Again, this is the opposite of loudspeaker practice. Because the coupling mechanism is so rigid, the voice coils have to flex somewhat to move the stylus up and down for vertical groove modulation. Doubtless this accounts for a certain degree of harmonic distortion in the vertical

movement, which would manifest in left-only or right-only soundstage locations, and in room reverberation and other ambience effects. This is another reason that Westrex included those flexible linkages.

Referring to the Sigma schematic shown in Photo 9, a matrix/de-matrix circuit ahead of the equalizer translates the left and right channels into L+R “sum” and L-R “difference” signals, and then directly back again to left and right. Adding about 6 dB additional gain in the L-R difference domain dramatically improves the stereo image. Measured separation is still a pitiful 10 dB, certainly nothing to write home about, but with this difference-signal enhancement the music sounds more as it should.

So, How Does It Sound?

I’ve included a link to an MP3 file of a cutterhead test track. It’s a brief excerpt from a classical piece that was digitally recorded by a friend at a live orchestra rehearsal. It can be found in the Supplementary Material section of the *audioxpress* website (see Project Files).

A whine is audible in the background when the music isn’t playing. This is most likely caused by the unfiltered output of the SigmaStudio development

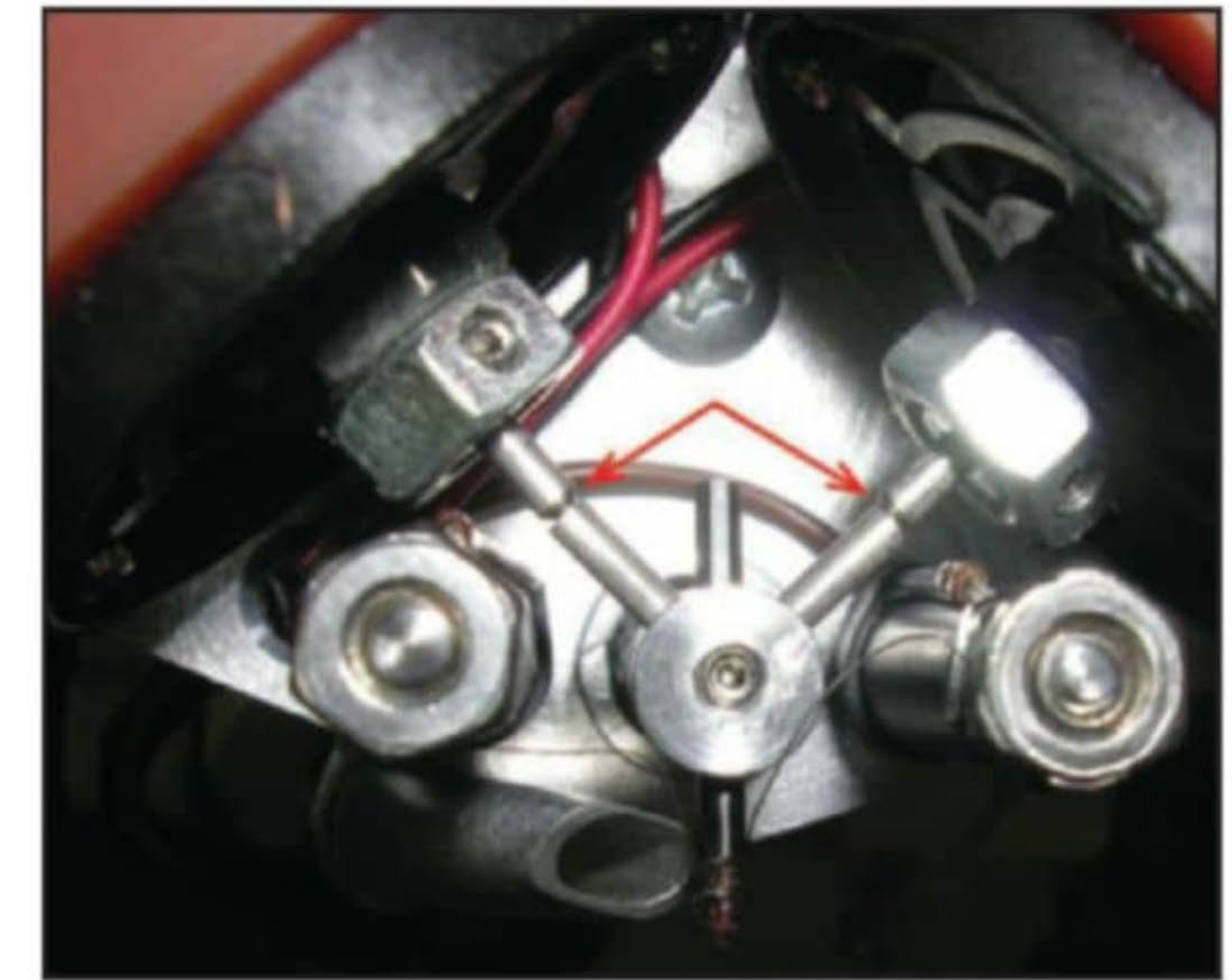


Photo 10: The ball and socket modification

board interacting with the Dayton Audio Class-D switching amplifier used to drive the cutterhead for these tests. This could easily be put right. Turntable rumble is present as well—the old broadcast turntables had an abundance of horsepower, which translates into lots of torque and lots of noise.

Afterthoughts and Modifications

A few weeks following initial tests and findings, the cutterhead received a few modifications:

- Tiny ball-and-socket-like joints were added to the pushrods coupling the drivers to the stylus holder. These give each driver freedom to move along its axis without the other driver





About the Author

Jim Wood is founder and board chairman of Inovonics, Inc., a California manufacturer of audio and RF products for radio broadcast. Jim has a longtime interest in audio and broadcasting and holds several related patents. He is a Life Member of both The Institute of Electrical and Electronics Engineers (IEEE) and the Audio Engineering Society (AES) and remains peripherally active in his company's product development, but certainly not to the exclusion of "fun" projects like this one.

having to twist from side to side. **Photo 10** shows these "articulated" pushrods.

- The rubber-damped rod securing the stylus holder to the backplate was modified to reduce its stiffness and allow it to move more freely. These mechanical changes greatly improved stereo separation and actually helped flatten the uncorrected frequency response. Only five equalizer sections were then required to make a flat recording.
- The drive electronics were revised as well, with the addition of a fast, unobtrusive wideband peak limiter, and an independent high frequency "acceleration" limiter. The first protects against overcutting into adjacent grooves at lower frequencies, and the second controls top-end energy.

Acceleration Limiting

Analog audio delivery systems frequently utilize complementary pre-emphasis and de-emphasis to mitigate noise in the transmission or recording path. Among the well-known and standardized examples of this are FM broadcasting, analog magnetic tape recording, optical film tracks, and phonograph records.

A drawback to the use of pre- and de-emphasis, however, is that it necessarily imposes a shortcoming in the form of signal headroom that is frequency-dependent. In other words, if you deviate your FM carrier to its limits, bring magnetic oxide to a still-reasonable level of compression distortion, or modulate a groove so that it almost "kisses" its neighbor at low frequencies, then you would horribly overmodulate, saturate the tape or over-cut at the pre-emphasized high frequencies.

Fortunately, a statistical analysis of music and speech shows that naturally occurring high-frequency components are significantly lower in level than, say, voice frequencies. The silky highs we hear from our tweeters, even the "rosin on the bow," tell us that our

system is reproducing the treble range, but those overtones are much lower in absolute level than the fundamental frequencies they help to define... until relatively recently, that is.

Beginning in the late 1950s, and increasing as the years have passed, there is an industry-wide trend to distort the audio spectrum in favor of higher and higher energies at high frequencies. Pop music vocals demonstrate this. A female singer working close to the mic, plus the actions of a producer and engineer (both of whom may well be going deaf!), can create a track that nearly makes your ears bleed when the lyrics go, "Honey, you're sssso, sssso ssssweet."

Chromium-dioxide-based cassette tape, followed in short order by CDs and other digital delivery systems, allowed these hyped-up highs to reach our homes. Previously, phonograph records and FM radio automatically limited high energy at these frequencies due to the frequency-selective nature of their signal headroom. The 12" "disco" 45 RPM releases of popular hits were an attempt to expand top-end headroom, which improves as the groove velocity increases.

In FM broadcasting, an entire industry grew from a need, both to secure the loudest sound on the dial, and to limit high-frequency energy without an apparent loss of top-end. The CBS Labs "Volumax" began an FM audio processing revolution, enabling the "loudness wars" that continue to this day. The Fairchild "Conax" (Constant Acceleration) limiter and similar systems built into Neumann and Ortofon disc mastering systems served a similar purpose for phonograph records.

A frequency-selective limiter quickly reduces only the treble range during high-frequency peaks (vocal sibilants and snare drums, for example), and then just as quickly restores full response. With the right processing algorithms, this action quite effectively deals with the headroom shortcoming without appreciably dulling the sound. In the project presented here, such a limiter made a significant contribution by purposely limiting top-end energy that could not have been engraved in the groove, or tracked in playback for that matter.

Conclusion

Given the simplicity of this cutterhead design and the dirt-cheap nature of the materials involved in its creation, I consider this project a "qualified" success. A follow-up version would certainly incorporate further improvements, and might even strike off in a different direction in the fundamental approach. Web and patent searches suggest several different means of modulating a stereo record groove; this project has been a very humble beginning. 



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

February 2019

February 5-8

ISE 2019

Amsterdam RAI, The Netherlands

<https://www.iseurope.org>

www.avixa.org | www.cedia.net

www.iseurope.org

February 16-17

CanJam NYC

New York Marriott Marquis, New York, NY

<https://canjamglobal.com/shows/nyc2019>

February 25-28

GSMA Mobile World Congress

Fira Gran Via, Barcelona, Spain

www.mobileworldcongress.com

March 2019

March 20-22

electronica China

18th International Trade Fair for

Electronic Components, Systems, and Applications

China Import & Export Fair Complex,

Guangzhou, China

Shanghai New International Expo Centre, China

www.electronica-china.com

March 20-23

146th AES Convention

The Convention Center, Dublin, Ireland

Conference: March 20-23, 2019

Exhibits: March 21-23, 2019

www.aes.org/events/146

March 22-24

Salon Audio Montréal/Montréal Audio Fest

Hotel Bonaventure, Montréal, Canada

www.montrealaudiofest.org

March 27-29

2019 AES International Conference on Immersive and Interactive Audio

York, UK

www.aes.org/conferences/2019/immersive

March 30-31

CANJAM Singapore

Pan Pacific Singapore, Singapore

<https://canjamglobal.com/shows/singapore2019>

April 2019

April 2-5

Prolight+Sound Frankfurt Musikmesse

Frankfurt Messe Exhibition Center,

Frankfurt Germany

www.musikmesse.com

www.prolight-sound.com

www.messefrankfurt.com

April 8-11

NAB Show 2019

Convention: April 6-11

Las Vegas Convention Center (LVCC),

Las Vegas, NV

www.nabshow.com

April 12-14

Audio Expo North America (AXPONA)

Renaissance Schaumburg Hotel &

Convention Center, Schaumburg, IL

www.axpona.com

May 2019

May 3-5

Lone Star Audio Fest

The Embassy Suites Dallas Park Central,

Dallas, TX

www.lonestaraudiofest.com

May 9-12

HIGH END Show

MOC Munich-Germany

www.highendsociety.de

May 13-17

177th Meeting of the Acoustical Society of America

Louisville, KY

acousticalsociety.org

June 2019

June 8-14

InfoComm 2019

Conference: June 8-14

Exhibits: June 12-14

Orange County Convention Center

Orlando, FL

www.infocommshow.org

June 9-10

ALMA International Symposium and Expo (AISE) 2019

Orange County Convention Center

Orlando, FL

<https://almaint.org>

June 11-13

CES Asia

Shanghai, China

www.cesasia.cn

June 18-20

2019 AES International Conference on Audio Forensics

Holiday Inn Porto-Gaia, Porto, Portugal

www.aes.org/conferences/2019/forensics

June 22-23

CanJam SoCal

Irvine Marriott, Orange County, CA

<https://canjamglobal.com/shows/socal2019>

July 2019

July 18-20

Summer NAMM

Music City Center, Nashville, TN

www.namm.org/summer/2019

July 26-28

California Audio Show

Venue TBC

www.caaudioshow.com

July 27-28

CANJAM London

Park Plaza Westminster Bridge, London, UK

www.canjamglobal.com/shows/london2019

August 2019

August 9-11

Hong Kong High End Audio Visual Show

Hong Kong Convention & Exhibition Centre

(HKCEC)

www.audiotechnique.com

www.audiotechnique.com/av-show

August 26-29

SET EXPO

International Technology Fair on

Equipment and Services for Television

Engineering, Broadcasting, and

Telecommunications

Expo Norte, São Paulo, Brazil

www.setexpo.com.br

August 27-29

2019 AES Conference on Headphone Technology

The Golden Gate Club, San Francisco, CA

www.aes.org/conferences/2019/headphones

September 2019

September 6-11

IFA Berlin (The International Funkausstellung)

www.ifa-berlin.com

September 10-14

CEDIA

Colorado Convention Center, Denver, CO

www.cediaexpo.com

September 12-15

Prolight+Sound NAMM Russia

Sokolniki Exhibition and Convention

Centre, Moscow, Russia

www.namm-musikmesse.ru

www.prolight-namm.ru

September 13-17

IBC Show

RAI Amsterdam, The Netherlands

Exhibition: September 14-18

www.ibc.org

Project Files

To download the test MP3 file, visit

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





Industry Calendar

September 15-17

PLASA Show London

London, Olympia, UK
www.plasashow.com

September 22-24

GSMA Mobile World Congress Americas

Los Angeles Convention Center (LACC),
Los Angeles, CA
www.mwcamericas.com
www.gsma.com | www.ctia.org

September 23-25

Prolight+Sound Middle East

Dubai International Convention and Exhibition
Centre, Dubai, United Arab Emirates
www.messefrankfurt.com
www.prolightsoundme.com

Dates TBC

Consumer Electronics China (CE China)

Guangdong, China, China
www.ifa-berlin.com
http://www.cechina-ifa.com

October 2019

October 5-8

Rocky Mountain Audio Fest (RMAF)

Marriott Denver Tech Center, Denver, CO
www.audiofest.net
CANJAM RMAF
www.canjamglobal.com/rmaf2019

October 13-16

HKTDC Hong Kong Electronics Fair

(Autumn Edition)

Global Sources Electronics Show | electronicAsia

www.hktcdc.com/hkelectronicfairae

Dates TBC

The Loudspeaker Sourcing Show 2019

Nansha Grand Hotel in Guangzhou, China
www.loudspeakersourcingshow.com

October 16-19

AES New York

147th International Convention

Jacob K. Javits Convention Center,
655 W 34th St., New York, NY
Program: October 16-19
Exhibition: October 16-18
www.aes.org/events/147

October 16-17

NAB Show New York

Jacob K. Javits Convention Center,
655 W 34th St., New York, NY
www.nabshowny.com

November 2019

Dates TBC

Capital Audiofest

Hilton Hotel at Twinbrook Metro,
1750 Rockville Pike, Rockville, MD
www.capitalaudiofest.com

November 9-10

Shanghai Headphone Festival

Shanghai Marriott Hotel City Centre,
Shanghai, China
www.canjamglobal.com/shanghai2019

Dates TBC

New York Audio Show

Park Lane Hotel, Central Park South, New York, NY
www.chestergroup.org/newyorkaudioshow

November 18-24

Live Design International (LDI) Show

Las Vegas Convention Center, Las Vegas, NV
www.ldishow.com

November 13-15

Inter BEE

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

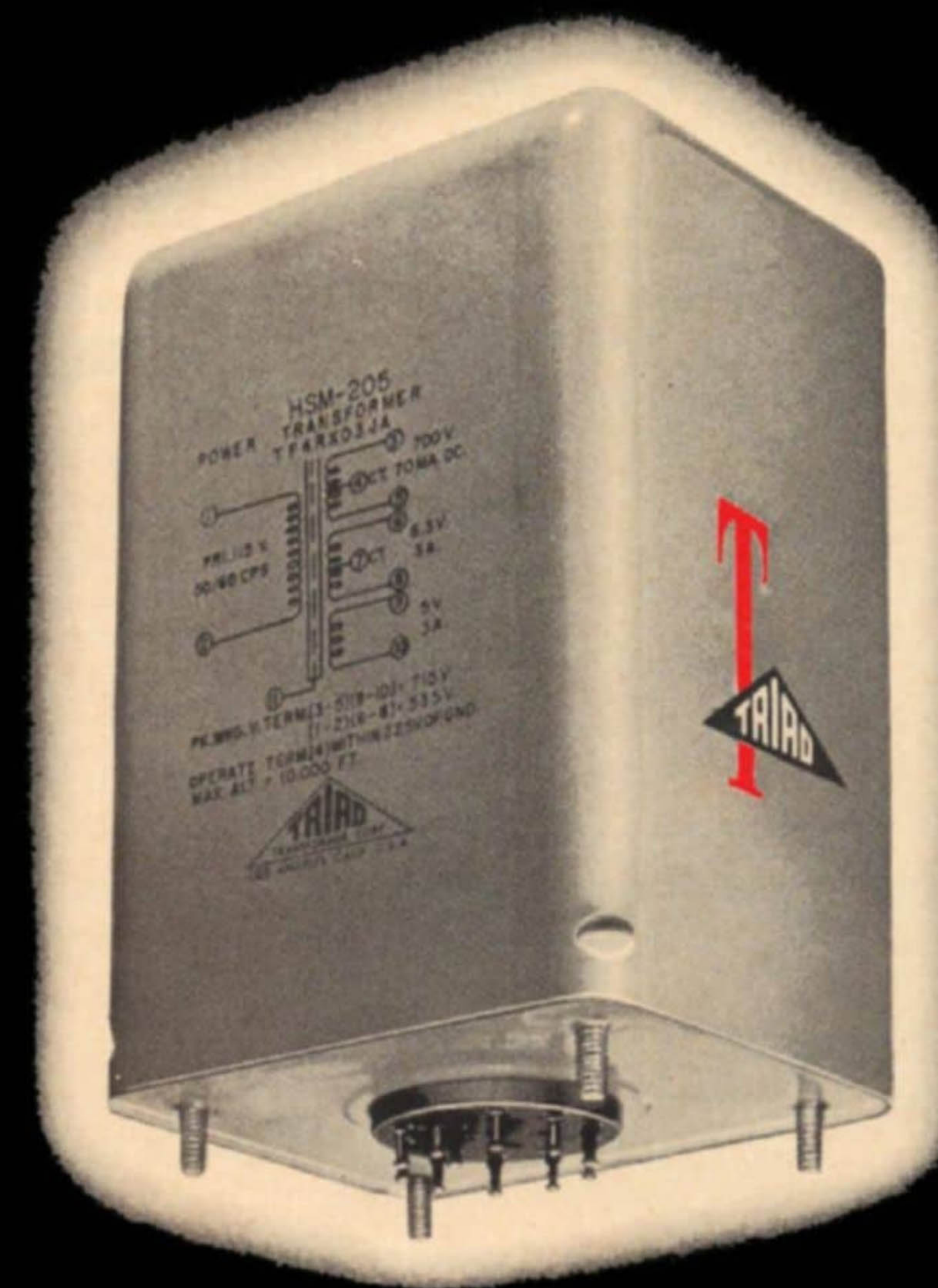
December 2019

December 2-6

178th Meeting of the Acoustical Society of America

San Diego, CA
acousticalsociety.org

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The NAMM Show
january 24/27 2019 ^{california}

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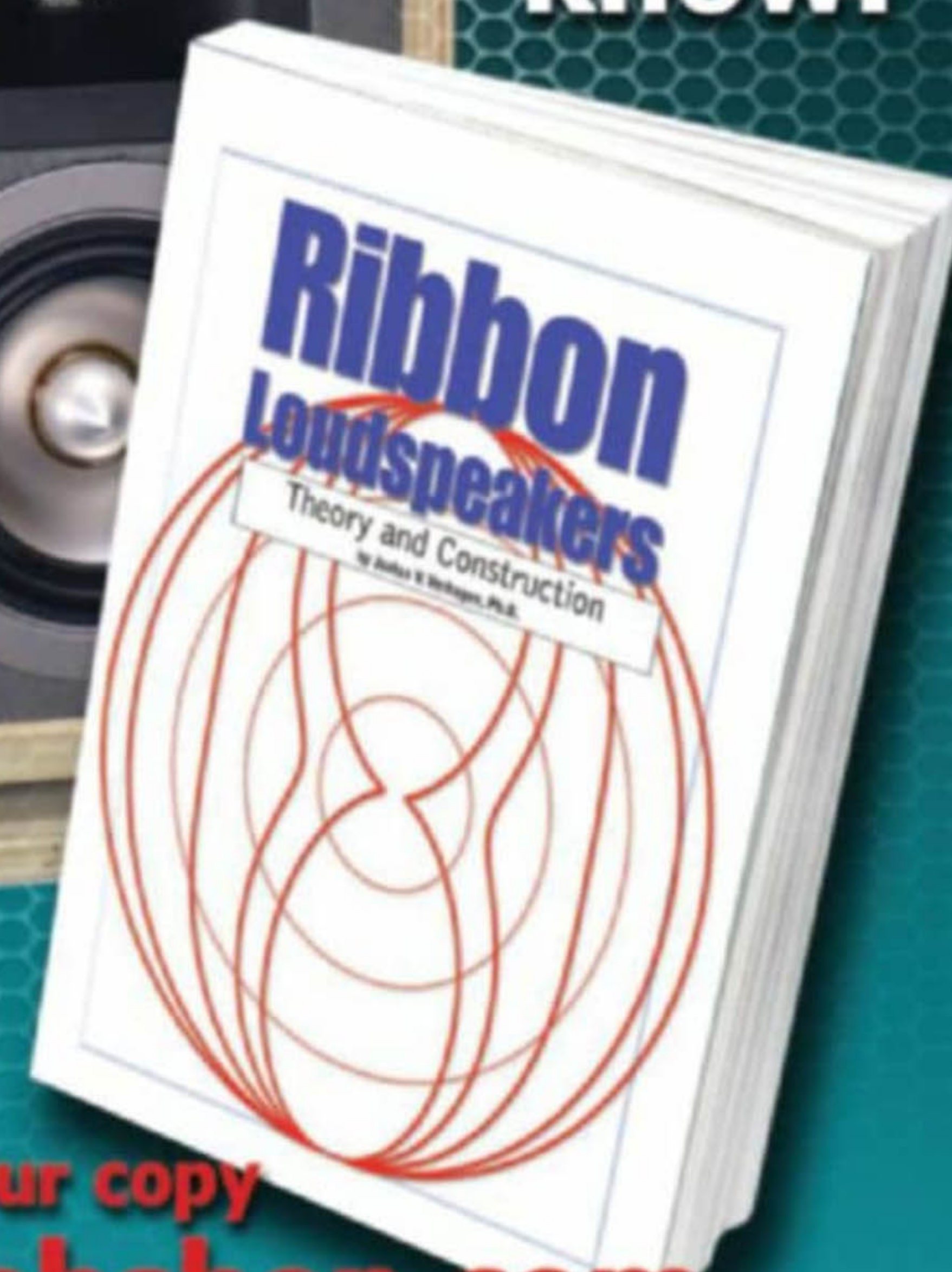


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