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ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

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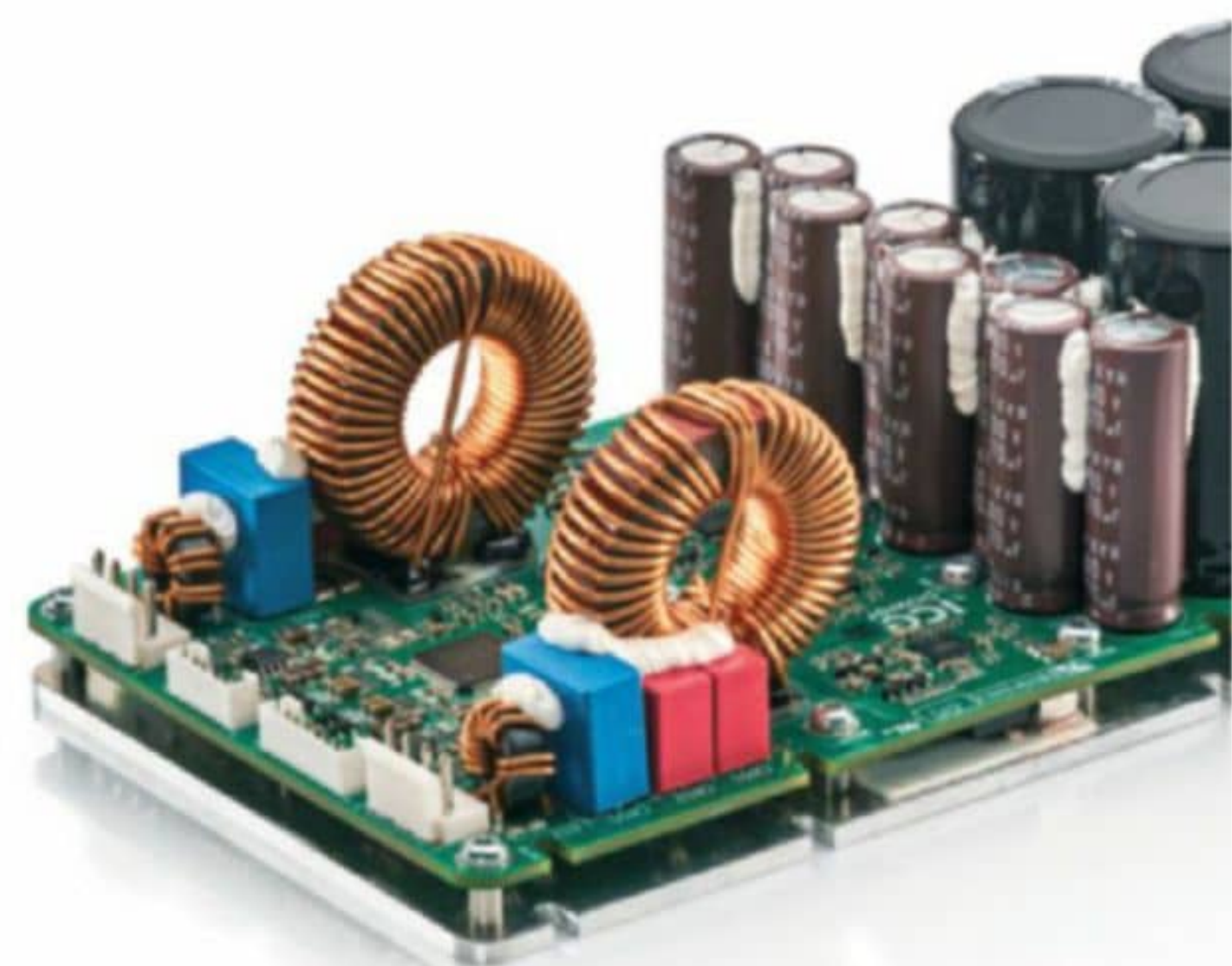


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Audio, Power, and Efficiency

The first half of 2023 allowed us to get back on the road and attend some of the busiest trade shows ever for the audio industry. From CES and a deep immersion into cutting-edge innovation in the consumer space to professional audio shows at ISE, the NAMM Show, and NAB. We also attended AXPONA and more recently, an extremely successful High End Munich show in Munich, Germany. With a few other events in the mix, this provided us a very good overview of the state of the audio industry. From the most affordable (and highly competitive) segments in consumer electronics, to the pro audio segments of live performance and touring, content production, media, and broadcast.

We do it to better understand the audio industry and gain a broad outlook of the trends and the opportunities. More importantly, we can follow the companies that are active in each space, and understand how relative things can be.

How a team of engineers invests years perfecting a high-end audio design aiming for unprecedented sonic performance—expense is no object—and another team invests months designing a new system used in live performances in front of thousands of people in a stadium, or a live broadcast for millions—where failure is not an option.

How a relatively successful small company in the studio monitor space, with a few employees can be successful across many of these market and application segments, while a much larger manufacturer completely focused on the hifi/high-end audio space, with 10 times the number of products in its catalog, can be successful and admired in its native market, and obscure in all other markets.

Each other's products can be admired and respected by their respective user community and there's no point on comparing them since their priorities are completely different.

This paradox becomes particularly interesting in audio amplification, where the understated 19" rack design that powers a line array in a music festival—and contains 16 high-power state-of-the-art amplifier channels, the most advanced power management technologies with predictive modeling algorithms to manage the power supply, DSP and hardware sensors for real-time information of all parameters—is sold for the less than an 800W at 4Ω hifi monoblock weighing 120lbs generating enough heat to keep a house warm through the winter. And it fascinates me as new flagship developments in audio amplification for hi-fi normally rely on very subjective claims of improved performance and marginally measured incremental improvements for products that already offer very good quality features and specifications.

This might sound like a harsh and cold perspective of the industry but it's one that product developers need to keep in mind when they're asked to design a Class-A amplifier, requiring heavy transformers, large heatsinks, and expensive components, or to design an efficient platform that saves power and is good for the environment. It's all part of the job!

Directly impacting some of those segments, in the last few years we had prominent advancements in Class-D audio amplification. Particularly amplifiers using Gallium Nitride (GaN). This technology offers much higher power, with less noise, and more important for today's requirements, GaN is much more efficient. GaN amplifiers use less power and generate less heat, although they seemingly still need to see improvements in idle losses— as we report in this issue's Market Update.

In an extreme use case for the latest audio amplifier technologies that require maximum efficiency, we find the latest generation audio chips and microspeakers used in smartphones, tablets, and laptops, which are now actually designed to function as fully-featured stereo reproduction systems.

Those use fully integrated smart amps that also include audio codecs, DACs, and sophisticated software features, apart from the power amplifier stage, all engineered to work together to provide the best audio quality experiences possible. And that's where developers find the best jobs opportunities currently in the audio industry. Yes, maybe that's not the engineering job you dreamed of, but that might be the road leading you to the most successful industry achievements—all things considered.

J. Martins
Editor-in-Chief

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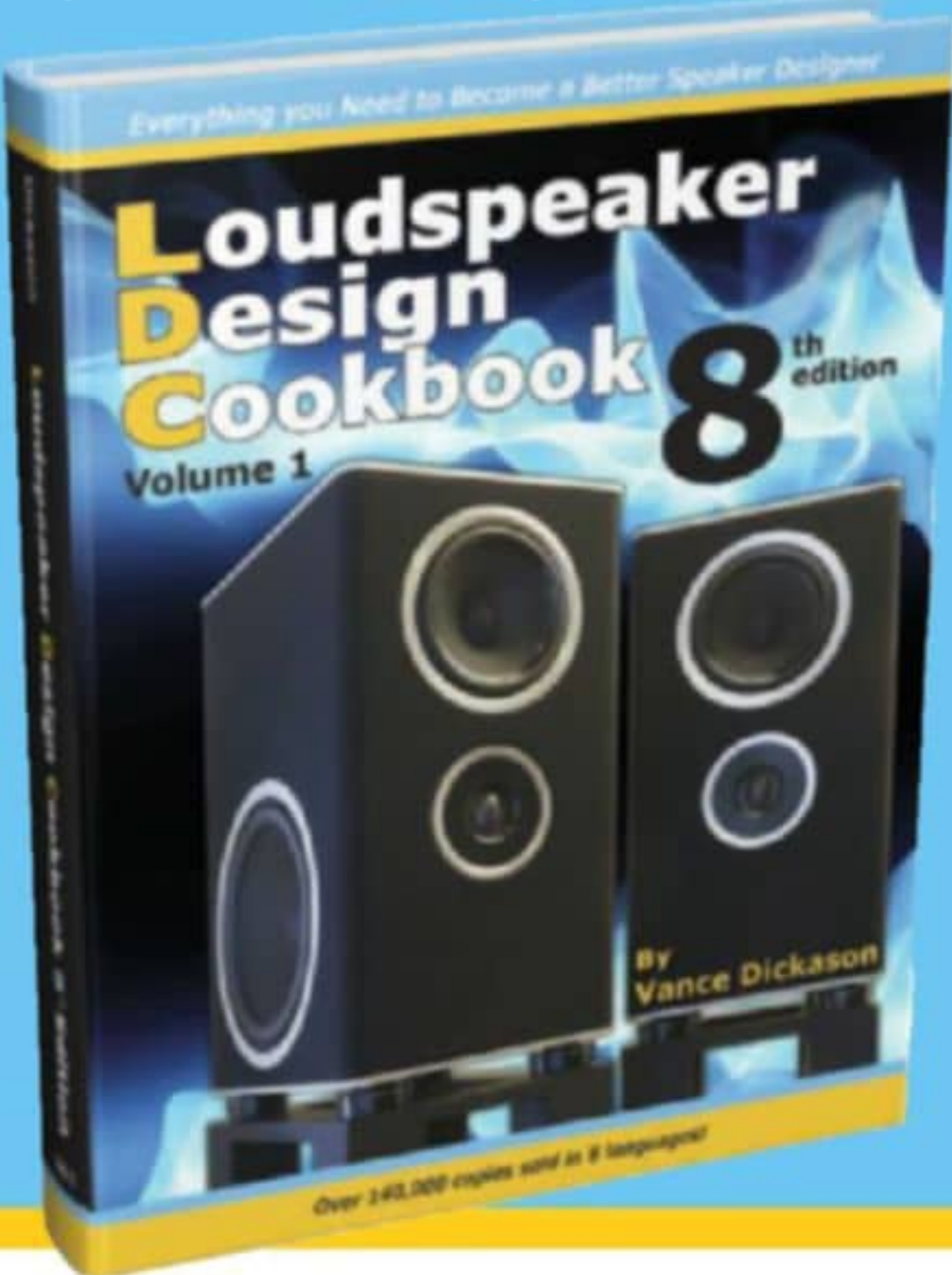
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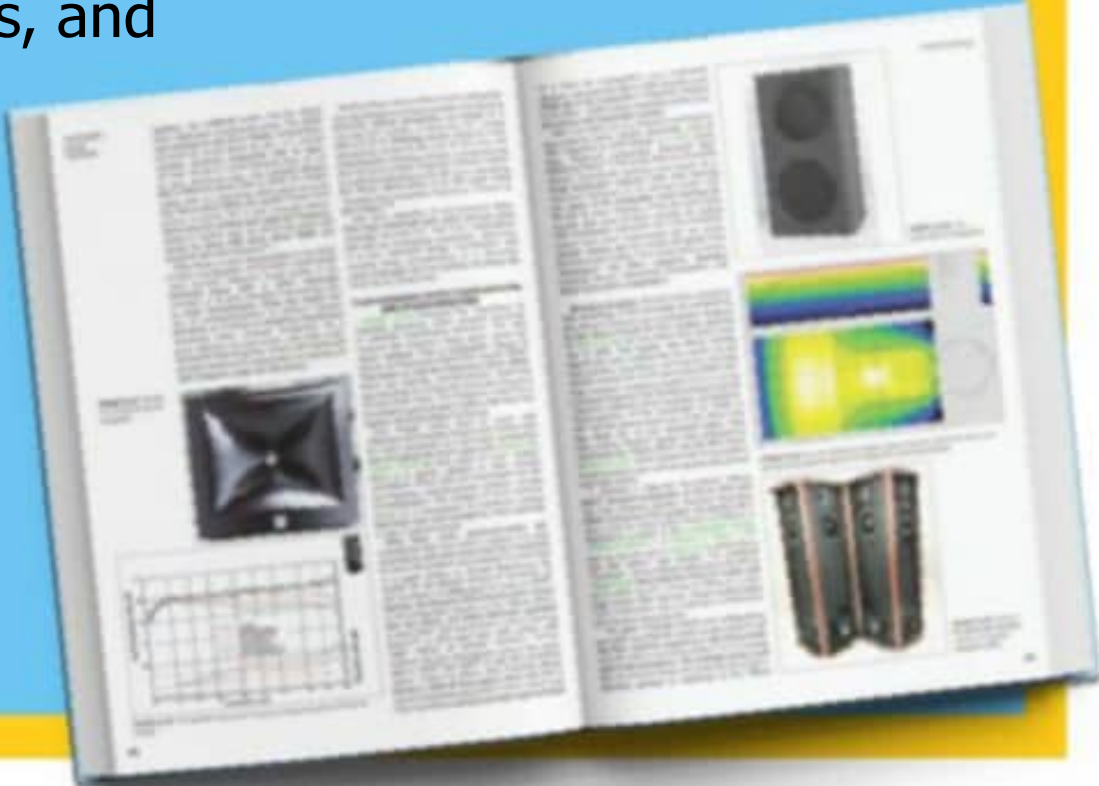
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Vance Dickason's top-selling *Loudspeaker Design Cookbook* is an essential must-read for serious speaker designers, audio engineers, and anyone looking to master speaker-building technology. With more than 140,000 copies in print, author Vance Dickason's *Loudspeaker Design Cookbook* is not only regarded as the loudspeaker engineering "bible," but the #1 resource on the construction of high-quality loudspeaker systems for two-channel, home theater, pro sound PA speakers, and studio monitors, as well as car audio applications.

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May 2023 audioXpress Article Update

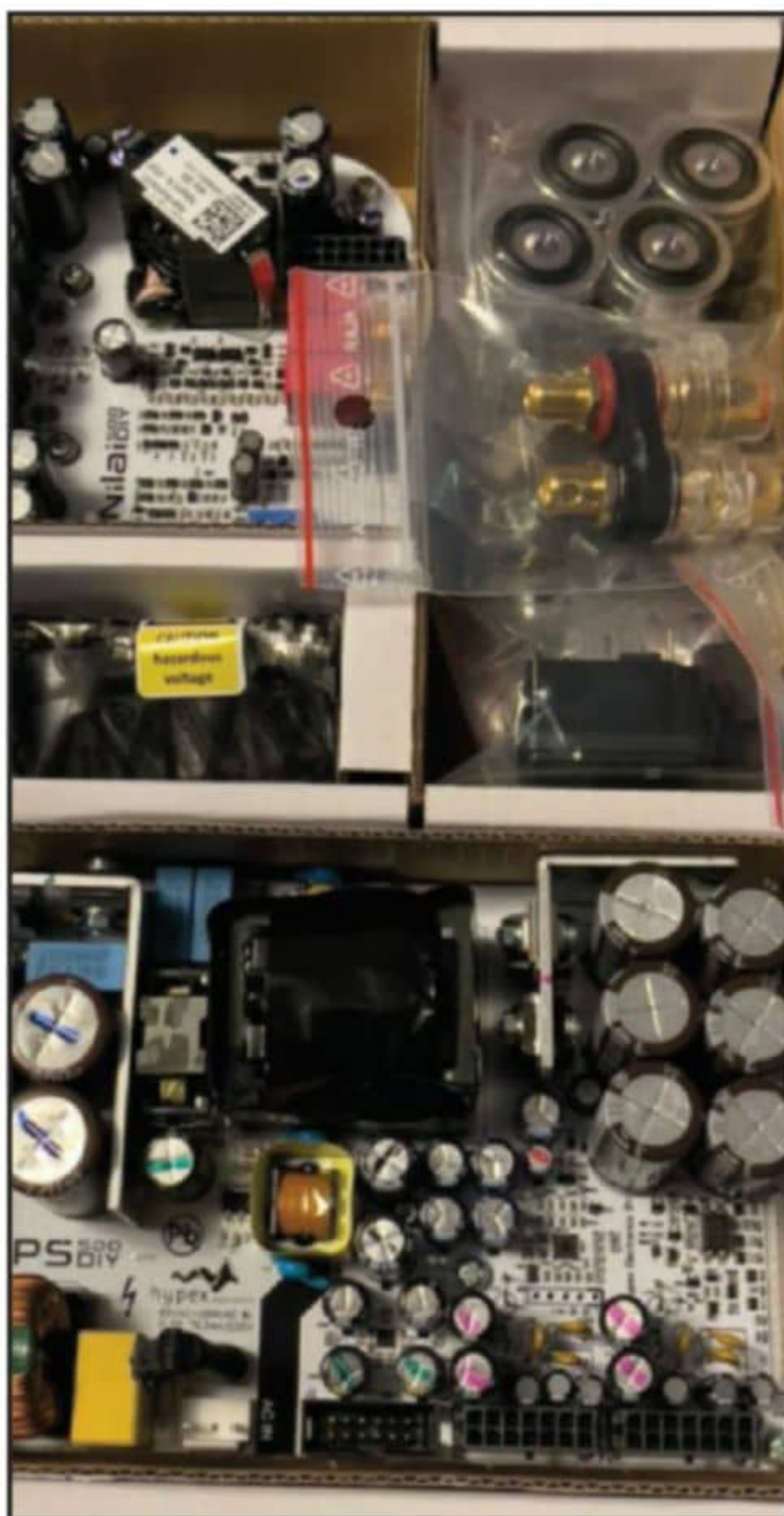
audioXpress received a request from author Menno van der Veen for an update and clarification of "The Vanderveen Trans PP80 Valve Amplifier" article, which was published in the May 2023 issue of *audioXpress*, pp. 21-28:

There is a divergence between the values of R12, which are wrongly indicated ONLY on the driver-PCB where the value reads 47kΩ. The correct value is: R12 = 41k2. In all circuit diagrams and the BOM this value is mentioned correctly.

The START-UP procedure contains a mistake in Step 5. The correct version is:

Check on the driver-PCB on op-amps-1 and -2 at their op-amp-outputs (pin-1 and pin-7) to read less than 5 mVDC with reference to ground.

Menno has also asked that if our readers find other items that are incorrect to please be so kind to let him know at: info@mennovanderveen.nl



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By Jan Didden

Grimm Audio has put its mark on both the pro audio and hi-fi scene. Jan Didden visited the company in The Netherlands to meet the people behind its products, and to learn more about its very unique in-house development philosophy .

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audioXpress interviewed Rémi Guastavino at GRAS Sound & Vibration to uncover the details about its new EQset measurement microphone series.

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Linkplay Technology’s new WiiM Pro is a versatile music streamer. *audioXpress* received a pre-launch unit to review subjectively and objectively.

42 **Room Gain**

By René Christensen

Room acoustics is often discussed in the hi-fi community. In these discussions, the term “room gain” sometimes comes up. This article was inspired

by Erik Wiederholtz (CTO of Perlisten Audio) and his interest in the topic. He also provided measurements and photos from his own listening room for a sealed subwoofer as well as for a vented subwoofer.

46 **Acoustical Measurement and DSP** **Application in Automotive Audio** **Frequency Bands, Sound Philosophy, and NVH**

By Philipp Paul Klose

This article is a continuation from the series “Acoustical Measurement and DSP Application in Automotive Audio.” Here, the author discusses the philosophy, fine tuning of the parameters, and noise, vibration, and harshness (NVH) issues in car cabins.

52 **Bass Reflex Performance Envelope** **Part 1 — The Foundation**

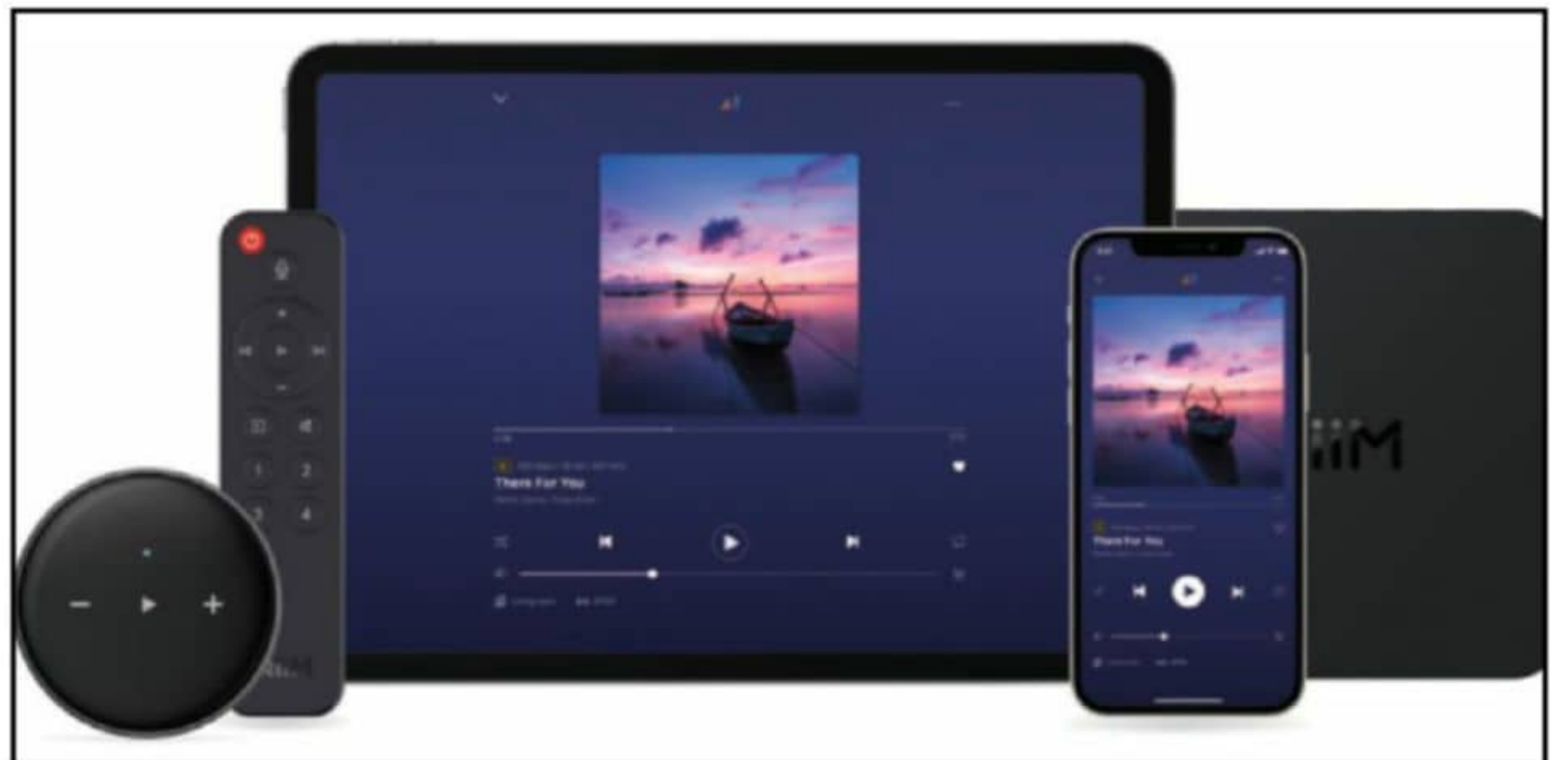
By Gary Gesellchen

This article takes a novel approach to estimating the performance envelope available from bass reflex loudspeakers. Part 1 discusses how bass-reflex works, the analysis approach, and the assumptions.

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● Audio Amplifiers

MARKET UPDATE

8 Evolution of the Species Extended

By J. Martins

Audio amplification is on the move with Class-D not only dominating but improving exponentially. This 2023 Market Update details important technology announcements and some key product launches.

20 Hypex Nilai500DIY 500W Mono Power Amplifier Kit A New DIY Benchmark

By Stuart Yaniger

The highly anticipated Hypex Nilai500DIY 500W mono power amplifier kit is specified to deliver 500W with 1% THD at 4Ω, improves output noise and idle losses significantly, and comes with a discrete buffer stage and voltage regulators, just like the venerable Hypex NC400 module it will be replacing. *audioXpress* offers an extensive review with measurements of the full kit, including the PS500 DIY power supply and DIY preamplifier kit.

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HOLLOW-STATE ELECTRONICS

62 Ten Decades of Home Music Systems A Family History

By Richard Honeycutt

The history of home music systems is directly intertwined with the use of tube electronics and how these devices were crucial to the development of radio, television, sound recording, and reproduction, as well as analog and early digital computers, among many other things. This article revisits part of that history from a personal perspective.



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Evolution of the Species Extended

By
J. Martins



Audio amplification is on the move with Class-D not only dominating but improving exponentially. In 2023, companies are revealing important technology announcements and some key product launches.

In 2022, *audioXpress* published an extensive and unintentionally highly prescient Audio Amplifiers market update, covering an extensive number of new audio amplifier products that had been announced. There was just one minor hurdle—the supply chain and particularly electronic component disruptions continued to be felt throughout the year and extending into 2023, causing the need for many manufacturers to delay product launches and, in some cases, to redesign the product that was originally unveiled.

This is not something we expect will last for much longer, since manufacturers have learned those hard lessons and taken precautionary measures. The consequences though, are that some of the products that we can discuss in this Market Update for 2023 are essentially delayed launches, while others are anticipated unveilings of what's to come.

Not many new products were launched, but a lot has happened in terms of technology in the different market segments. Mostly we are seeing an expansion of the technologies and fields of research that we anticipated in our previous articles. Still there are very significant new facts that make this market update even more exciting.

As an example, in our 2022 Market Update we discussed the excitement of the new Hypex Nilai500DIY amplifier modules and kits, which improved upon all the specifications of the popular NCORE NC400 module, one of the most highly regarded and best-selling products in its segment. Almost at the same time, the Dutch company also announced the new and improved iteration of its popular NCORE Class-D amplifier technology, specifically for the OEM market. *audioXpress* published the review of the NCx500 OEM amplifier module in its May 2023 issue, and only this month are we able to feature the Nilai500DIY review.

And as Hypex shared with *audioXpress*, just in the weeks prior to closing this report, although production is now finally reaching a normal pace, the reality is that these modules remain in limited availability with every production batch being previously sold out.

This is a good problem to have, but it also means that only now product developers are seriously considering embracing these new Hypex modules and technology.

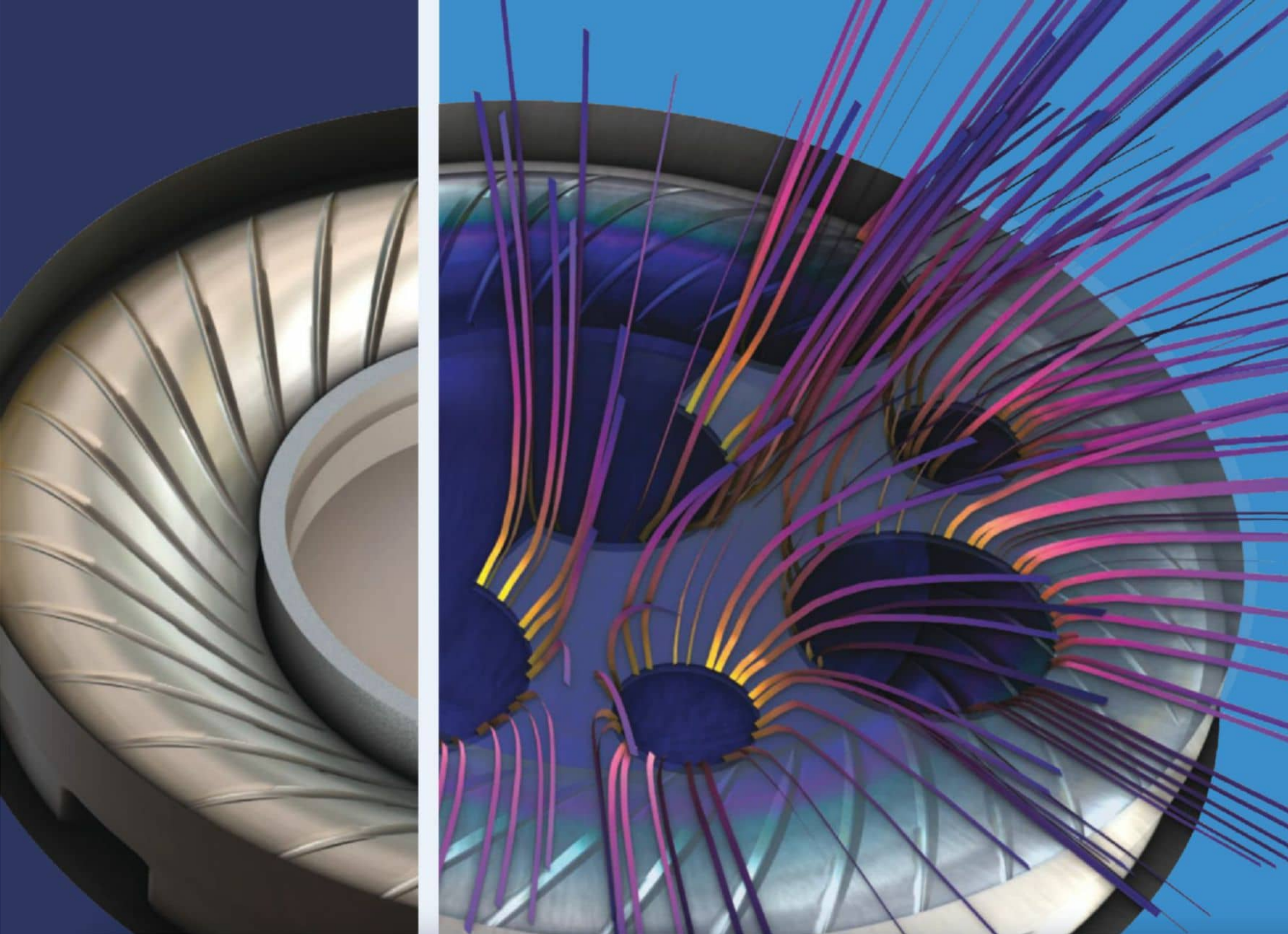
Not to say that the new products that were announced in the past 12 months were not impressive. What this means is that we don't have as many new products to report this year, but we can expect some important and significant launches coming in the next two to three years in high-end audio, hi-fi, and high-quality audio systems for the home, and even for studio and recording.

Another example of a delayed launch was the promised new Purifi 1ET9040BA single-channel Eigentakt board we wrote about last year. Yes, it so happens that even the experienced Purifi team couldn't anticipate the unpredictability of the supply chain extending to 2023, and as we've just learned, this 400W at 8Ω/40A output current bridged amplifier is now expected to become available only at the end of the third quarter. More importantly, the Purifi 1ET9040BA has been submitted to a redesign that improved its performance even more. So that's a product that we look forward to writing about later in the year.

The GaN Space

An important trend that needs to be signaled in the audio amplification market is the consistent expansion of gallium nitride-based solutions being adopted in the audio industry, both in the consumer and professional audio segments. Gallium nitride (GaN) solutions are quickly evolving mainly because of power conversion applications, as available devices provide faster switching speed, smaller size, higher efficiency, and now, lower cost compared to silicon.

Originally, GaN-based power transistors and integrated circuits were adopted in applications replacing MOSFETs and have evolved with applications such as cellular network base stations, LIDAR systems, and obviously power conversion for data centers, BLDC motors, and high-volume automotive applications.



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The SoundImpress by SoundImports DIY amplifier kits and pre-assembled amplifiers combine the best and most popular Class-D amplifier modules and audio components the DIY audio market currently has to offer from Purifi, Hypex, and ICEpower. The company is headquartered in Groningen, the Netherlands.



Schiit Audio designed a differential monoblock power amplifier to top its product catalog. Featuring a choke-input power supply, the Tyr (\$1599) brings “old-skool” techniques to a modern power amp that also includes unique topologies and intelligent oversight and protection, delivering 200W per channel into 8Ω and 350W per channel into 4Ω.



The complete range of Purifi Audio Eigentakt amplifier modules, including the original (and now extremely popular) 1ET400 module, seen in the photo with the connectivity boards for one and two channels. The photo also shows the 1ET7040SA OEM module (500W at 4Ω/250W at 8Ω 70V/40A), next (on the right side) to the yet-to-be-released and completely redesigned 1ET9040BA bridged amplifier module.

Now, the market for GaN solutions is exponentially expanding due to its applications in fast chargers for consumer electronics, and obviously electric-powered automotive/mobility systems and renewable energy. Also, because GaN is inherently radiation tolerant, space applications including new-generation low-orbit satellites are a natural field for the technology.

GaN as a technology is in its infancy, but the larger players are all taking notice and eager to explore its superior efficiency, switching speed, and reduced size. We can expect to see both acquisitions of the key GaN pioneers by large semiconductor companies, as well as the creation of specialized manufacturers to leverage existing devices in audio designs.

Class-D audio was one of earliest test-beds for GaN devices, but it was only in the past five years that audio manufacturers have started to look at the available GaN FET and IC devices from basically three companies: two pure-GaN players, GaN Systems and Efficient Power Conversion Corporation (EPC), and Wolfspeed (originally Cree Research).

In the next few years, we will start to see audio manufacturers leveraging these devices to target different requirements, including smaller form factors for portable applications, very high power for professional audio, and overall better audio performance for home audio. In every field, designers are now starting to discover ways to deal with the thermal challenges that these small devices still originate, which remain a challenge particularly for integrated applications, such as powered speakers.

EPC’s GaN devices are promoted directly for Class-D audio applications and the company offers three reference designs for amplifiers up to 200W or 100W in two-channels (8Ω). According to the company, its solutions already enable double the power with the same/smaller footprint up to 500W per channel with lower idle power consumption and easier multichannel scaling.

EPC basically provides the discrete and integrated GaN solutions that are the building blocks to enable audio designers to create new innovative products with already demonstrated superior audio performance, such as better SNR, lower THD+N, and lower T-IMD. Wolfspeed, a pioneer in silicon carbide (SiC) applications is now investing strongly in GaN for many applications but has not yet directly targeted Class-D audio amplification as the two other companies did. Wolfspeed offers a wide range of devices for power electronics used in electric vehicles (EVs), electrical grids, and portable power applications. But the fact that Wolfspeed still doesn’t offer reference designs for audio applications doesn’t mean that developers are not looking at its low-cost solutions—already used in test instrumentation—as another alternative in GaN-based amplifiers.

The most recognized GaN player for audio applications is undoubtedly GaN Systems, from Ontario, Canada. *audioXpress* has reviewed and reported extensively about successful audio designs that are using GaN Systems’ technology, and there is a visible growing interest in the audio development community to explore its evaluation boards.

In a major announcement with consequences that are still to be fully understood, in March 2023 Infineon Technologies announced a definitive agreement to acquire GaN Systems for \$830 million



The EPC9106 development board offers a 150W/8Ω Class-D audio amplifier using a Bridge-Tied-Load (BTL) design, composed of four ground-referenced Half-Bridge Output Stages featuring the EPC2052 GaN device from EPC.

USD. GaN Systems has established itself as a technology leader in the development of GaN-based solutions for power conversion and has more than 200 employees. Of course, with this new acquisition, the German semiconductor giant is targeting the major GaN applications and most likely, Class-D amplification was not even a major consideration.

GaN Systems was very successful with the strategy of GaN-powered audio and the continuous release of Class-D high-performance GaN FET amplifier boards and audio boost converter solutions. The news of this acquisition means that Infineon will now become a leading component supplier of audio amplifications solutions.

The German company already leads the segment with very efficient Class-D solutions and expanded that leadership for battery-powered audio designs since the acquisition of the Danish startup Merus Audio and its multilevel amplification technology. With the acquisition of GaN Systems, Infineon will now be able to significantly extend that portfolio to higher power categories.



Together with its own reference designs, the GaN Systems suite at CES 2023 offered the first public demonstration of new Class-D amplifiers developed with Dutch company Axign, currently the most powerful available in the market using GaN technology.

According to Jim Witham, CEO of GaN Systems, Infineon will enable exploring the full potential of GaN with its in-house manufacturing capacity and unleash the full potential of its technology. For now, GaN Systems is still operating independently, and the company continues to promote its partners' achievements for audio applications.

Resources

Efficient Power Conversion (EPC) Corp. | <https://epc-co.com/epc>

Elegant Audio Solutions | www.elegantaudiosolutions.com

GaN Systems | www.gansystems.com

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Schiit Audio | www.schiit.com/products/tyr

SoundImports | www.soundimports.eu/en/brands/soundimpress

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Infineon Technologies

Ultra-Efficient 2x 37W MERUS Multilevel Switching Audio Amplifiers

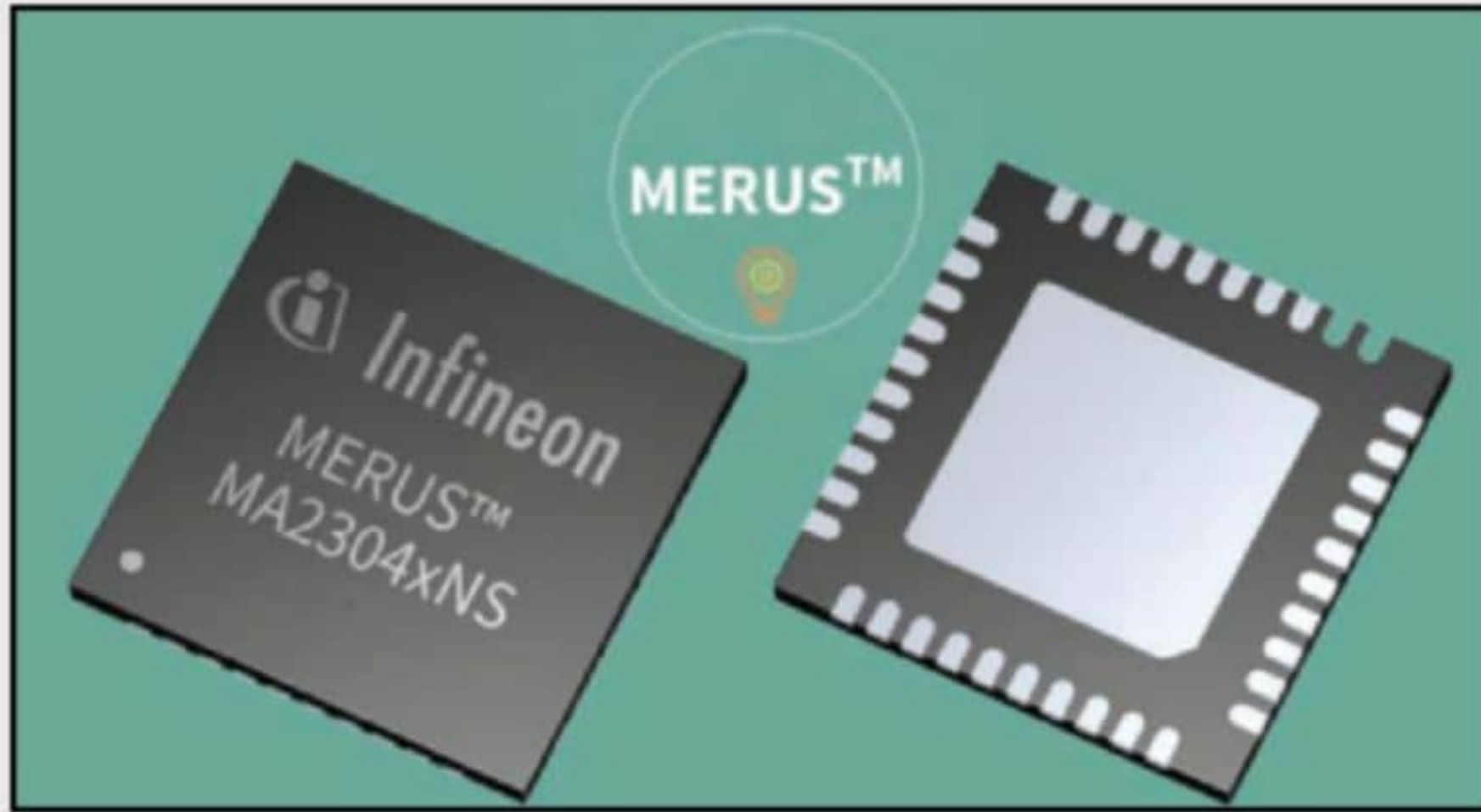
Since acquiring Danish startup MERUS, Infineon's range of amplification solutions based on multilevel switching amplifier technology continued to expand and currently offers a very attractive platform for consumer audio.

When designing portable and battery-powered audio devices for consumer speaker applications, maximizing battery playback time and, at the same time, maintaining excellent audio performance in a compact form and at a low system cost is essential.

MERUS multilevel switching technology from Infineon is designed to achieve low switching losses and highly efficient amplification at both low and high output power, bringing the desired benefits in such applications.

Recently, Infineon launched a new family of 2x 37W audio amplifiers featuring the second generation of MERUS multilevel technology, optimized for low power and battery-powered applications. The new devices are now the most efficient in the Infineon MERUS family, with an idle power mode at least five times lower than traditional Class-D audio amplifiers.

The new family consists of MA2304DNS with an integrated and fully configurable digital signal processor (DSP) and MA2304PNS with audio limiter and volume control features. Both devices feature the second-generation multilevel-switching Class-D audio amplifiers with an inductor-less operation and provide a bridge-tied load (BTL)-rated output of 2x 37W and a power supply voltage range (PVDD) of 10V to 20V for



The new ICs feature the second generation of this technology and come in a compact 40-pin QFN package. They primarily address applications such as battery-powered speakers, Bluetooth/wireless/smart speakers and soundbars, conference speakers, and multi-channel/multi-room audio systems.

high-fidelity audio performance. The five-level voltage modulation offers an ultra-low idle power of 52mW without the need for complex dynamic rail-tracking systems. This is at least five times lower than the traditional Class-D audio amplifiers, making these devices ideal for battery-powered speaker applications with extended battery life and/or reduced battery cell cost.

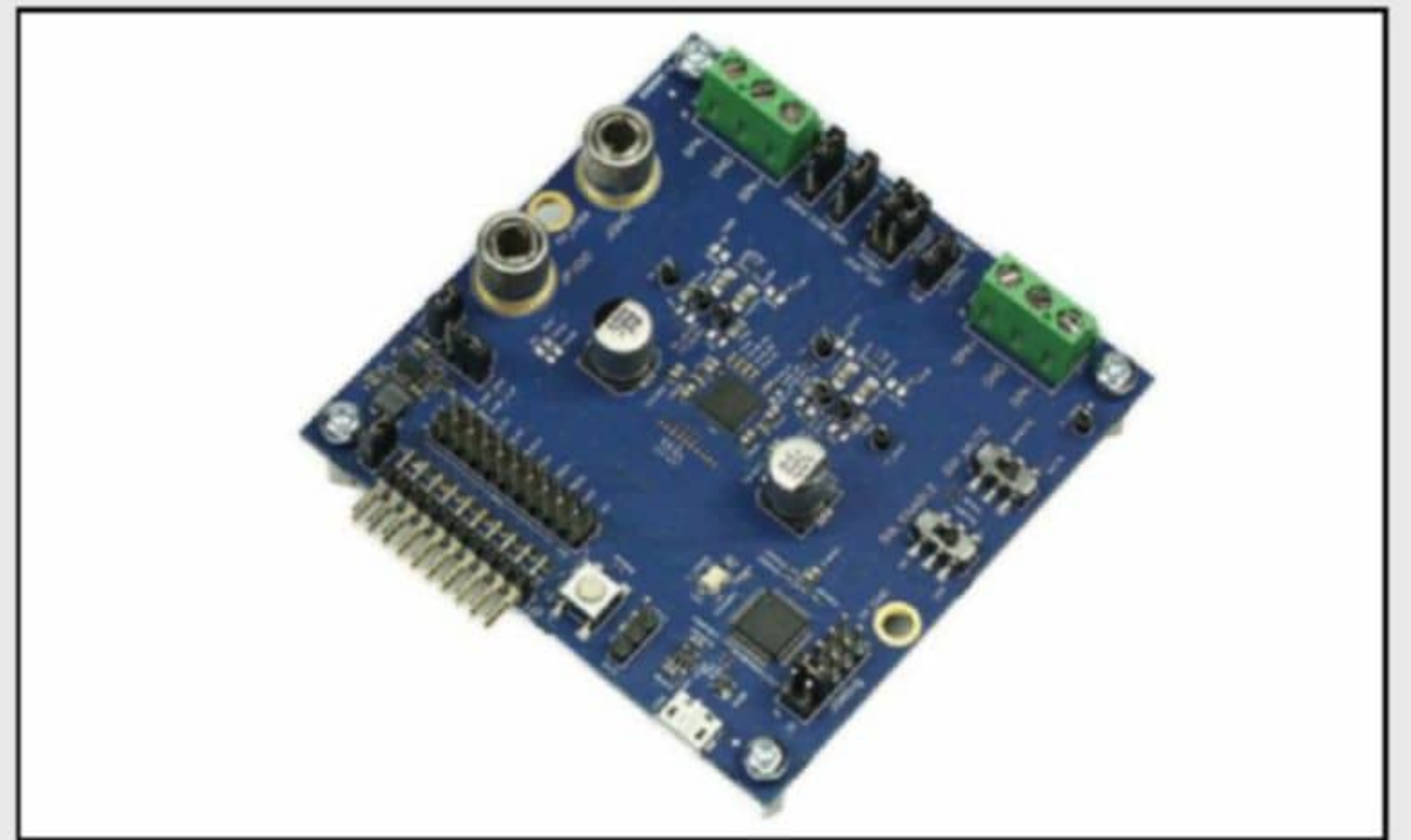
At low output power, the efficiency can be higher than 80% offering extended battery life and easy thermal management in multi-channel designs. Multilevel switching also relaxes electromagnetic compatibility (EMC) requirements and enables inductor-less applications at full power range with lower cost. Compared to traditional two- and three-level Class-D audio amplifiers, the reduced electromagnetic interference (EMI) emissions allow for fast time-to-market without compromises in audio performance or efficiency. In mains-powered multi-channel applications, the reduced and scalable EMI enables otherwise impossible industrial designs without needing a heatsink or a traditional LC filter.

A high-order internal feedback loop ensures low total harmonic distortion (THD) and low noise floor for excellent audio performance. The inductor-less application contributes to the smallest output filtering, avoiding bulky inductors, thus reducing system cost without output power limitations. The integrated DSP in the MA2304DNS is highly configurable and can correct imperfections and tune real speaker applications with equalization, limiting, and other similar functions. The MA2304PNS offers a simpler volume control and limiter for systems where an application/connectivity processor does the audio processing.

MERUS Partner Ecosystem

With the new MERUS devices, Infineon also announced new ecosystem partners that are ready to support interested developers and manufacturers with their design-in efforts. Partnering with the experts, the company aims to boost the adoption of these new amplifiers with a unique Class-D modulation of up to five different output voltage levels.

Two major associated partners of Infineon, currently developing unique audio solutions based on the MERUS Class-D



Evaluation boards are available for the MERUS audio MA2304DNS and MA2304PNS ICs. The board contains a digital input and a variety of output and setup/selection features. It also contains an on-board power supply generator (selectable 1.8V or 3.3V buck-converted) so only one external power supply (PVDD) is necessary. It can be controlled and programmed with its software GUI.

audio amplifiers, are Eleven Engineering and Volumio, in the Global and EMEA business regions, respectively.

Eleven Engineering, the Canadian company behind the cutting-edge wireless audio standard SKAA, supports manufacturers and brands with high-quality audio designs in key audio markets such as home theater, whole home, portable audio, and outdoor. As the creator of the SKAA technology for wireless audio, the Infineon partnership extends the company's range of engineering services in both wireless networking and battery-operated speakers, leveraging Eleven's off-the-shelf motherboards, Strawberry and Saskatoon, combining SKAA and MERUS silicon on field-proven PCBAs.

Volumio is the company that created the free and open-source Linux music player of the same name, which meanwhile has evolved into a complete catalog of products, from DIY "build your own music players" based on a Raspberry Pi, Tinker Board or PC, to a range of dedicated music streamers, such as the Volumio Primo, Rivo, or Integro. Volumio also offers OEM and engineering services in system integration and licensing. "We have chosen to design our Integro integrated streaming amplifier around the MERUS multilevel Class-D audio technology because it offers the best synthesis of sound quality, efficiency, and power," says Michelangelo Guarise, Founder and CEO of Volumio. "Our customers just love that such a compact design can deliver a powerful and crisp musical presentation, and this has been possible thanks to the MERUS platform."

Volumio provides its music players designed for high-quality music playback, controlled with any device with a browser, and offering an intuitive UI that allows the integration of different music formats and services. Volumio OEM provides a tailored audio system to companies based on the Volumio software, already trusted by 30 OEM partners.

Eleven Engineering | www.elevenengineering.com

Infineon | www.infineon.com/merus

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Peak Amplification

Elytone Electronics Creates New Audio Amplifier Division

Elytone Electronics, an OEM/ODM manufacturer from Taiwan and provider of custom audio solutions and magnetic components for more than 40 years, has introduced a new range of networked audio products and launched a new division and new brand Peak Amplification, offering a complete range of high-power amplifier modules.

As Elytone states, the strategy signals the beginning of a transformation for the company, launching sophisticated network audio products that will enable a wide array of connected, networked applications with advanced audio applications, powerful DSP, and Power-over-Ethernet (PoE) capabilities. In particular, the Networked Audio Processor 1 (NAP1), is a powerful complete audio and DSP processor with an ARM A5 Cortex processor and an Analog Devices SHARC DSP with on-board memory. The NAP1 is already getting significant interest for a diverse range of end products, showing its versatility for networked audio applications.

The Elytone team has also created a range of Dante audio networked systems using multiple NAP1s. Specifically in the new Peak Amplification brand, Elytone is introducing an extended portfolio of Class-D amplification products from 300W to 1800W in single- and two-channel configurations. The Peak Amplification portfolio consists of several amplifier series, each tailored to different performance, price, and power ranges and intended for different product applications.



For now, Peak has launched the “Integrated” series, consisting of single- and two-channel high-performance Class-D amplifiers, which incorporate a compact auto-switching switched-mode power supply (SMPS). The modules are optimized for integration within many audio applications and include features such as universal mains, PFC/Simple Control, OCP/OTP, and are safety and EMC compliant. These modules can be combined as building blocks to generate a wide variety of 1 to 8 channels applications in professional, consumer, and commercial audio products. The Integrated models include two High Power two-channel amplifier modules with integrated universal SMPS with PFC, rated from 1000W to 1800W, with support for Hi-Z and low-Z modes. There are also multiple Superior Performance, Best Value, and dedicated Subwoofer modules. The small form factors and efficient power of these modules will support from studio monitors to high power, high performance professional amplifiers.

All Peak Amplification products build on Elytone’s decades of experience in amplification design and manufacturing, and all are supported by rigorous quality control and certifications.

Peak Amplification | www.peakamplification.com

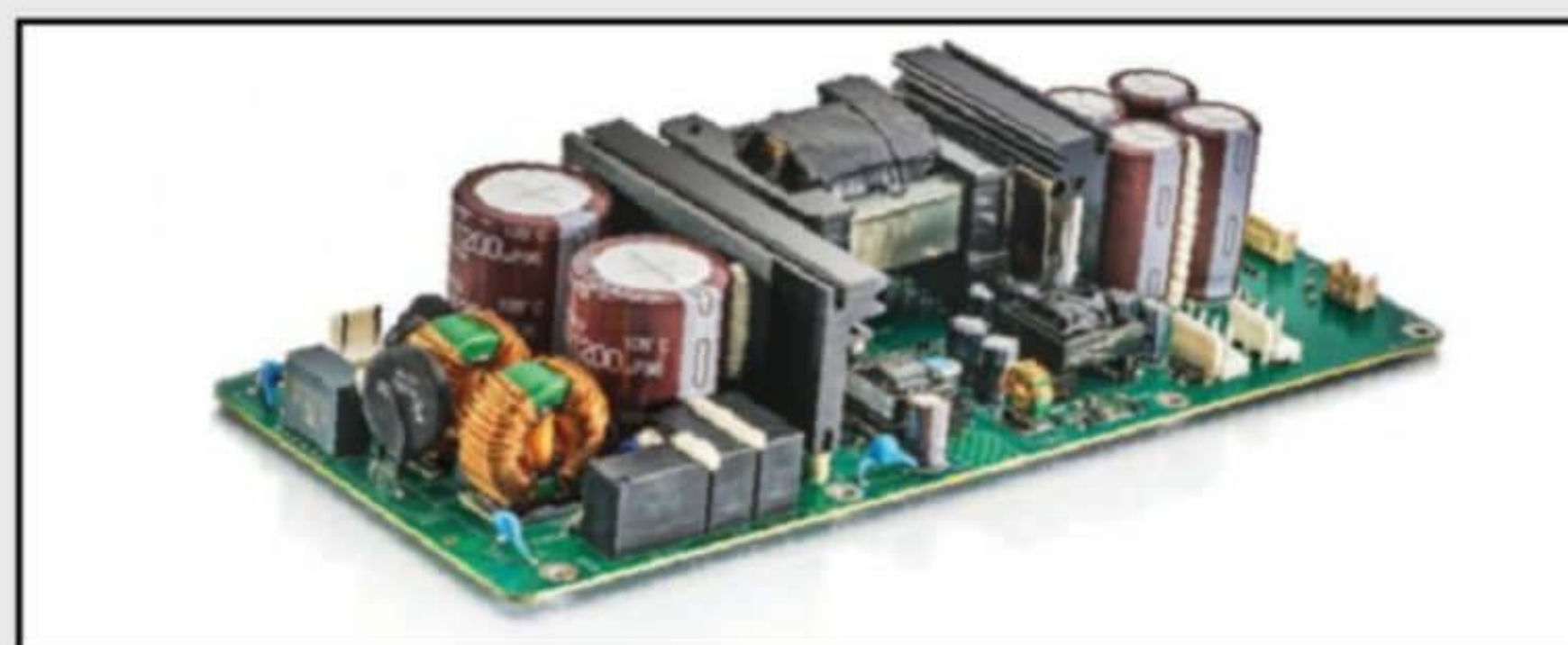
ICEpower

New 1000S Power Supply for Two-Way and Three-Way Designs

In 2022, in addition to the 1500S power supply, ICEpower released the 300A2 and the 400A2 amplifier modules, which the company didn’t hesitate to call its best-sounding amplifiers produced to date. While the 1500S has more than enough power to run up to 16 channels of amplification, not every application requires that many outputs. That is where the 1000S comes in.

Based on the resonant LLC converter of the tried and tested ICEpower 700AS1 module, the new 1000S offers a stable and consistent power supply solution in a compact format to pair with the company’s 300A2 and 400A2 modules for applications in hi-fi, commercial amps, whole home audio, and active speakers.

According to the Danish company, the demand for a 1000W power supply has been requested for a long time and its new 1000S provides an essential component for audio applications such as two-way, three-way active speakers, and hi-fi stereo amplifiers, which will enable audio manufacturers to design products with more power and features than ever before.



The 1000S power supply is a highly efficient design, ensuring that it runs cool and quiet even when under heavy loads, making it ideal for audio applications, providing 1000W and capable of peaking 1400W.

Available in a range of voltages, the 1000S enables product and system designers to combine it with the existing 300A2, 400A2 modules or a custom-designed amp on a wide variety of applications, including for 70V high-Z applications. For a best-in-class hi-fi amp, ICEpower suggests pairing the 1000S with a 400A2 for two channels of 400W. For high-powered mono amplifier applications, the 1000S can be used to power a bridged 400A2 for 800W at 8Ω.

The 1000S features universal AC input, a standby power supply compliant with the latest EU standards and strong 5V and ±15V auxiliary supplies for pairing with the best preamps and DACs. The board is pre-approved for safety, EMC, and EMI.

ICEpower | www.icepoweraudio.com

Linear Systems

Low-Cost, Low-Noise LS844 JFET for High-Performance Applications

Expanding its efforts to reassure the audio and electronics industry about the stability of its supplies and product availability, Linear Systems announced the re-release of the low-noise, low-cost LS844 series dual JFET with immediate large-scale availability and large-volume pricing. The re-released LS844 is the latest evolution of a classic dual JFET developed and introduced by Linear Systems more than 20 years ago. The company has refined the part's design and fabrication processes to improve its low noise, matching and common mode rejection ratio (CMRR) qualities while retaining the linearity and other characteristics sought by audio and other circuit designers.

The LS844 is a low-noise junction field-effect transistor (JFET) ideal for precision, high-end audio, sensor, and discrete operational amplifier applications. It features a unique design construction of interleaving both JFETs on the same piece of silicon to provide excellent matching and thermal tracking, as well as a low-noise profile having nearly zero popcorn noise and lower intermodulation distortion.

"The release of this high-performance dual JFET at a low price enables design-ins for high-volume production," Linear



The LS844 is now available for \$1.19 USD each in quantities of 1,000 pieces in the SOT-23 6L package.

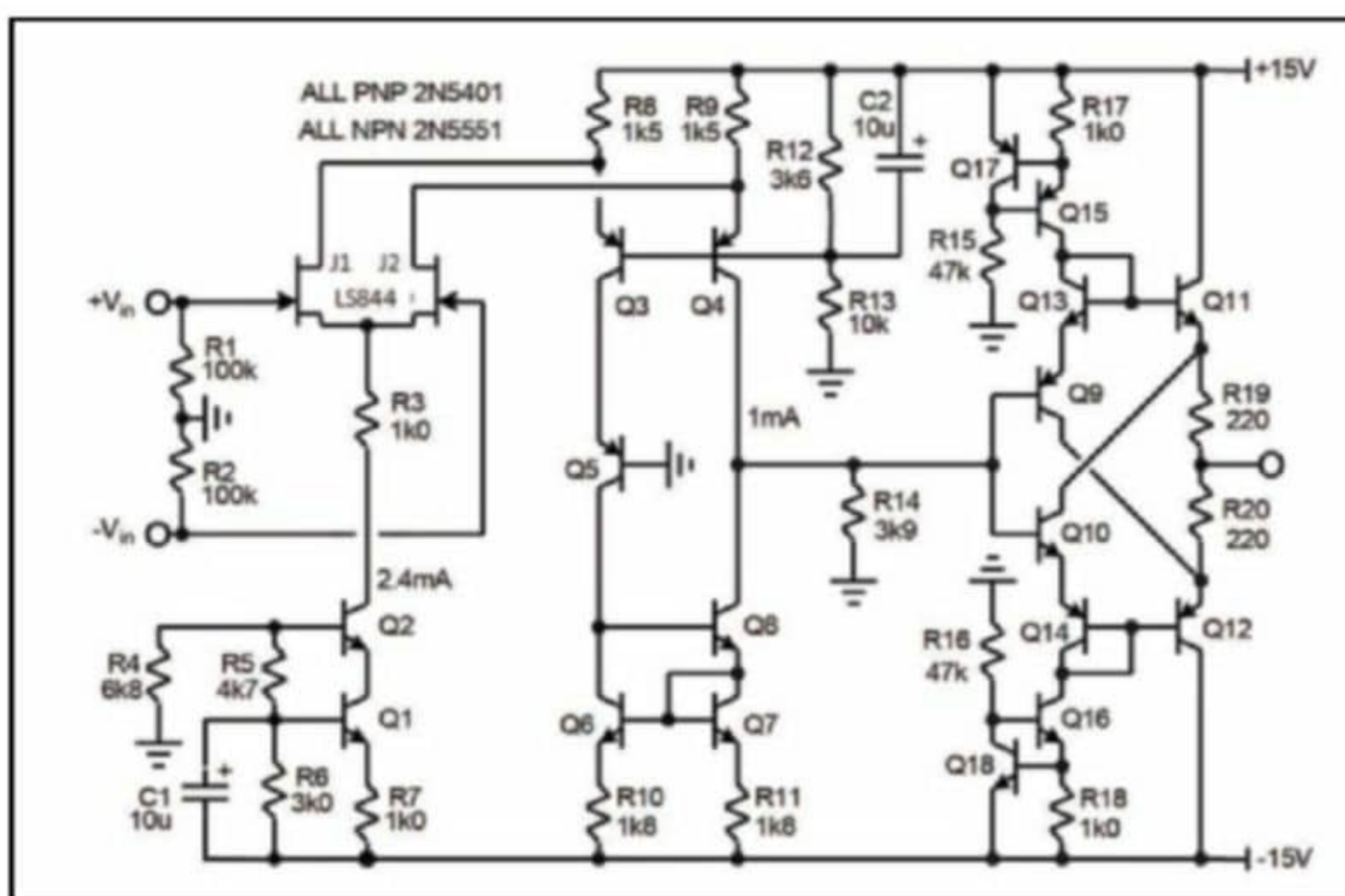
Systems President Tim McCune says. "Applications ranging from microphones to sensor inputs based on the LS844 will have vastly better signal clarity than circuits based on other options at this price point."

Applications for the LS844 include wideband differential amplifiers, high CMRR front ends, temperature compensated single-ended input amps, tight tracking current sources, low jitter high speed comparators, impedance converters, acoustic sensors, piezo-based sensors, sensitive charge amplifiers, and much more.

The LS844 is part of Linear Systems' family of low-noise JFETs. The lead parts of this family are the LSK389 dual series and the LSK170 single series ultra-high-performance JFETs. The N-channel dual LSK489 has slightly higher noise than the LSK389 but lower input capacitance than the LSK389. In addition to the LSK and LS series, all Linear Systems' JFET standard products have substantially lower noise than those made by other manufacturers and are marketed under the Improved Standard Products trademark.

A detailed datasheet and a valuable application note written by Bob Cordell with circuit examples for pre and power amplifiers, microphone preamps for dynamic and condenser designs, phono preamp, and much more is available from the Linear Systems website.

Linear Systems | www.linearsystems.com



This schematic shows a discrete JFET amplifier that employs a folded cascode and a diamond buffer output stage. This is much like the low-noise discrete input stage of Bob Cordell's VinylTrak phono preamplifier.

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New Amplifier Modules for Class-D Audio Solution

Already in 2023, GaN Systems extended its lead in Class-D GaN-powered audio with the release of two complete Class-D high-performance GaN FET amplifier boards. GaN Systems now provides a comprehensive solution of turnkey amplifier designs, allowing audio systems designers across automotive, home, and professional markets to mix and match designs to maximize performance for specific applications.

The new designs are optimized for sound quality, efficiency (no heatsinks), size, and cost—enabling audio design engineers to create premium audio products more quickly at affordable prices while reducing time to market. These new designs leverage GaN Systems GS61008P GaN power transistors. GaN Systems' power transistors are critical components of next-generation audio products and address essential needs in the Class-D audio market for consumer electronics and smart home designs, as well as car audio.

The OL200DS and CL200M are self-contained 200W per channel (8Ω) Class-D amplifier modules designed for powered loudspeakers and stand-alone stereo and multichannel amplifiers. The GaN Systems OL200DS is designed as two independent half-bridge outputs for both stereo and single-channel applications. The GaN open-loop approach provides an ideal switching form for Class-D amplifiers, delivering a dual feedback loop that performs better than a single loop/closed filter. The OL200DS also supports a quad half-bridge topology for 50W per channel.

Similarly, the new GaN Systems CL200M is a 200W (8Ω) dual half-bridge or bridge-tied-load (BTL) module for ground-

referenced output. Thermal protection is provided with the CL200M for worst-case thermal environments, and 96% efficiency reduces heat and system size.

Two new high-voltage boost converter reference designs also include a 210W continuous/300W peak power source and a 300W continuous/500W peak power source for manufacturers of stand-alone mono, stereo, and multichannel automotive and marine amplifiers. Complete audio boost converter solutions include wide-range DC supply voltage input from 9VDC to 16VDC operation, 32V output, and are load dump compatible with differential-primary-drive transformer topology.

GaN Systems | www.gansystems.com

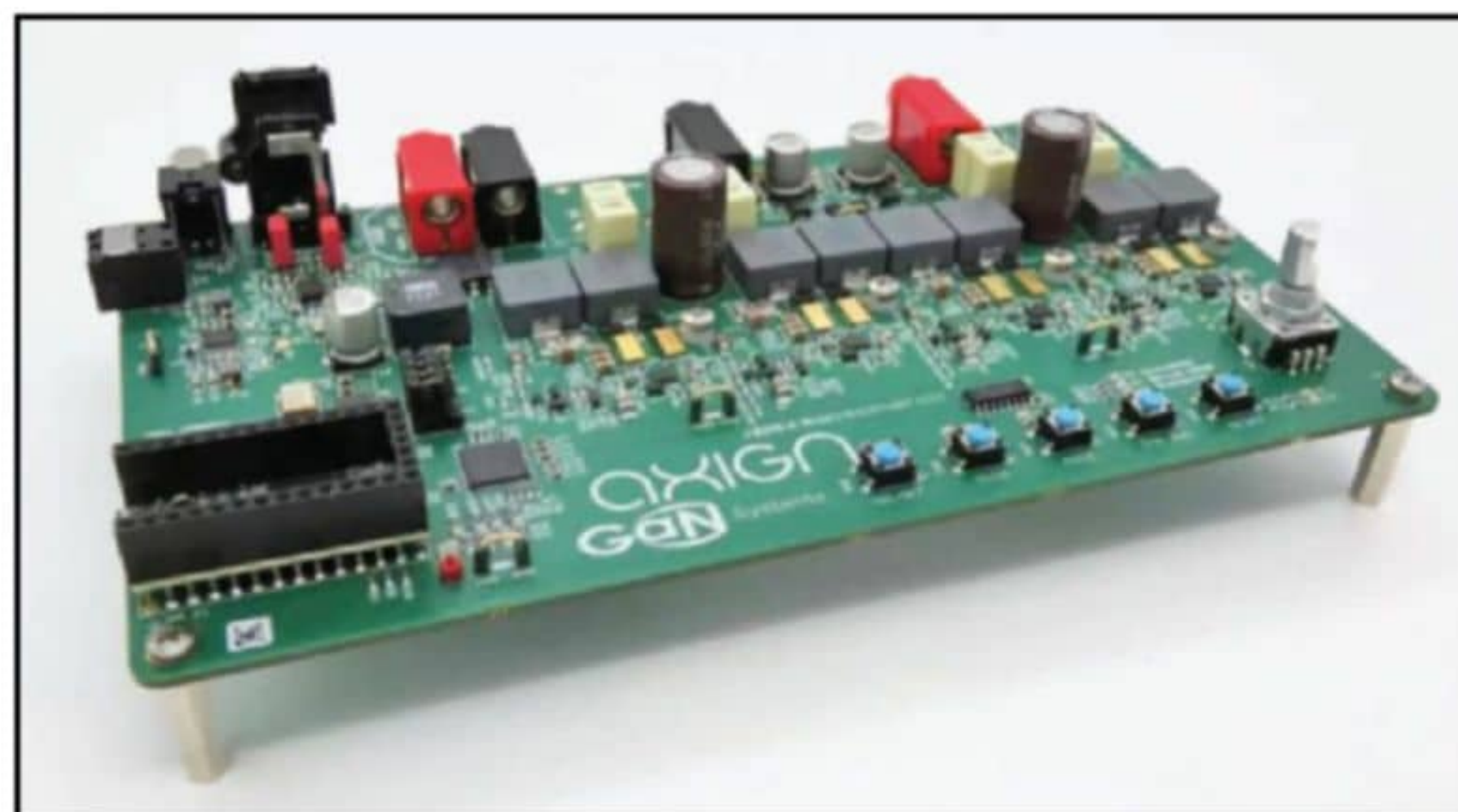


GaN Systems 1000W Class-D Heatsinkless Audio Amplifier Design

GaN Systems has expanded its collaboration with Dutch company Axign to unveil an even more powerful, 1000W Class-D GaN audio amplifier board. The new reference design, announced at CES 2023, merges the more recent technologies from GaN Systems and Axign to enable smaller, more efficient, audio systems that offer better audio quality for automotive, professional, and prosumer audio applications.

The cooperation between GaN Systems and the Dutch fabless semiconductor company with its audio amplifier feedback loop controller ICs has already delivered an impressive range of reference designs serving all applications. In January 2022, the two companies have announced a groundbreaking GaN-based 500W Class-D audio amplifier reference design, which complemented the two-channel, 200W per channel (8Ω) Gen2 Class-D audio amplifier board, and companion 400W audio-grade SMPS combined solution from GaN Systems—designed by Skip Taylor and Elegant Audio Solutions.

For the 2022 500W GaN reference design and the latest 1000W audio amplifier board, the design was developed internally by



The new Axign/GaN Systems 1000W Class-D audio amplifier optimizes total idle losses for the GaN power stages and LC output filters.



The new Axign sound processors will reduce system size and system cost, to compete with the biggest semiconductor players in audio applications.

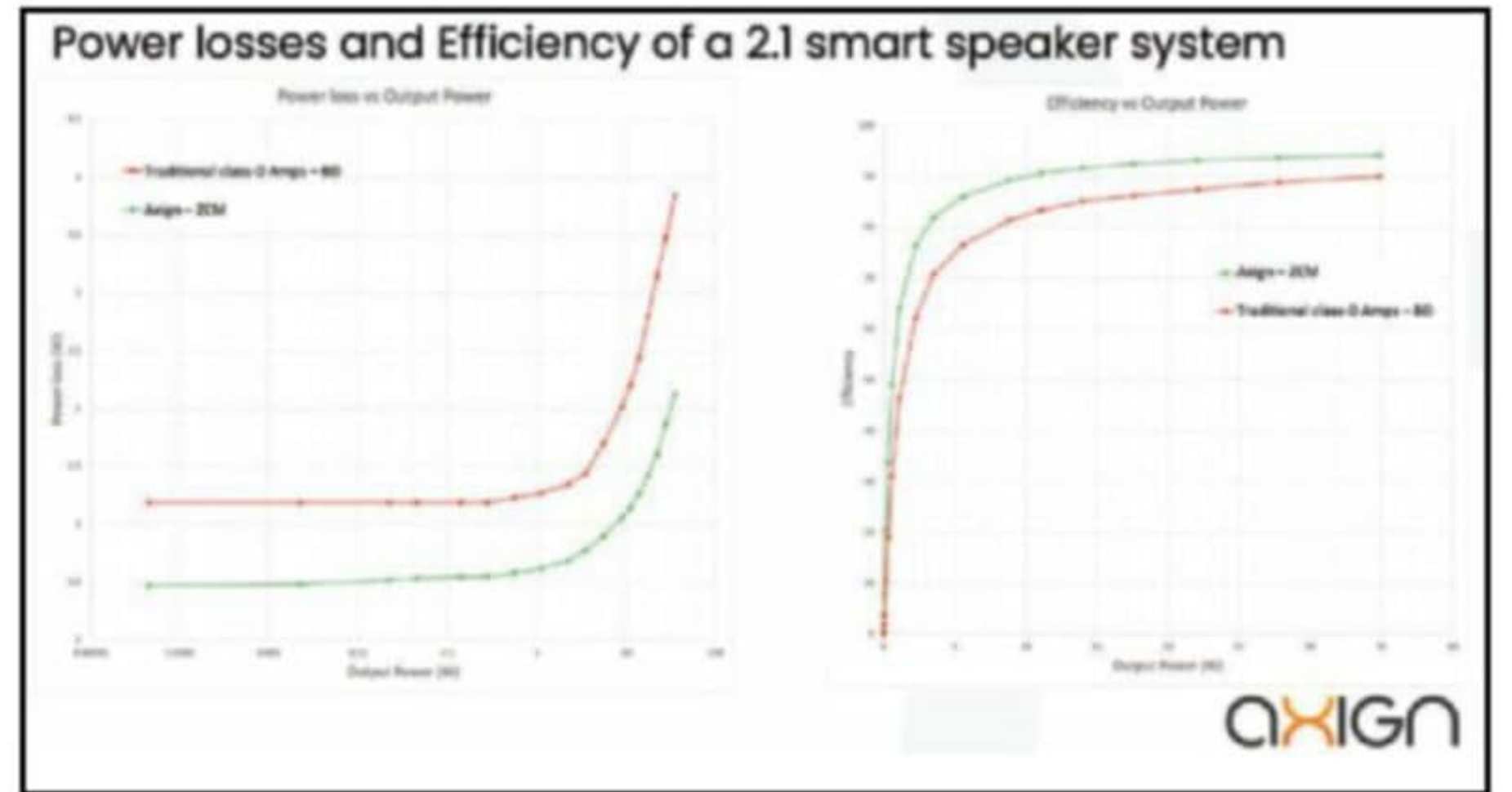
the Axign engineering team, to directly confront the need for optimizations detected in previous projects using GaN devices.

The new Axign/GaN Systems 1000W Class-D audio amplifier stands out due to its ability to run cool while still being able to deliver 1000W at 1 Ω in parallel bridged-tied load (PBTl) at 50V or 2 Ω PBTl at 70V without a heatsink. According to Axign, the powerful audio amplifier board can work in 2 $\times$ 500W mode with 97% efficiency, or bridged to full 1000W with 98% efficiency, with less than 3W of total idle losses for the GaN power stages and LC output filters. In addition to these key features, developers can explore its unique post-filter feedback, 50V, and 70V output capability, 70% reduction in idle losses, and 50% reduction in power loss.

This means that this new Axign amplifier will help audio designers solve the challenges of efficiency and heat generation in high-power GaN audio amplifiers. This enables designs that respond to highly demanding applications, such as high-end



Axign's products have been adopted by Harman's Luxury Audio division, and JBL has recently selected the AX5689 chip for the 4305P and the 4329P Studio Monitors, both active and connected speakers for home audio applications. These self-contained models are fully optimized to translate high-resolution audio streaming sources.



Comparison of power losses and efficiency between the latest Axign audio amplifiers and traditional Class-D (Image Source: Axign)

systems working with high-resolution audio sources or active speakers.

"Achieving superior sound quality with very high efficiencies and low idle losses in a better and smaller form factor are just a few advantages of our 1000W heatsinkless Class-D audio amplifier with GaN Systems," says Richard Langezaal, VP of Business Development for Axign.

Axign Advanced Sound Processors

With this new development, Axign stands at the front of audio amplification optimization efforts with GaN technology, even though its focus and approach might be applied with similar benefits to conventional MOSFET designs.

For Axign, it makes complete sense to stay in the digital domain all the way to the loudspeakers and the company's post-filter feedback loop technology for Class-D amplifiers with balanced feedback from the loudspeaker is now discreetly already powering existing products from major manufacturers and still has considerable potential for improvement.

Power consumption reduction is one of Axign's key focus areas. Idle losses are unwanted power losses, which produce heat or discharge the battery faster. Due to Axign's digital control loop and its post filter feedback, idle losses can be reduced up to 15 times compared to previous modulation schemes and solutions. This enables high-power GaN and MOSFET-based audio amplifier solutions to remain cool, and that is precisely what the current AX5688 and AX5689 controllers are doing already in products available in the market.

Still using its AX5689 Class-D controller, Axign's 2 $\times$ 500W (2 Ω BTL) reference design with GaN FETs delivers class leading audio performance with less than 1.5W idle power consumption per BTL power stage channel and LC components. Axign also demonstrated a 2 $\times$ 250W (4 Ω BTL) MOSFET solution for cost-optimized applications that delivers class leading audio performance with 95% efficiency at 250W in 4 Ω , and less than 1.5W idle power consumption. These reference products make full use of the company's post-filter-feedback, and novel "Zero Common Mode" switching topologies to allow for heatsink free operation while achieving >110 dB-A dynamic range and <0.01% THD+N.

The latest Class-D solutions from Axign target even lower LC output filter losses, lowering the heat in the application and maximizing battery playtime where applicable. To achieve those goals, Axign launched two new devices, now called Axign Sound Processors, optimized for the lowest power consumption levels.

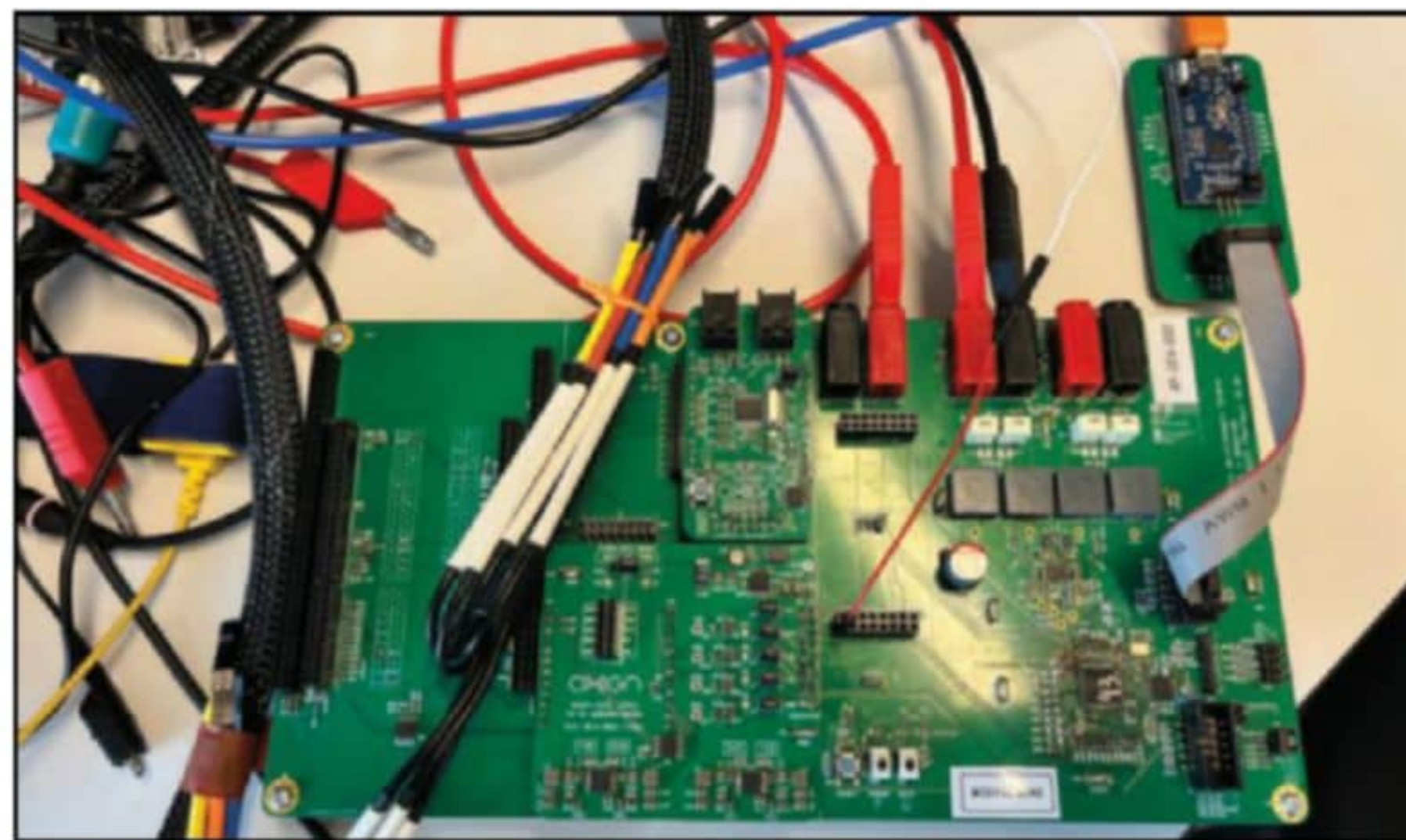
The existing Axign digital audio converters and amplifier controllers (AX5688, two-channel BTL/one-channel PBTL, and AX5689, four-channel BTL/two-channel PBTL) are migrating into Axign Sound Processors (AX5690 and AX5691). According to the Dutch company, these new chips will be able to run the most complex programmable algorithms to better control the dynamics of loudspeakers based on real-time post-filter voltage and current feedback. These new products support a wide operating range without critical external components, to reproduce the best audio quality, while saving energy.

For output power requirements up to 60W in 4Ω BTL per channel, Axign introduced its own two-channel power stages (AX1225 and AX1250), which are further optimized for lowest power consumption levels. These target applications in battery-powered or mains-powered active speakers (Bluetooth speakers, soundbars, and subwoofers) and streaming audio amplifiers.

Targeting Idle Losses

A breakthrough recently confirmed by Axign using its digitally controlled post filter feedback Class-D audio amplifier solutions is the ability to reduce the idle losses in the output filter inductors in full-bridge BTL configurations to almost zero watts. Next to the already proven audio quality and their load impedance independency (from minimum load to open load), the Axign solutions are now even more optimized to deal with inductor losses.

As Richard Langezaal explains, these inductor losses dominate the power losses under normal music listening circumstances causing the temperature to increase due to idle losses, which is particularly serious in compact closed cabinets, small active speakers, smart speakers, and soundbars. "Heat problems occur often when driving very compressed music tracks at maximum volume settings, as the drivers are also inside the same cabinet. The woofer channel(s) in particular will produce a lot of heat.



Axign's new Sound Processor test board

The thermal headroom needs to be managed and a combination of lowest idle losses and low conduction losses is needed," says Langezaal

"To reduce the heat in the inductors in idle, the common mode voltage on the speaker lines needs to be reduced from VDDP/2 down to zero. This is not trivial. Some suppliers tried to reduce the inductor losses by using PWM modulation schemes as VDDP/6 or multi-level PWM switching at the expense of reduced audio quality or more heat at high power. Axign pushes the common mode voltage down to VDDP/50, called ZCM PWM switching. This results in cool inductors in idle, and the lowest power losses at all output power levels, while maintaining Axign's excellent audio quality due to their post filter feedback loop. This is a unique combination of performance parameters, not seen before."

In the newest designs using next generation sound processors, Axign is also adding Class-H power supply tracking, depending on the input signal, to further reduce the losses in compact and battery-powered solutions. The company is offering heatsinkless Class-D audio amplifier solutions from 25W in 4Ω (with low-cost integrated power stages), up to 300W in 4Ω (MOSFET based), or 500W in 2Ω (GaN based).

Axign | www.axign.nl

GaN Systems | www.gansystems.com

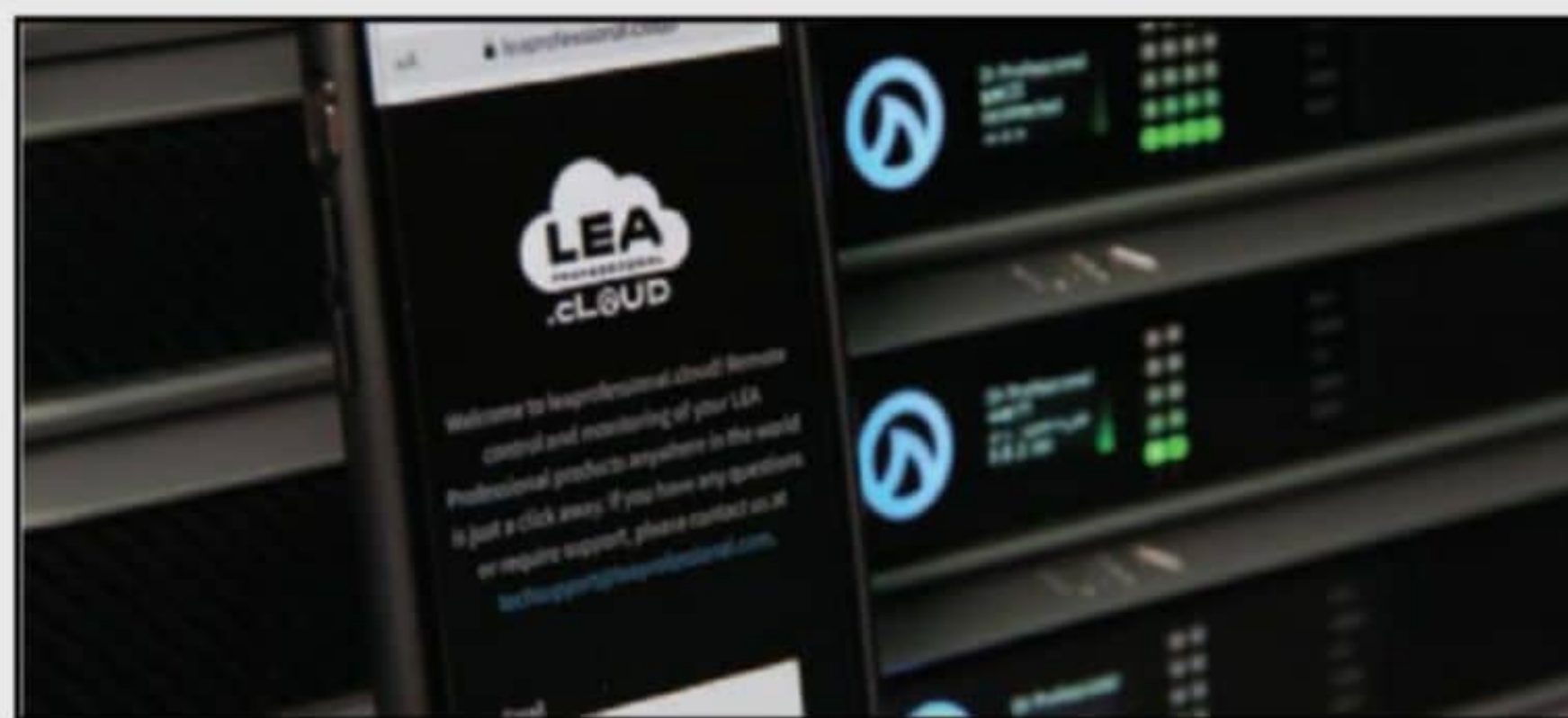
LEA Professional

High Power 1500W per Channel Cloud-Enabled Amplifiers

Underscoring its commitment of listening to pro audio integrators and installers and responding with technologies that meet their needs, LEA Professional, the South Bend, IN, manufacturer of connected pro-audio amplifiers with cloud support, launched its new Connect Series 1504. This new high-power amplifier promises to reset the standard in the segment for large installation deployments.

Entering the market just before the pandemic, LEA Professional gained quick recognition as a manufacturer of cloud-connected amplifiers for professional audio applications, with just the right product at the right moment. The company has been perfecting and updating its products since their debut,

and vastly enhanced the cloud interface experience based on feedback from actual users of the product. And firmware



upgrades continuously improved the features, including adding IEEE 802.1X security protocol.

The new LEA Connect Series 1504 four-channel model drives up to 1,500W per channel (into 2Ω, 4Ω, and 8Ω as well as 70Vrms and 100Vrms) and is available in Network Connect (1504) and Dante Connect Versions (1504D). The 1504 and the 1504D models include advanced DSP with linear phase FIR filters and up to 4 seconds of delay. The Smart Power Bridge feature of these amplifiers provides ultimate flexibility allowing up to 2× 3000W configurations. And with three different methods of network connection, including a built-in Wi-Fi Access Point, users benefit from an existing Wi-Fi network or connect through Ethernet to a Local Area Network.

Unique to any other professional amplifier manufacturer, the new Connect Series models also leverage the built-in Amazon Web Services (AWS) IoT Core. This provides systems integrators with powerful, reliable, and secure remote monitoring and control capabilities across their entire lineup of IoT-enabled amplifiers through the LEA Cloud. The cloud platform has supported integrators not only by allowing them to work remotely, but also helping ensure they are provided a steady, stable income in the form of recurring revenue.

The Connect Series also has access to the LEA Professional Web UI, which allows users to get a clear status check of every amplifier connected to the network and a status of every channel on any given amp to monitor channel performance. In addition, the platform gives users the capacity to view and adjust channel

settings such as input settings, signal generator, crossover, equalizer, limiter, and load monitoring.

Amplifier Management Tool Sharkware

The Indiana-based amplifier manufacturer also recently launched its first application software, Sharkware. Developed in part based on integrator and customer feedback, the free Windows and Mac supported software includes customer-requested features such as offline design, graphical EQ, locked speaker tunings, amplifier grouping, and granular user access controls to all LEA Professional Connect Series amplifiers.

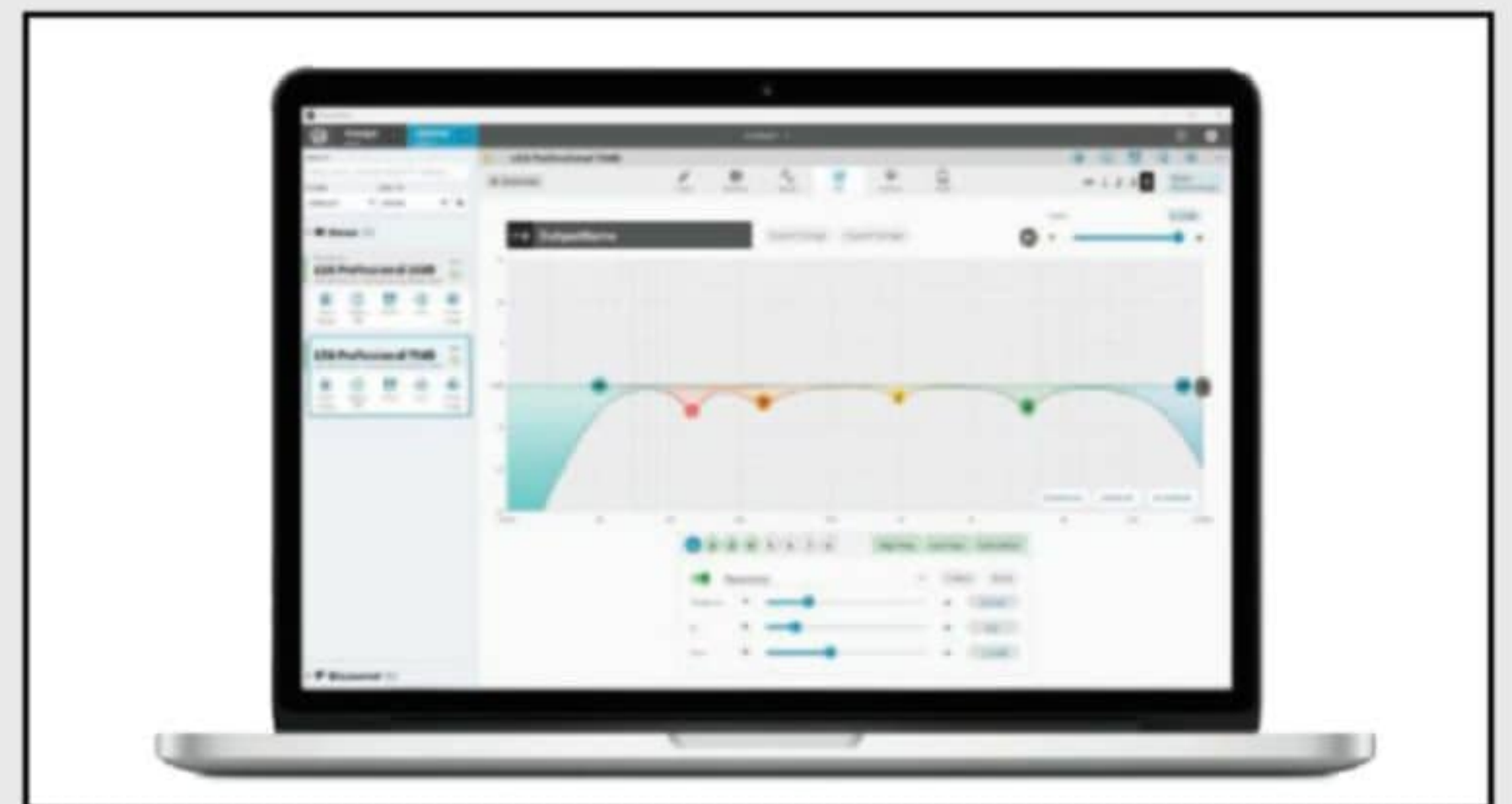
Sharkware allows integrators to work both in offline design and online modes on a system's amplification configuration to ensure a more seamless, accessible, and flexible user experience with all LEA Professional Connect Series amplifiers.

Sharkware joins the company's already successful Web UI and Cloud applications. Sharkware also incorporates key features from LEA Professional's existing Web UI tool, allowing users to monitor the status of every amplifier connected to the network as well as the status and performance of individual channels. In addition, the software gives users the ability to view and adjust channel settings, such as input settings, signal generator, crossover, equalizer, limiter, and load monitoring for all LEA Professional Connect Series amplifiers.

LEA Professional | www.leaprofessional.com



LEA Connect Series 1504 four-channel audio amplifier



Sharkware is a free Windows and Mac software for configuration, control, and monitoring of LEA Professional's Connect amplifiers.

Powersoft

UNICA Next-Generation Cloud-Based Amplifier Platform

UNICA is the new market reference in professional audio amplifiers introduced this year by Powersoft. Several years in the making, answering all the core industry requirements and feedback from users, this new next-generation, advanced audio amplifier platform reinforces Powersoft's leading role in the industry. The cloud-based UNICA platform offers simple integration, straightforward configuration, and low maintenance costs in one easily upgradable, future-proof package.

In Italian, UNICA means both Unique and Single, and this is what Powersoft's UNICA delivers to the audio industry: a unique and single new-generation amplifier platform that is versatile enough to cover a wide range of applications, while at the same time delivering the highest level of audio quality, reliability, and efficiency for which Powersoft is renowned. Most important, UNICA is able to deliver consistency across the entire family range of Powersoft amplifiers.

As the company describes, UNICA is the most advanced amplifier platform in Powersoft's 28-year history, leveraging a new cloud-based technology to provide a safe, consistent user experience for any size of install. With an eye firmly on future requirements, UNICA promises users simple integration, straightforward configuration and low maintenance costs, all in one easily upgradable package.

For now, the UNICA range comprises three four-channel (UNICA 9K4, 12K4, and 16K4, with 9,000W, 12,000W, and 16,000W, respectively) and three eight-channel models (UNICA 2K8, 4K8, and 8K8, with 2,000W, 4,000W, and 8,000W).

Featuring high power density and class-leading efficiency and backed by no less than five new patents and four existing ones, UNICA represents the third generation of Powersoft amplifier platforms.

In conjunction with Universo, Powersoft's new cloud-based platform, UNICA offers cloud-native connectivity and remote monitoring functions without the need for specific product registration or configuration. Cloud integration allows for the storing and sharing of projects online, keeping the amplifier's firmware up to date, or simple integration in third-party systems through APIs.

"Specifying amplifiers from a single product family that can cover a wide range of SPL requirements translates into tidier racks and a new level of user experience. While user-friendly features such as a straightforward third-party communication protocol and simple amplifier replacement with Easy Swap help eliminate any support headaches," explains Claudio Lastrucci, Powersoft's R&D director and co-founder.

As the company also details, UNICA's all-new power supply section takes efficiency to the next level, resulting in less power dispersed through wasted heat. The latest generation of single-stage power factor correction (PFC) and state-of-the-art system-on-chip (SoC) technology processing grant low power consumption and thermal dissipation, while Powersoft's proprietary Smart Rails Management (SRM) allows the supply rails to adapt in real-time to the required output voltage to maximize efficiency and reduce idle losses. An Eco Mode further diminishes consumption when engaged, resulting in cooler racks and drastic reductions in long-time running costs.

The proprietary next-generation architecture also allows high-quality amplification through newly designed input and output



sections, boasting the highest signal-to-noise ratio ever achieved on a Powersoft amplifier. This new technology platform grants faithful reproduction of line-level and digital signals in high-performance installations.

Compared to existing Powersoft products, UNICA also benefits from enhanced Power Sharing developed for the output section, with each output channel capable of delivering up to twice the power when loaded asymmetrically.

It additionally introduces the next generation of Powersoft's industry-leading DSPs, based on the latest system-on-chip technology. The DSP section is operable even when only PoE+ power is available, simplifying deployment and pre-configuration operations.

Easy Maintenance

System integrators typically spend valuable time and considerable effort in minimizing system downtime. Good design practices and service-level agreements are therefore fundamental tools to avoid audio equipment failure. And in the event of a failure, reactivity is paramount: trained technicians need to access the installation, assess the situation and, when remote or onsite repair is not an option, replace the unit. They then need to program the new unit and run a series of tests before the audio system can be operational.

Powersoft solved this major industry issue by developing a novel functionality for hot-swapping amplifiers: Because each UNICA amplifier can automatically archive backups on a flash memory drive situated on the rear panel, users can easily replace a faulty unit with a new one in a matter of minutes, without any configuration or product knowledge.

Powersoft | www.powersoft.com



Powersoft kicked off the 2023 edition of Integrated Systems Europe in style, wowing the first-day crowds with the highly anticipated launch of its third-generation fixed-install amplifier platform, UNICA.



Networking is achieved through three Ethernet ports, allowing both AoIP and control redundancy and splitting control vs. AoIP traffic. Thanks to a firmware-level protocol, audio over IP is managed through both Dante and AES67.



Hypex Nilai500DIY 500W Mono Power Amplifier Kit

A New DIY Benchmark

The highly anticipated Hypex Nilai500DIY 500W mono power amplifier kit is specified to deliver 500W with 1% THD at 4Ω, improves output noise and idle losses significantly, and comes with a discrete buffer stage and voltage regulators, just like the venerable Hypex NC400 module it will be replacing. *audioXpress* offers an extensive review with measurements of the full kit, including the PS500 DIY power supply and DIY preamplifier kit.

By
Stuart Yaniger

I recently had the pleasure of working with new power amplification and power supply modules from Hypex, a Dutch company, which has been at the forefront of the development of Class-D amplification. The NCore amplification is a self-oscillating topology with high power, high performance, and relatively low load dependence. This is achieved via extremely high negative feedback, stabilized by a manipulation of the loop gain. The NCx500 modules I had on my workbench were intended for designers who want to use them as components in products, integrating them into stand-alone amplifiers or as part of active loudspeakers.

Although there's no lack of integrators out there, a lot of people still want greater participation in the process but are intimidated by the engineering requirements or just don't have the time to do a design even with prefabricated modules. There's also the question of integrator competence—many of the ones I'm familiar with are extremely good, but there's always a risk. And a lot of details can have a profound impact on the performance of the finished product, not just things like distortion and noise, but oddball turn on and turn off thumps and unwanted sound effects. Wouldn't it be nice to have an integrated, ready-to-use version of ultra-high-performance amplification that is reliable and lets you get your hands into the works?

Enter the new Nilai DIY Class-D power amplifiers and preamplifiers from Hypex. When I questioned Hypex's Frank Veldman about the differences between the Nilai amps and the new NCx500, he responded, "Whereas NCOREx is considered more of an update to NCORE, using the same concepts, Nilai

does go further creating a higher-order feedback loop and thereby increasing loop gain within the audio band further."

Veldman elaborates:

"The aim with Hypex amplifiers has always been to have excellent performance across the entire audio band, for UcD and NCORE this meant distortion levels were practically frequency independent. For the development of NCOREx and Nilai this has been reconsidered and while extensive (listening) tests obviously showed the importance of proper high frequency performance, it did not necessarily lead to full frequency independence. For new developments this meant high frequency performance has been as important as ever in Hypex designs, the philosophy of what is happening below that changed to some extent.

"These new insights can be seen in the performance of both NCOREx and Nilai, the main difference here is that the Nilai adds another 15dB of loop gain, resulting in a total of >75dB across the entire audio band. In the current Nilai500 DIY module this increase in loop gain has not only been used to improve performance, but like the NCx500, part of it has been used to offset the increase in open loop distortion of the power stage originating from the desire to further improve its efficiency. In the current Nilai amplifier (and this will be the same for any future implementations) we have sacrificed performance for improved efficiency on the power stage end of the amplifier, this mostly comes down to increasing deadtime. The extra loop gain we have with the Nilai is in part used to counter the negative effects this has on distortion, therefore

the actual distortion measurements, while I think they look excellent, don't quite resemble the huge improvements there are over, for instance, NCORE.

"Because of the choice for a fully complete discrete amplifier with input stages and such all running Class A, total efficiency of the module is not exceptional. When purely looking at the power stage of the amplifier, we are looking at about halving the idle losses compared to the known NCORE amplifiers.

"A second, smaller benefit in the Nilai is the improvement of its frequency response which is now basically flat in the audio band. This is combined with a second-order roll-off, which makes it less susceptible to HF noise. NCORE is usually down about 0.5dB at 20kHz, the NCx500 has slightly increased bandwidth and thereby at about 0.25dB, both having a first-order roll-off. Nilai being a second-order roll-off is indeed easily flat to 20kHz and actually a bit beyond that."

The technology used for the preamplifier is much more conventional, comprising discrete op-amps and the same regulators (at least visually!) as used in the input stages of NCx500 amps. The "Class D" on the front panel is branding rather than a description of the circuit (although the power supply is switch mode).

The Products

For those old enough to remember the old Heathkit, Eico, and Knight Kit products, the Nilais will seem a bit odd—the boards containing the signal and control circuitry are preassembled. This is a good thing in 2023 since surface mount parts are difficult for a lot of people (like me!) to handle. And the high frequencies involved in switching circuits make small details of lead dress and grounding extremely critical. So the Nilai kits are more reminiscent of the old Dynakit, which were among the most popular audio products of the 1950s and 1960s and came with pre-populated circuit boards. But unlike the Dynakits, the interconnections to the power supply and I/O are all done with plug connectors—no soldering needed.

I call the images shown in **Photo 1** and **Photo 2** "Christmas in February." The power amps and preamp parts arrived in attractive boxes just begging for an unboxing video—my YouTube presence being minimal, I'll leave that to someone else. The assembly instructions are done graphically, giving this an Ikea-like international feel. The diagrams are clear and easy to follow.

The heart of the DIY Class-D power amplifier unit is the Nilai 500W amplifier module (**Photo 3**), which is neatly laid out and constructed and is



Photo 1: The external packaging of the kits is graphically beautiful and promising. These are available to order not in the regular Hypex Electronics website but through www.diyclassd.com.



Photo 2: The kits include all parts, cables, preassembled modules, and the basic tools needed for assembly.

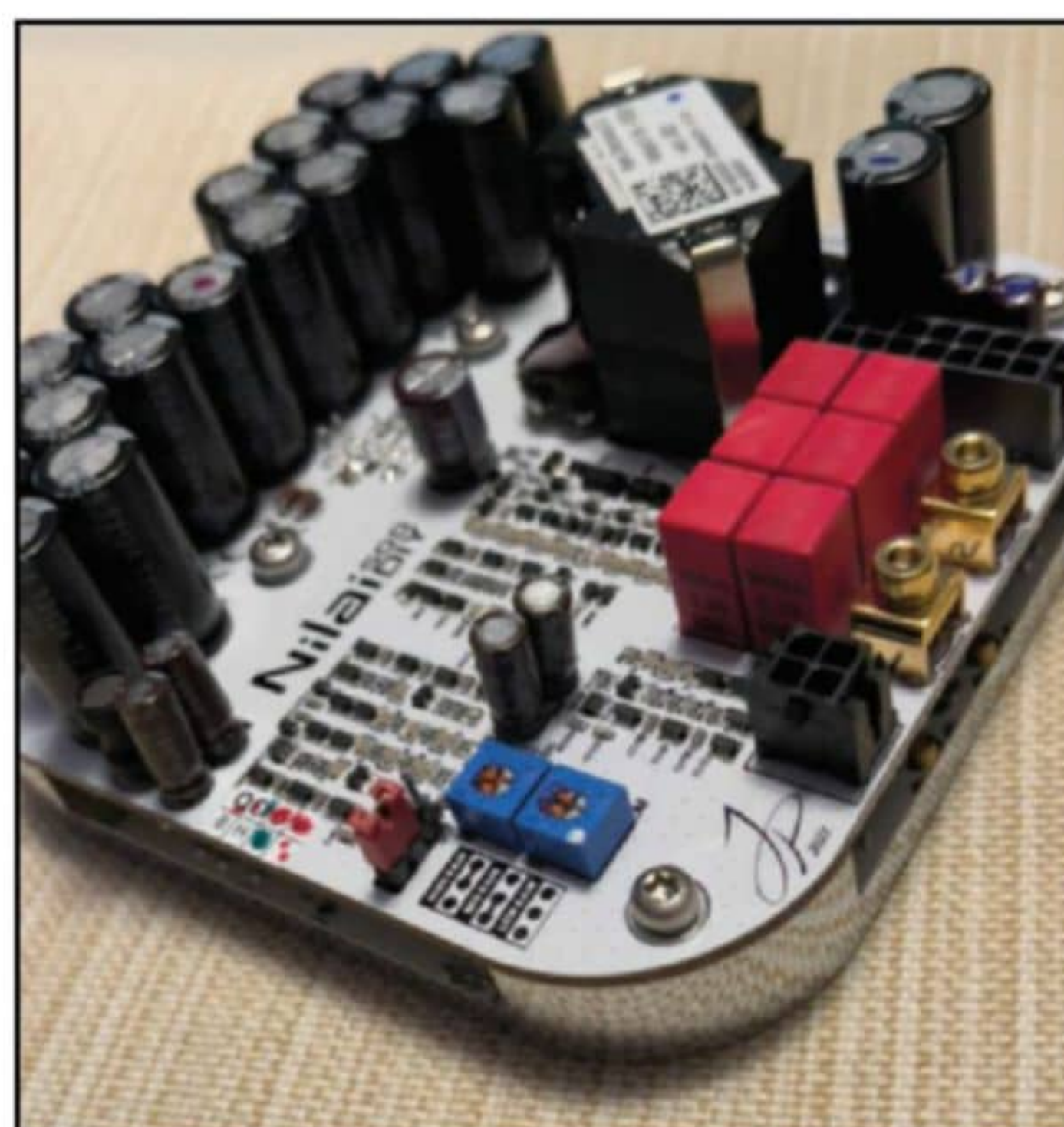


Photo 3: The Nilai power amplifier modules are mounted on a beautiful chromed-steel heat spreader.



Photo 4: When assembled, the Nilai power amps and preamps are compact, elegant, and understated in appearance.

mounted on a large steel heat spreader, similar to the one used for the Hypex NCx500 power amplifier module. Perhaps because this is a consumer-facing module, the Nilai's heat spreader is polished to a beautiful mirror finish. Some of the modules have a protective plastic film to protect the finish, so make sure this is removed before mounting the module.

On the rear panel, the power amps have balanced inputs exclusively and a switch to control the brightness of the front panel LED. They are finished in an attractive and unobtrusive black anodization and the case comprised two extruded side panels and plates for the front, rear, top, and bottom holding everything together. Hole placement was accurate making the assembly go very smoothly without forcing any of the screws.

Construction of the power amplifiers took me about an hour-and-a-half for the first unit, then

slightly over an hour for the second. There are a few places where the work is a bit tight for the ham-handed, so if you're of a certain age (ahem) some magnifying close-up glasses can be of help. I'll admit that for the second amplifier, I left off one of the side pieces for assembly to allow me easier access to some of the board connectors, saving that side for last. This also made sliding the top plate in a bit easier.

A few parts are also mechanically tight including the front panel, which is mounted with a pair of blind screws. No doubt this contributed to the unit's solid feel and lack of buzzing and rattling when driven hard. One other thing to note is that there are two separate circuit boards linked together that need to be broken apart—the instructions do not mention this.

The preamplifier took slightly longer to put together because of more I/O, fancier casework, and a front panel display, but the assembly was as simple and straightforward as the power amplifiers. One oddity was that on the back panel, one of the screw holes wasn't countersunk, so I only have 15 screws instead of 16 holding the I/O board on the back panel.

The Nilai preamp has two single-ended and one balanced input. At present, there's no optical, USB, or coaxial digital inputs, but there is an expansion area that might accommodate these inputs in the future. There are two balanced outputs and both conventional (1/4" TRS) and "balanced" headphone outputs (**Photo 4**)—I use the scare quotes here because for whatever reason, the headphone world has co-opted the term "balanced" to mean "differential."

The preamp includes a remote control, which is greatly appreciated (**Photo 5**). Now, I'll confess that I prefer apps to remotes because of the clutter factor, but I recognize that I'm in the minority. At least this remote is relatively small and quite nicely laid out.

Volume control, balance, and source routing are done via rotary optical encoders rather than conventional potentiometers. This is a much cleaner way to do things and generally a plus for reliability. The controls allow easy scrolling though and item selection in the various option menus.

Once the products are assembled, you have some moderately small, visually attractive (in a minimalist way), and very solid and expensive feeling monoblock power amps, which can be stacked or hidden away, depending on your taste in decor for your listening room.

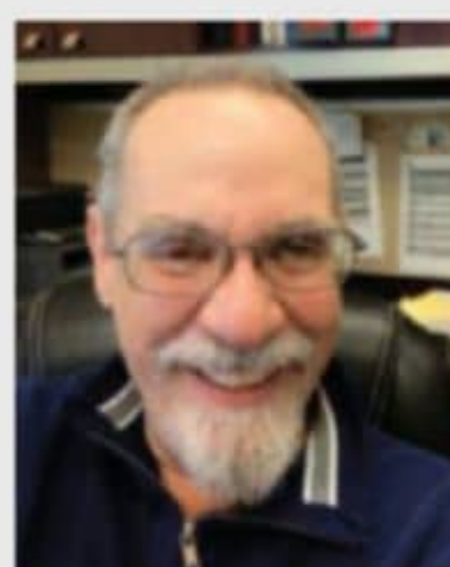
The preamplifier, of course, will be out and proud so that the controls and display can be accessed.

Photo 5: The preamp includes a remote control.



About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently runs a technology consulting agency in western New York. 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.



What screams “2023” is that the preamp is considerably larger than the power amps.

The Nilais In Use

Initially, I just used the monoblock power amplifiers driven by the preamp outputs of an NAD M10 integrated amplifier to run the top section of my biamped speakers, a reworked version of the NHT 3.3 four-way dynamics. The bottom section was powered with a pair of Hypex NCx500 modules that I conveniently have not gotten around to returning (soon, guys, soon, I promise). I swapped them at the end just in case there were any audible differences.

My first reaction was panic—the amps wouldn’t power up! Despite the simple assembly and my experience with building electronics from scratch, did I make a silly error or somehow misinterpret the instructions? I frantically opened the amps up to trace the wiring (and frantic enough to drop one of them on my foot while trying to pry off the tightly fitting front panel—the resulting broken toe is a small sacrifice for science).

Everything seemed in order; after a bit of head scratching, I realized that, unlike other amplifiers I’ve used, the amps would not power up without a voltage present there. Although the trigger is nominally 12V, I didn’t have such a source close at hand, so I grabbed a 9V battery and a couple of TRS cables and crossed my fingers. Success! Of course, if you’re using these as part of a system with the Nilai preamplifier, the trigger voltage is provided

and this kludge is unnecessary, although the user needs to run a 3.5mm TRS cable between the units.

I found out later that there is a way to power the amps on without the trigger. On the front panel, there is a small, unmarked button near the pilot light cleverly camouflaged to match the panel cover. It is, at least for my eyes and in less than glaring light, pretty difficult to see. Pressing that button allowed me to power the amp on and off without the trigger. I’d like to see this either in a contrasting color or with some sort of marking or indication.

My second problem was volume. Once the gain between top and bottom halves of my biamped system was set, I just couldn’t get my system to play loud enough. Clearly the Nilai amps required more voltage than the NAD’s preamp outputs were supplying. In a flash of inspiration, I did something uncharacteristic and actually read the manual. Aha! There are three jumper-selectable gain settings (11.8dB, 21.8dB, and 27.8dB) on the amplifier board, and the amp is shipped with the jumpers set for mid gain. One more disassembly, jumpers moved, and all was right with the world.

The Nilai amps were, unsurprisingly, absolutely dead silent, both electrically and mechanically. Pressing my ears right up to the speakers, I could not tell if the amps were on or off. I didn’t actually try to drive them into clipping, but they certainly never showed any signs of distress when cranked up with my favorite loud heavy metal.

More delightfully, after a full day of playing at a robust level, the amps were barely warm. In fact, after a day of use playing



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relatively quiet acoustic music, the amps were just about as warm. The efficiency cautions that Frank Veldman had given me made sense—there was far less difference in dissipation between the amps at idle versus going full tilt than is the usual, even for Class D.

I will confess that my use of the preamplifier was a bit more cursory. My primary music source is streaming, usually through the NAD M10, where the big attraction is Dirac Live. My alternative streaming source is a WiiM Pro, but that really functions best using its digital outputs, and at the point, optical or coaxial digital inputs are not available on the Nilai preamp (though the accessory plate does tease the future availability of this option). And of course, there's still no Dirac, and I must admit that I've become addicted to the sonic magic that software brings.

So as a compromise, I inserted the Nilai preamp, set for 0dB gain, between the NAD and the Nilai power amps. The best thing a preamp can possibly do is nothing at all, other than switching sources and raising and lowering the volume. And that's exactly what the Nilai did—nothing. No detectable tonal changes, distortion, or noise. Volume, balance, and source switching were noiseless, with no clicks, thumps, or scratches. There is absolutely nothing else one could ask of a preamp.

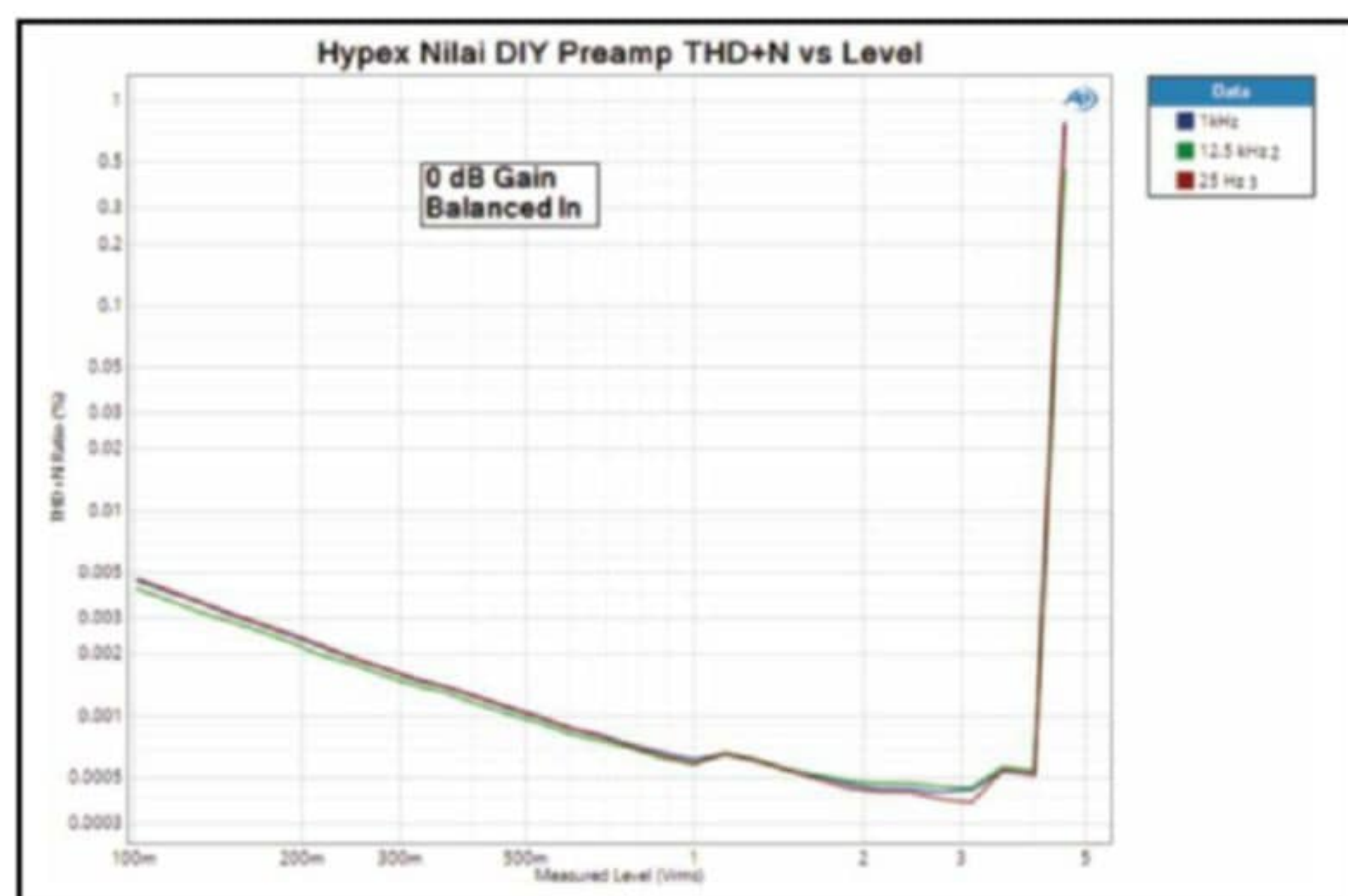


Figure 1: The Nilai preamplifier can output a maximum of 4V at all test frequencies.

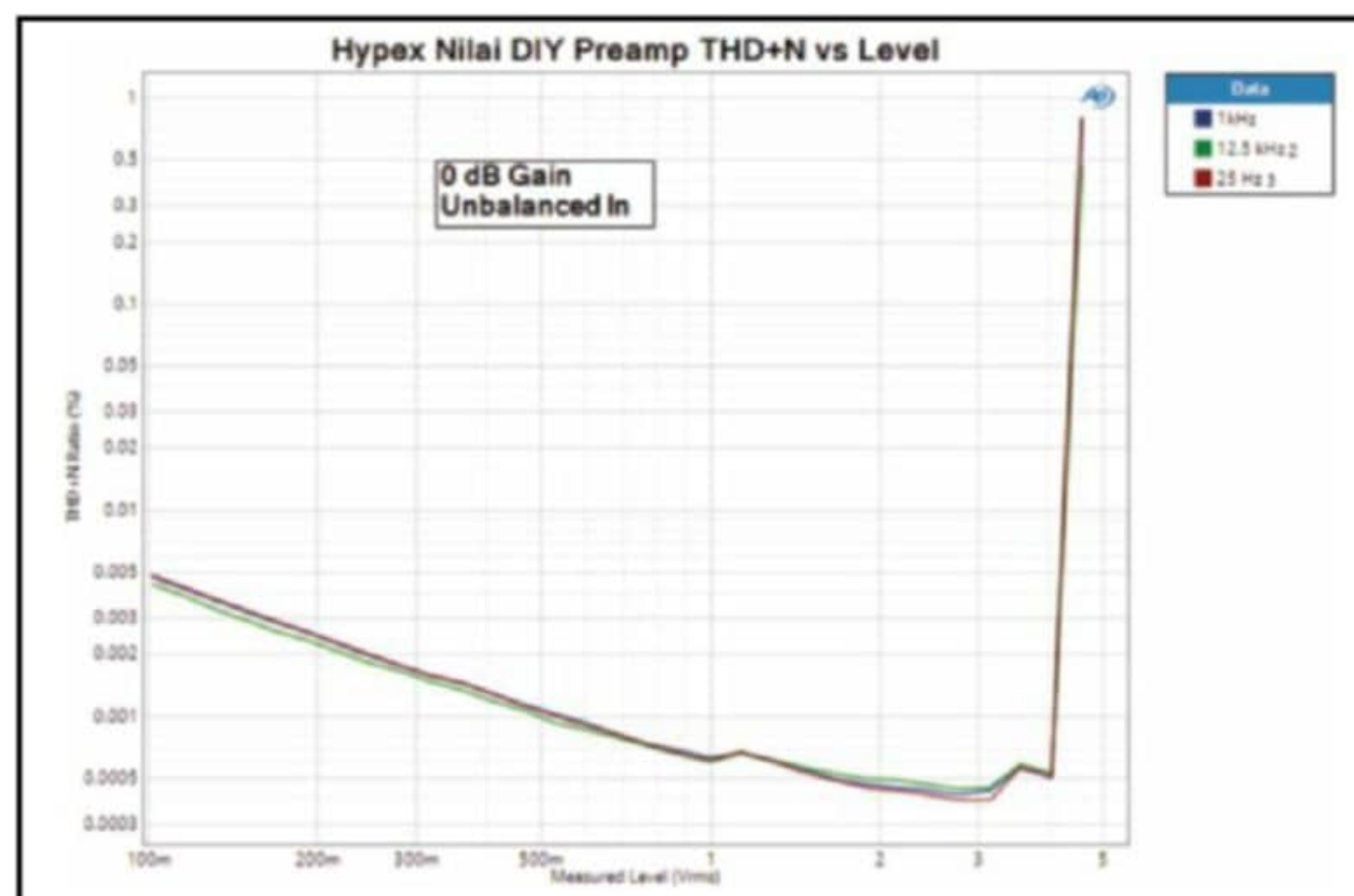


Figure 2: The unbalanced input gives the same high performance seen with the balanced input.

On the Bench

I pulled the Nilais out of my system (with some regret!) and set them up on my test bench next to an Audio Precision APx525 analyzer and a set of homemade high power dummy loads.

First up was the preamp. Most of the measurements were, frankly, boring. How so? I was so often just looking at the baseline noise and distortion of the audio analyzer. My default for measurement was unity (0dB) gain, but I confirmed that the preamp had a maximum gain of 13.1dB (about 4.5V/V).

Figure 1 and **Figure 2** show THD+N versus output level for three different frequencies (25Hz, 1kHz, and 12.5kHz) for the balanced and unbalanced inputs, respectively. The curves overlay one another nearly perfectly and are noise limited right up to the clipping point at slightly over 4V. This consistency is very unusual in feedback designs, where the distortion starts deteriorating at high frequencies as the open loop gain, and hence feedback, decreases. And of course, the distortion is down at my measurement residual, so I may be reporting ridiculously small numbers that are still too large. The 4V limit is above what most power amps require to reach full output. This is very impressive engineering.

Not surprisingly, when we plot THD+N versus frequency for a 2.5V output level for the balanced and unbalanced inputs (**Figure 3**

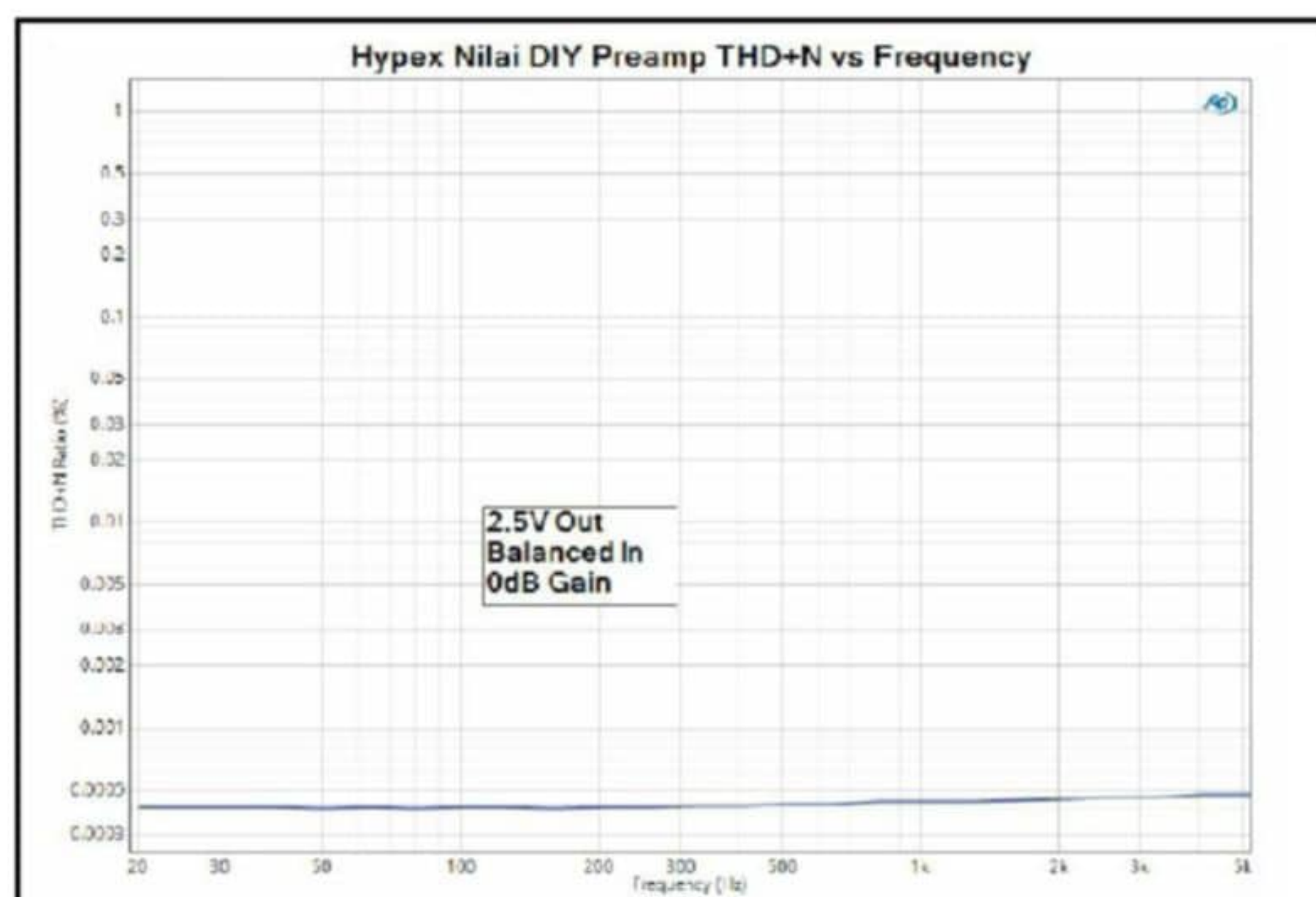


Figure 3: At 2.5V output, the distortion is extremely low and consistent across the audio band.

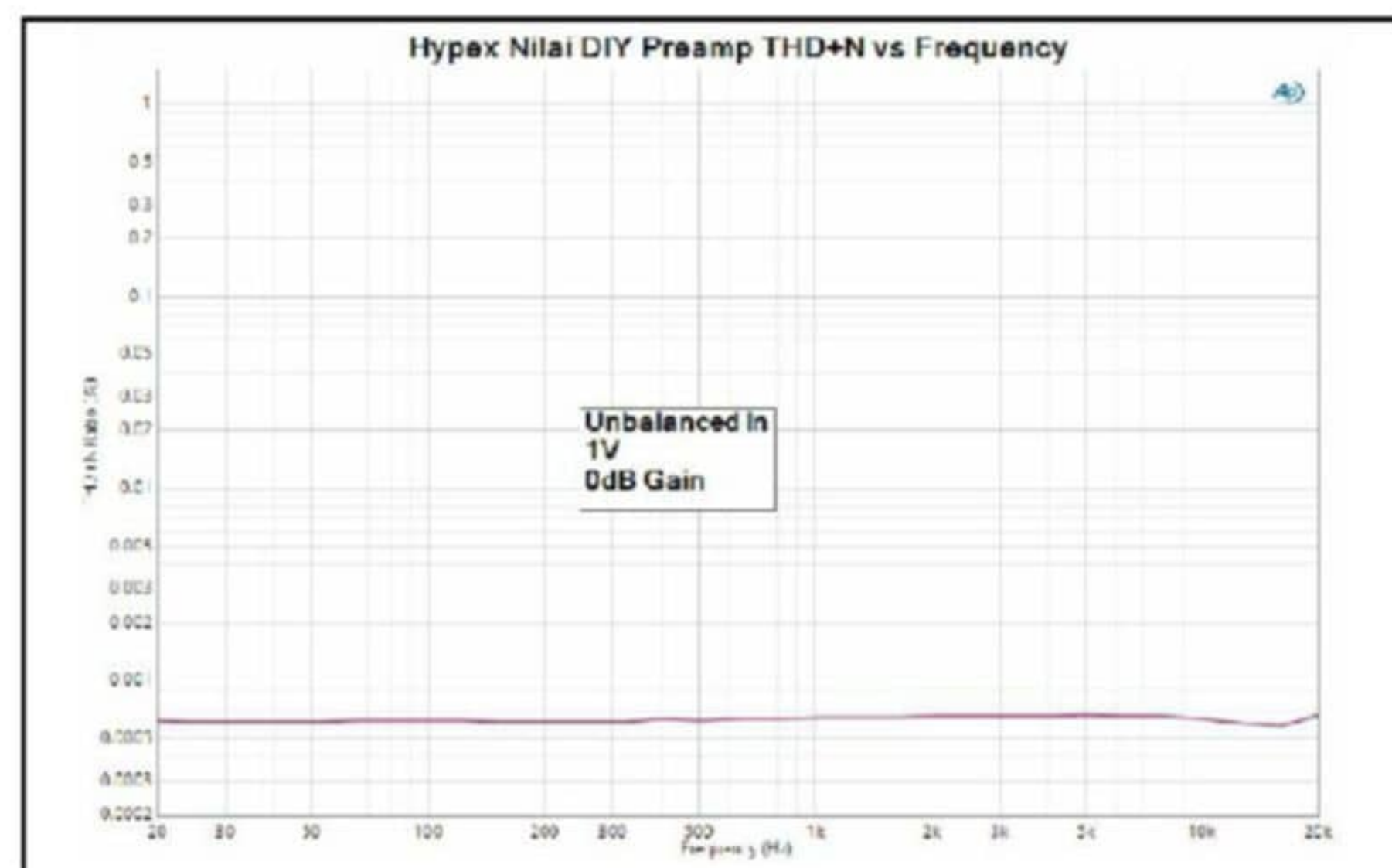


Figure 4: Once again, the unbalanced input gives the same high performance as the balanced input.

and **Figure 4**, respectively), the curves are at a very low level and basically flat.

One of the compelling reasons (if not THE compelling reason) to used balanced connections is to reject common mode noise. The

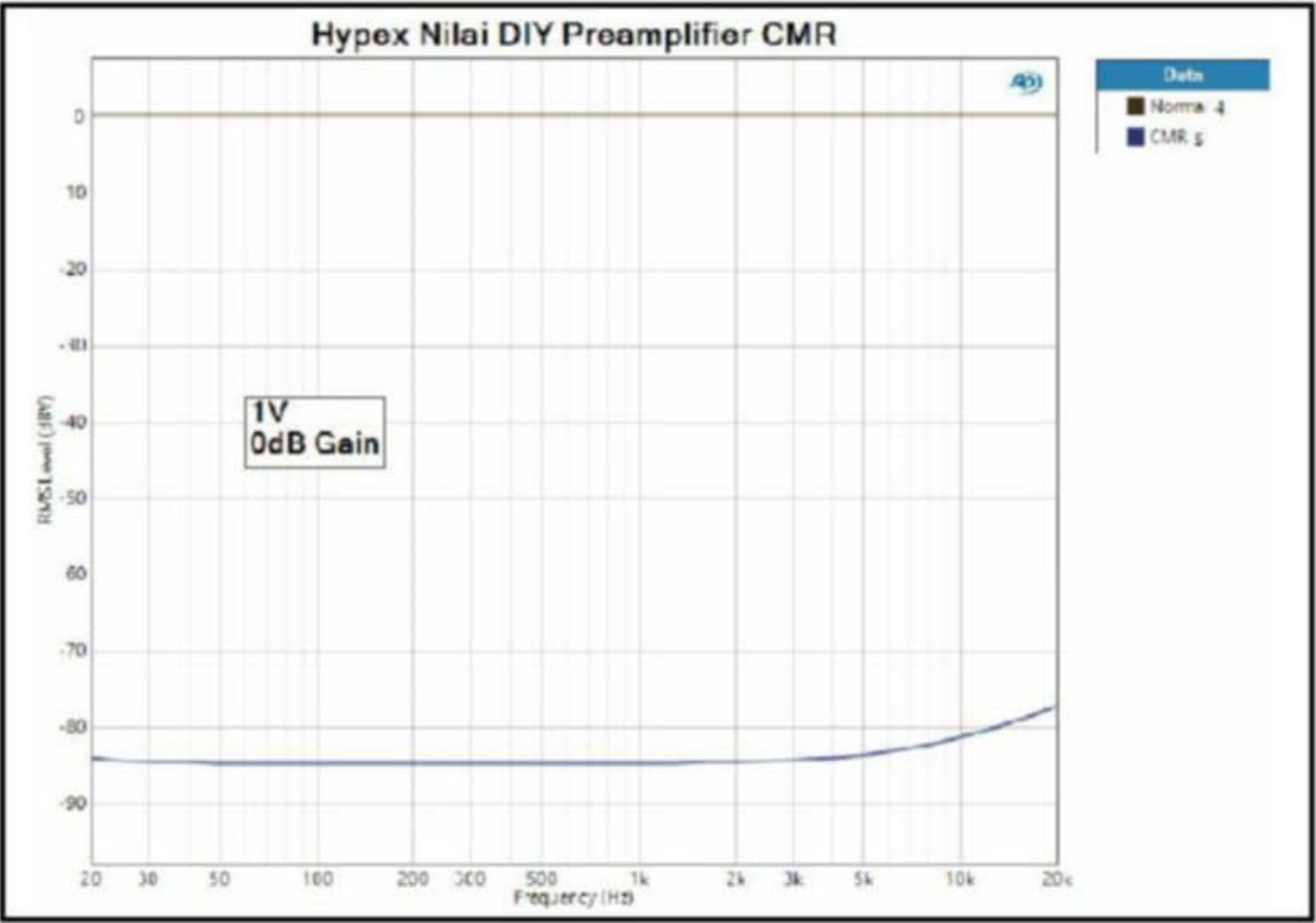


Figure 5: The Nilai preamp’s input has very high common more rejection across the audio band.

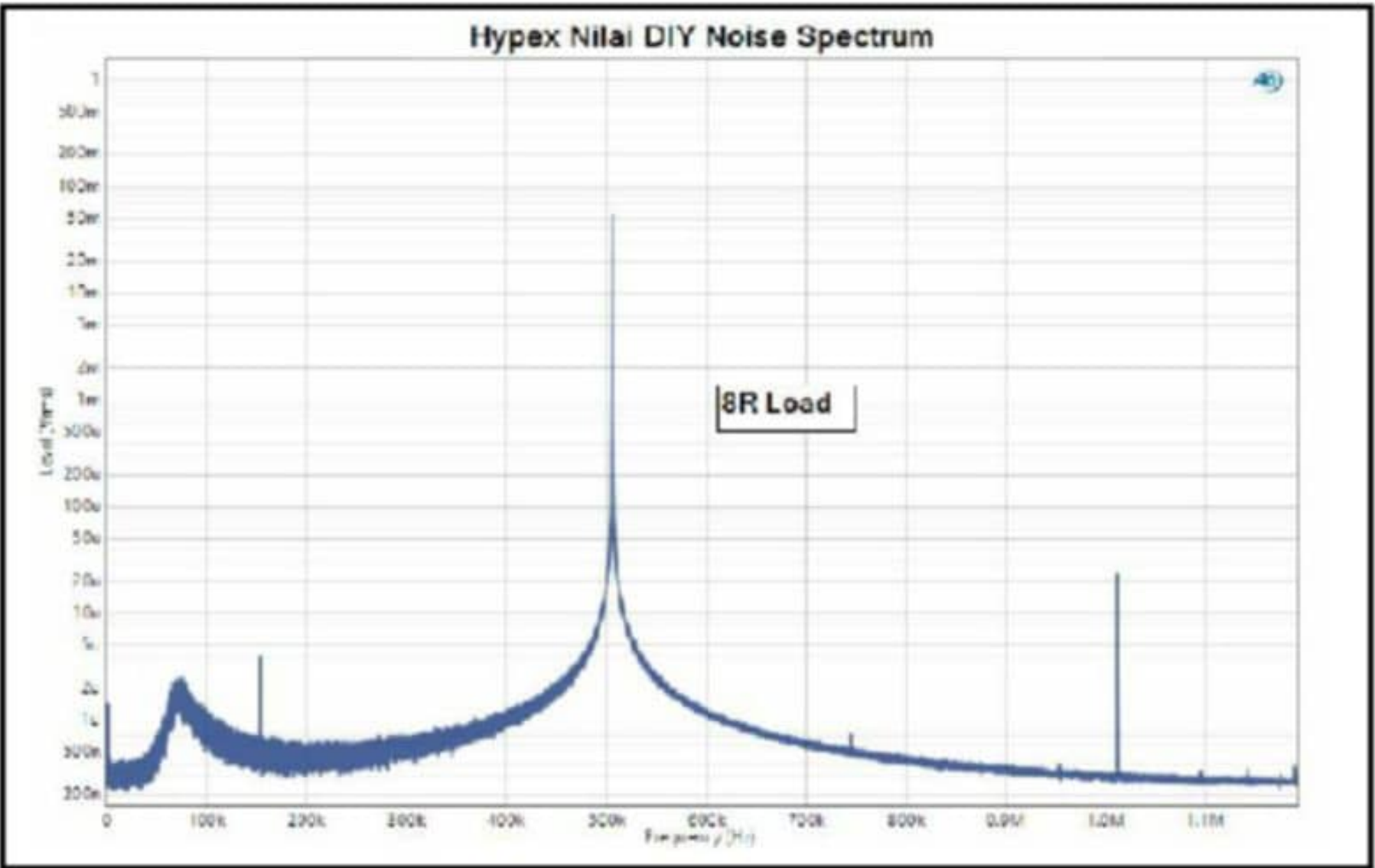


Figure 6: The Nilai power amplifier’s output noise shows the lowest level of switching leakage I have seen in any self-oscillating Class D amplifier.

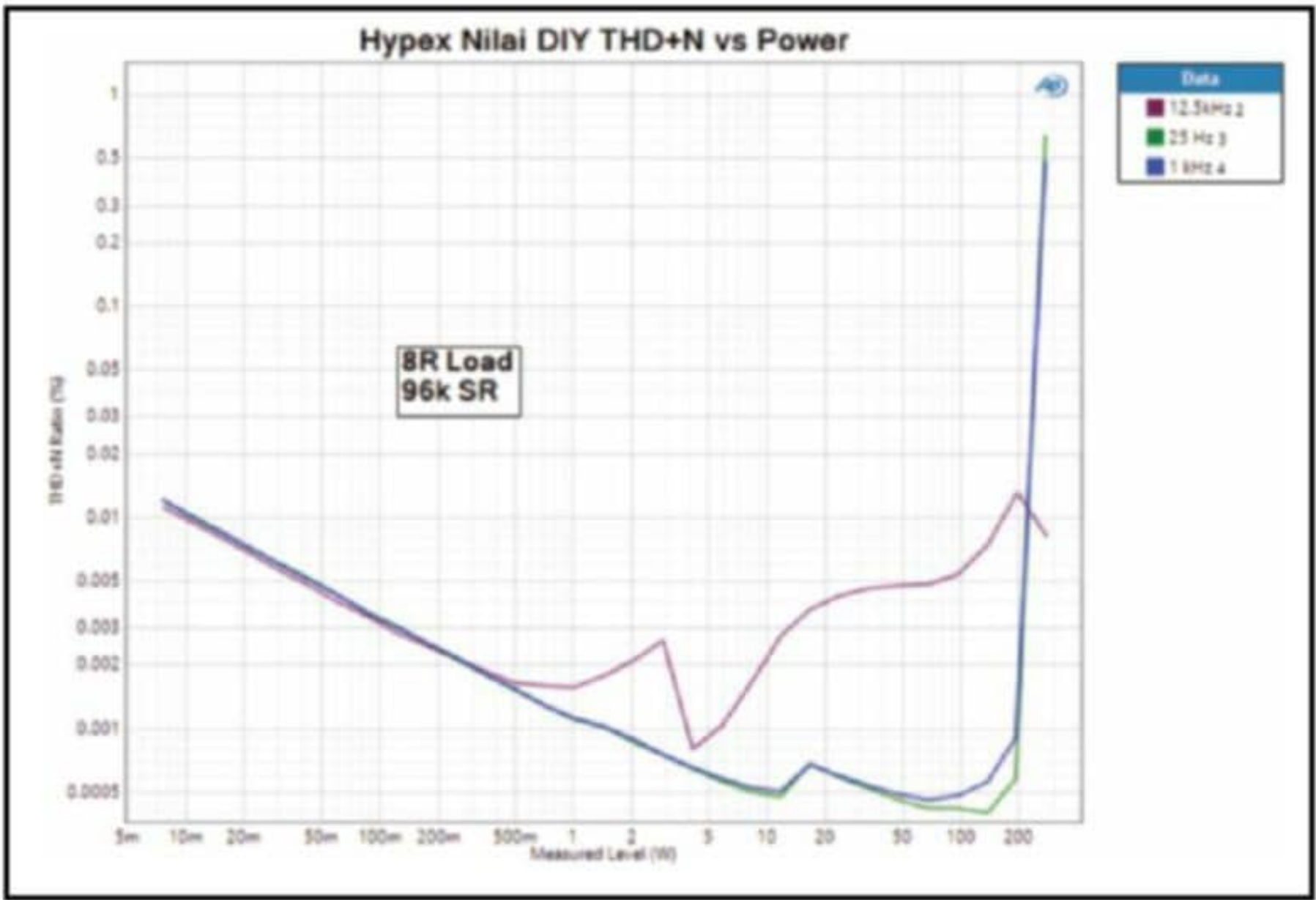


Figure 7: At 8Ω load, the Nilai power amp puts out over 280W at the 1% distortion point at all test frequencies.

comparative output at 1V drive with gain set at 0dB with normal (differential) mode and common mode is shown in **Figure 5**. The common mode rejection is about 85dB throughout most of the audio band and falling slightly to 77dB at 20kHz. This is very good performance, bespeaking an input topology resembling a high-quality instrumentation amplifier.

This preamplifier achieves the goal of doing essentially nothing to the input signal other than routing and amplifying.

Turning to the power amplifier, my measurements were all made with the gain set to the factory-supplied default jumper setting, which yielded a gain of 21.7dB.

I first hooked up an 8R dummy load and measured the noise spectrum at no signal in. This is shown in **Figure 6**. The effect of having a second order output filter is clear—to date, all of the self-oscillating Class-D amps I have tested have a residual of several hundred millivolts of the oscillation frequency on the output. As you can see, the oscillator leakage is only 50mV and the overall noise floor is exceptionally low, correlating with my sonic observations.

The main job of a power amplifier is to deliver power, so I ran distortion versus power curves with both 8R and 4R dummy loads (**Figure 7** and **Figure 8**, respectively). Each plot shows data taken

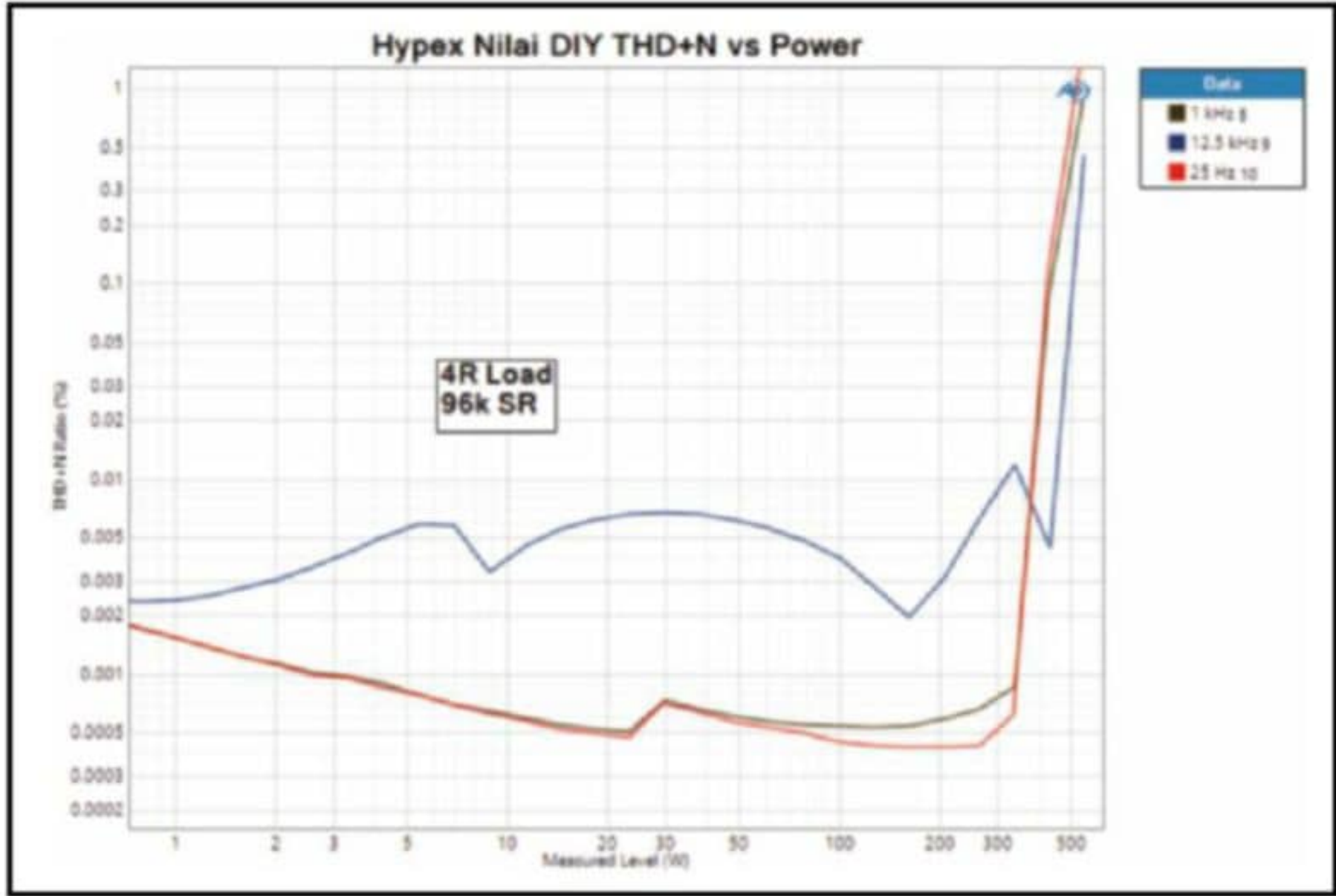


Figure 8: With a 4R load, the Nilai puts out over 500W at the 1% distortion point at all test frequencies.

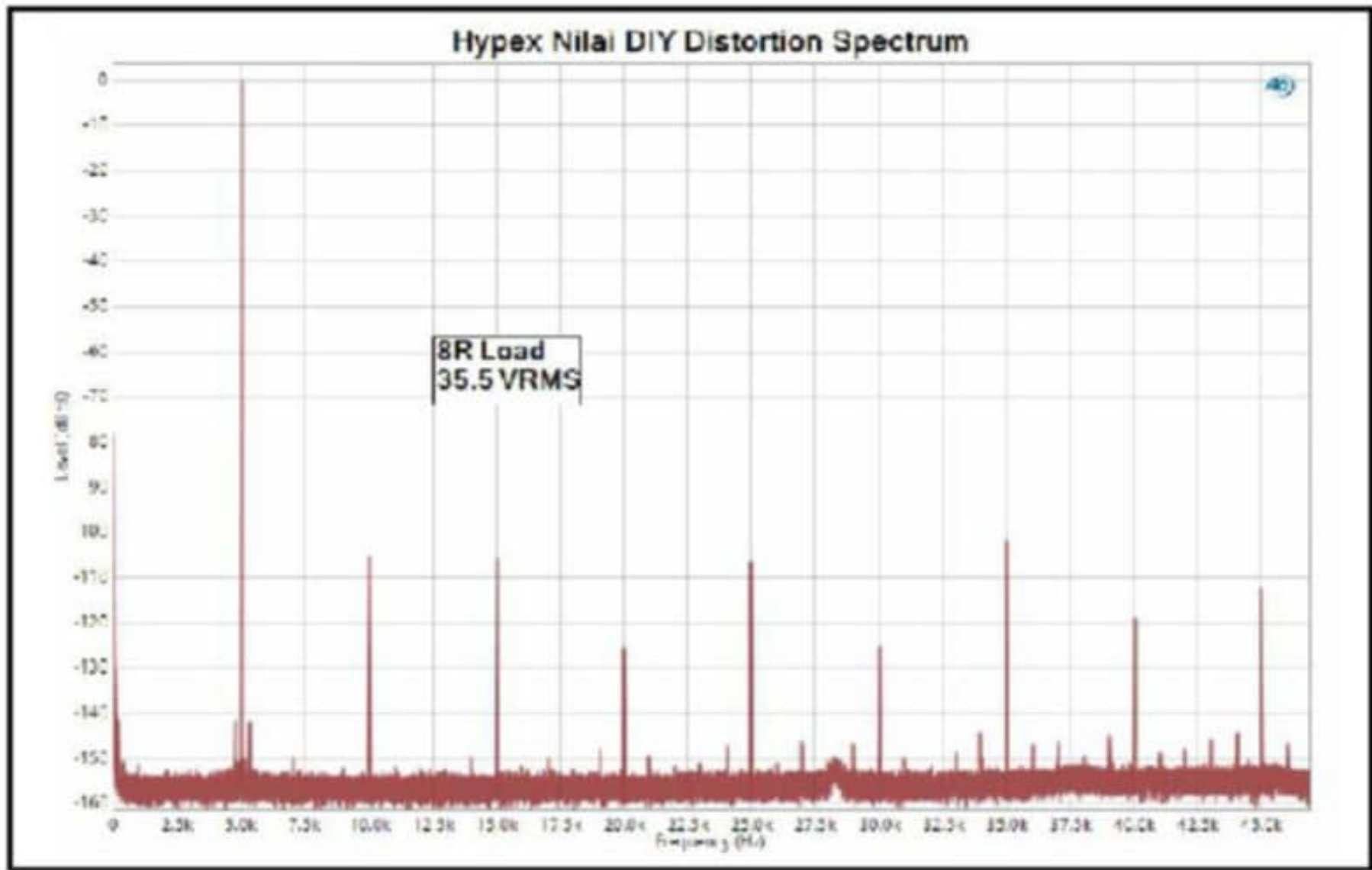


Figure 9: At 5kHz and 100W output, the distortion spectrum is remarkably clean.

at 25Hz, 1kHz, and 12.5kHz. For the 4R load, the power at 1% distortion exceeds 500W; the corresponding data for 8R shows the 1% distortion at about 280W. The 25Hz and 1kHz data track one another reasonably well. The 12.5kHz curves show some odd behavior at medium power with higher distortion. To put that in perspective, the elevated distortion is still only 0.005% or so, what would have been state of the art a couple years ago. And of course, the harmonics of those frequencies lie well outside of human audible limits, so this is of minimal concern.

Figure 9 shows the distortion spectrum at 5kHz with an 8R load at 100W. The low noise means that the distortion spikes are easy to resolve—but look at the levels! It is a challenge to measure distortion this low. Sweeping distortion versus frequency yields the curves shown in **Figure 10**, which were taken using an 8R load with 2.83V (1W) and 29.1V (about 106W) drive. There is some rise at higher frequencies (with the 48kHz sample rate used, harmonics above 24kHz are suppressed), but again, the levels are crazy low.

There's been some rumbling on the Internet that Hypex amplifiers do not react well to capacitive loads. This struck me

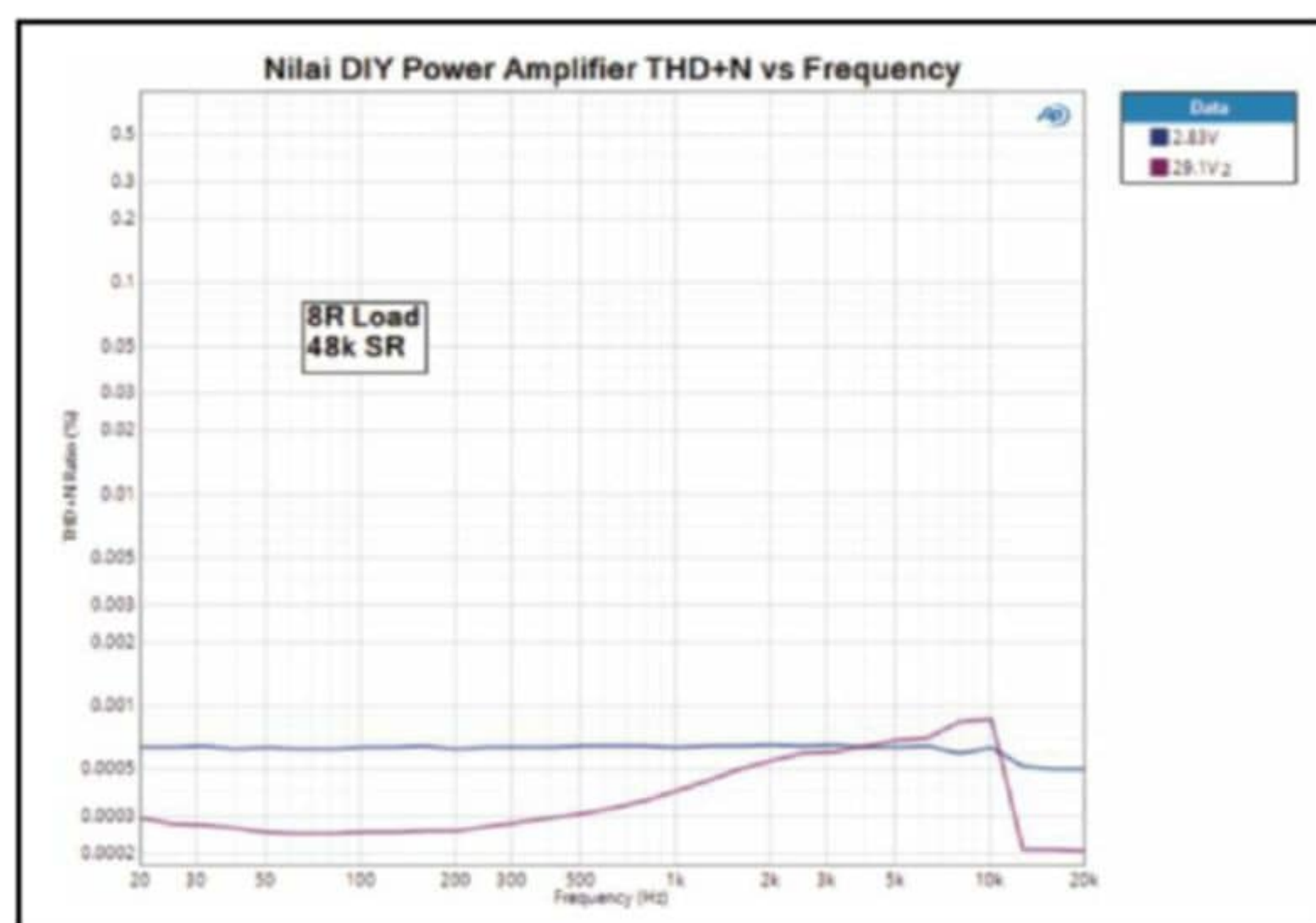


Figure 10: THD+N sweeps at low and high power show consistently low distortion.

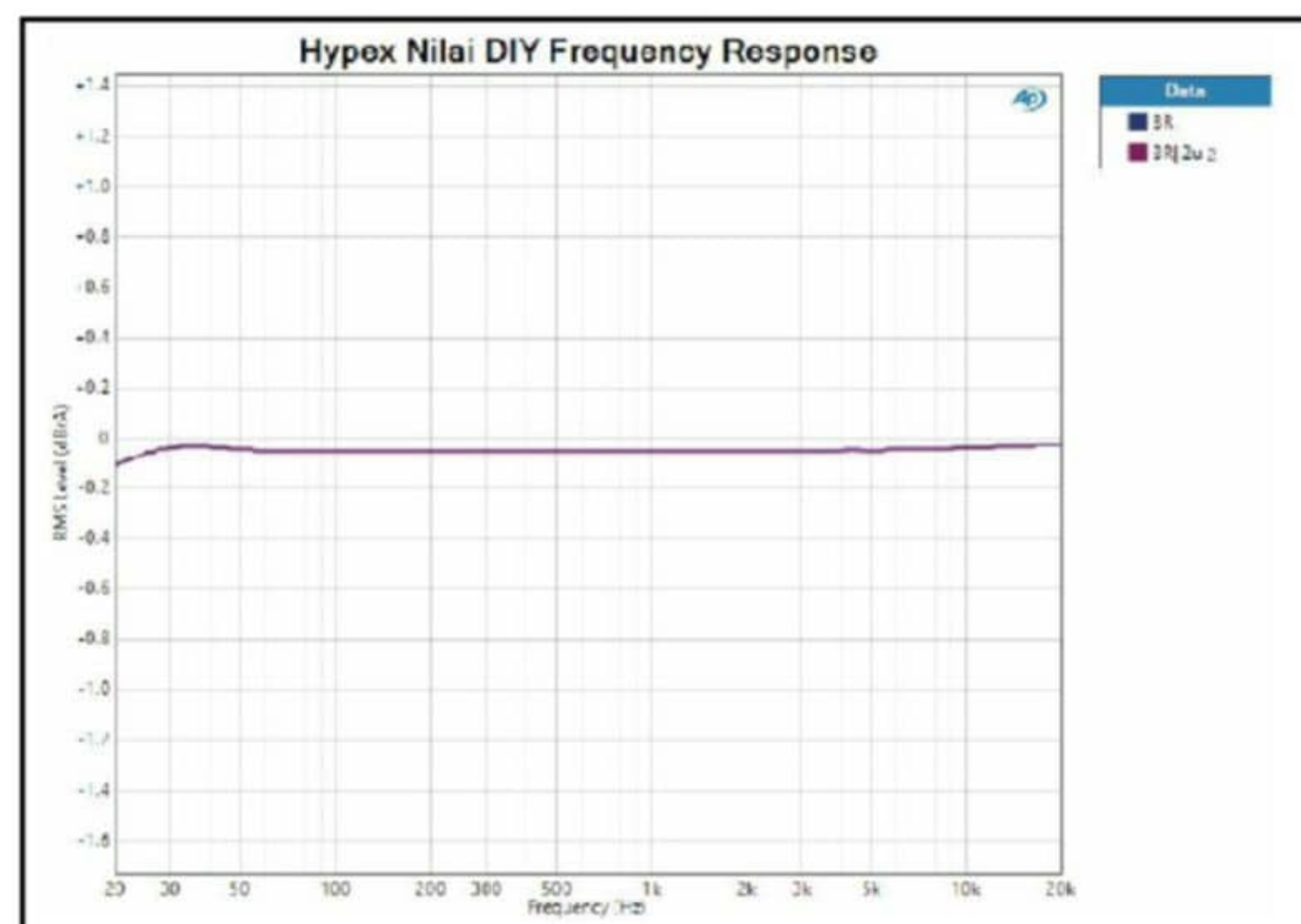


Figure 11: Placing an unrealistically high capacitance across the load barely changes the frequency response.

as quite curious. To explore these claims, I first ran a frequency response sweep with an 8R load, then for comparison, the same load with a 2μF capacitor shunting it, which is a truly horrific load, well exceeding the capacitance presented by a typical electrostatic loudspeaker. The results are shown in **Figure 11**.

If there's any problem there, I sure can't find it. Just to be certain, I ran a power sweep using a dummy load of 4R in parallel

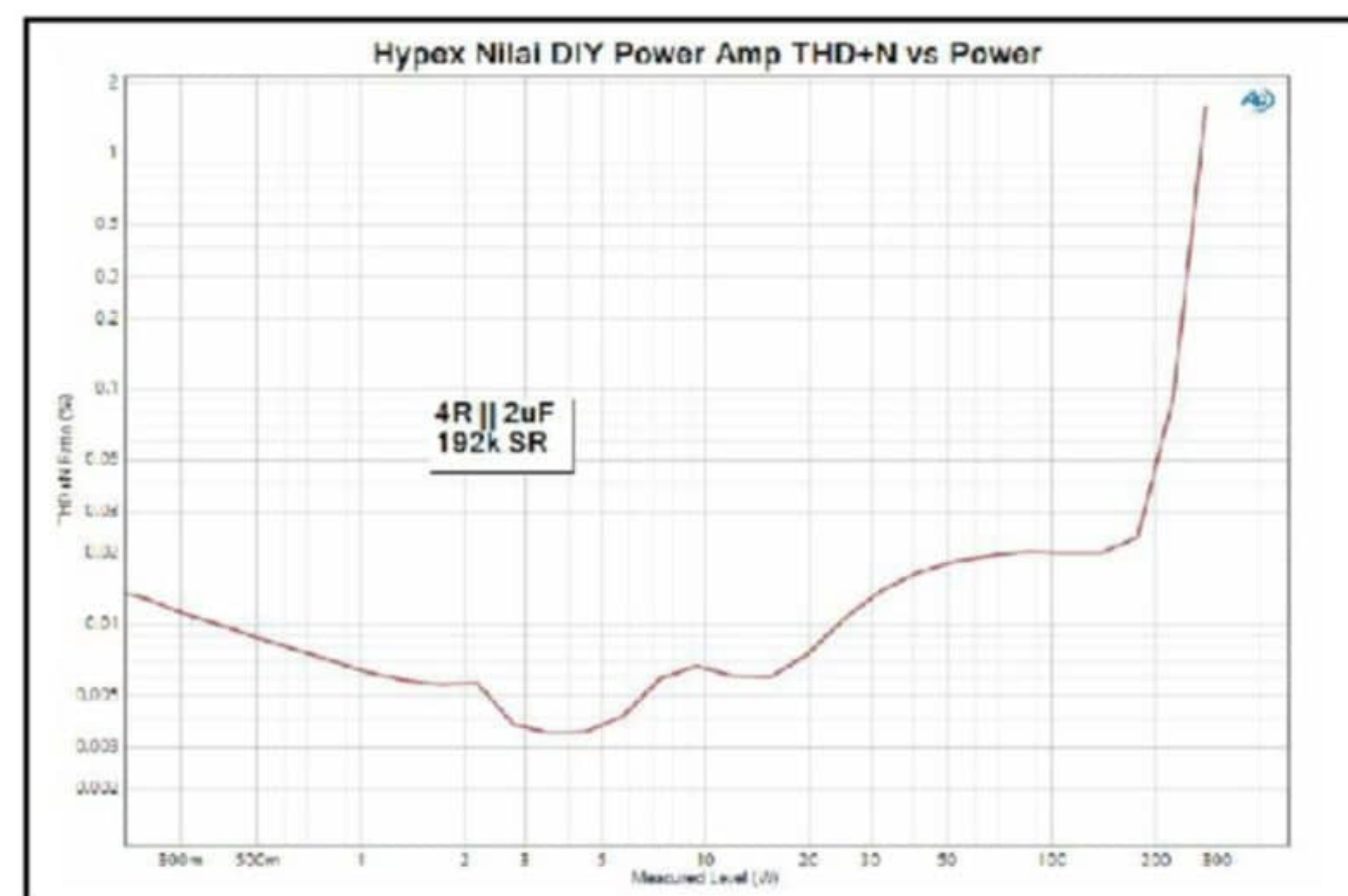


Figure 12: Repeating the power sweeps with a brutal 4R in parallel with 4μF load does not fluster this amplifier! See text for explanation of X-axis.

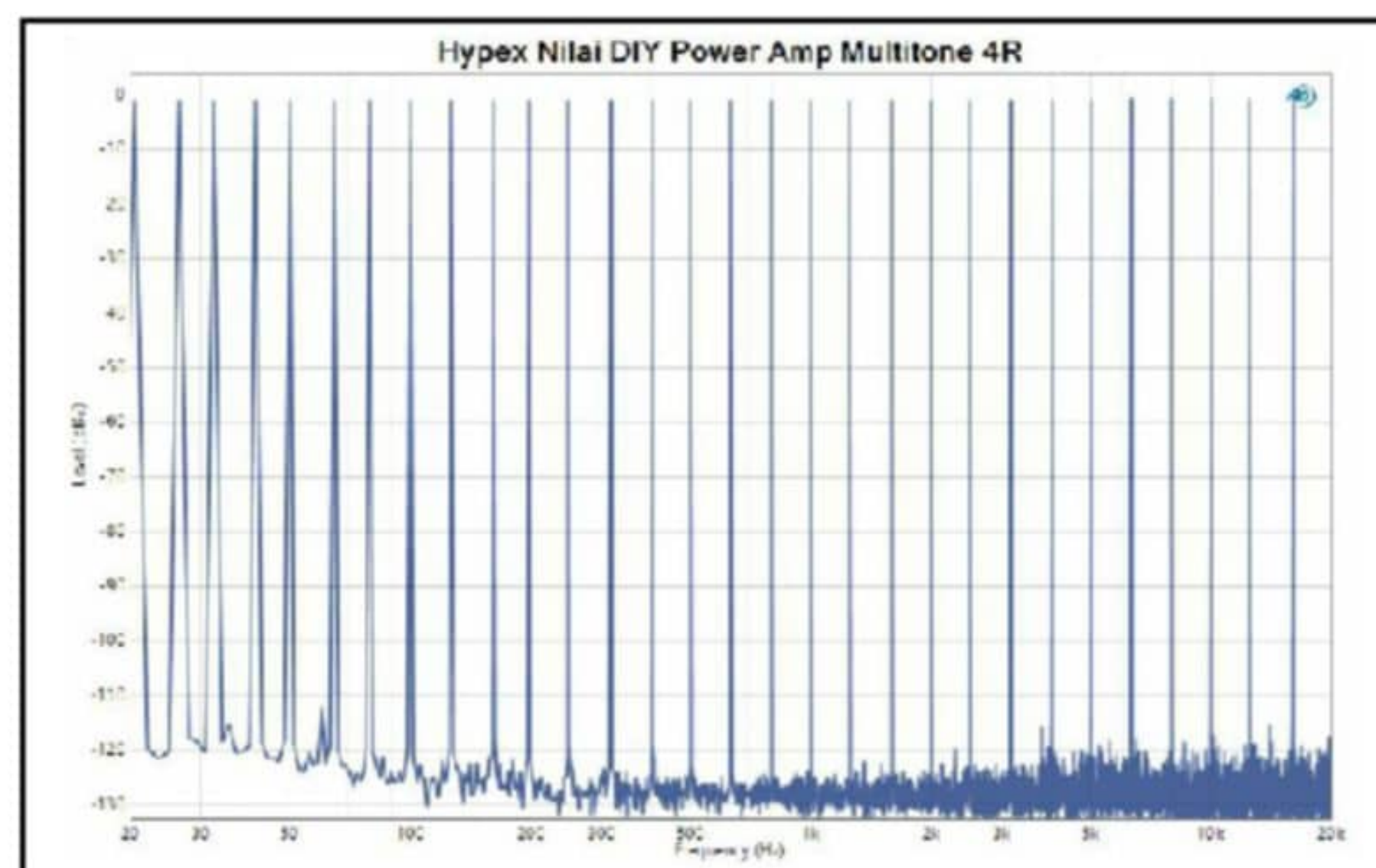


Figure 13: A multitone spectrum shows an enormous dynamic range when handling complex signals

Resources

"DIY Class D Preamplifier User Manual," Hypex BV,
www.diyclassd.com/media/ce/59/d9/1680600951/DIY%20PreAmplifier%20User%20Guide%20R01.pdf

J. Martins, "Hypex Presents New Nilai500 DIY Audio Amplifier Module," *audioXpress*, May 16, 2022, <https://audioxpress.com/index.php/news/hypex-presents-new-nilai500-diy-audio-amplifier-module>

"Nilai 500 DIY Mono Kit," Hypex BV,
www.diyclassd.com/media/4f/27/39/1678454104/Nilai500DIY%20Mono-kit_User%20Instructions_R1.pdf

S. Yaniger, "The Hypex NCore NCx500 Amplifier Module: Doubling Down on a Flagship," *audioXpress*, May 2023.

with 4μF (**Figure 12**). Note that the power on the X-axis is not going to be an accurate measure of actual power delivered since the APx software calculates it based on the measured voltage across a pure 4R reference. Nonetheless, there is nothing untoward in the data. Despite the rumors, this appears to be a total non-issue, but it is a very nice demonstration of the Nilai’s exceptionally good load tolerance.

Continuing, I ran a multitone measurement with a 4R load. Each component was at a 1V level. The results are shown in **Figure 13**; unsurprisingly, the spectrum is almost perfectly clean with no distortion or noise components rising above -110dBV (3μV) and most lying below -120dBV.

Finally, I also measured the common mode rejection (CMR) at the inputs (**Figure 14**). At hum frequencies, the CMR was better than 105dB, deteriorating to “only” 65dB in the top octave. This demonstrates superb balance and noise rejection. The amplifier’s output impedance was lower than my measurement limit (0.02Ω), which goes a long way toward making this amp truly load invariant. These measurements are absolutely exemplary.

Conclusions

The electronics under test here delivered the goods, noiselessly, efficiently, and with tons of power. I have some quibbles, but not with the performance—the measured results are absolutely outstanding. I think that in 2023, any preamp should have digital

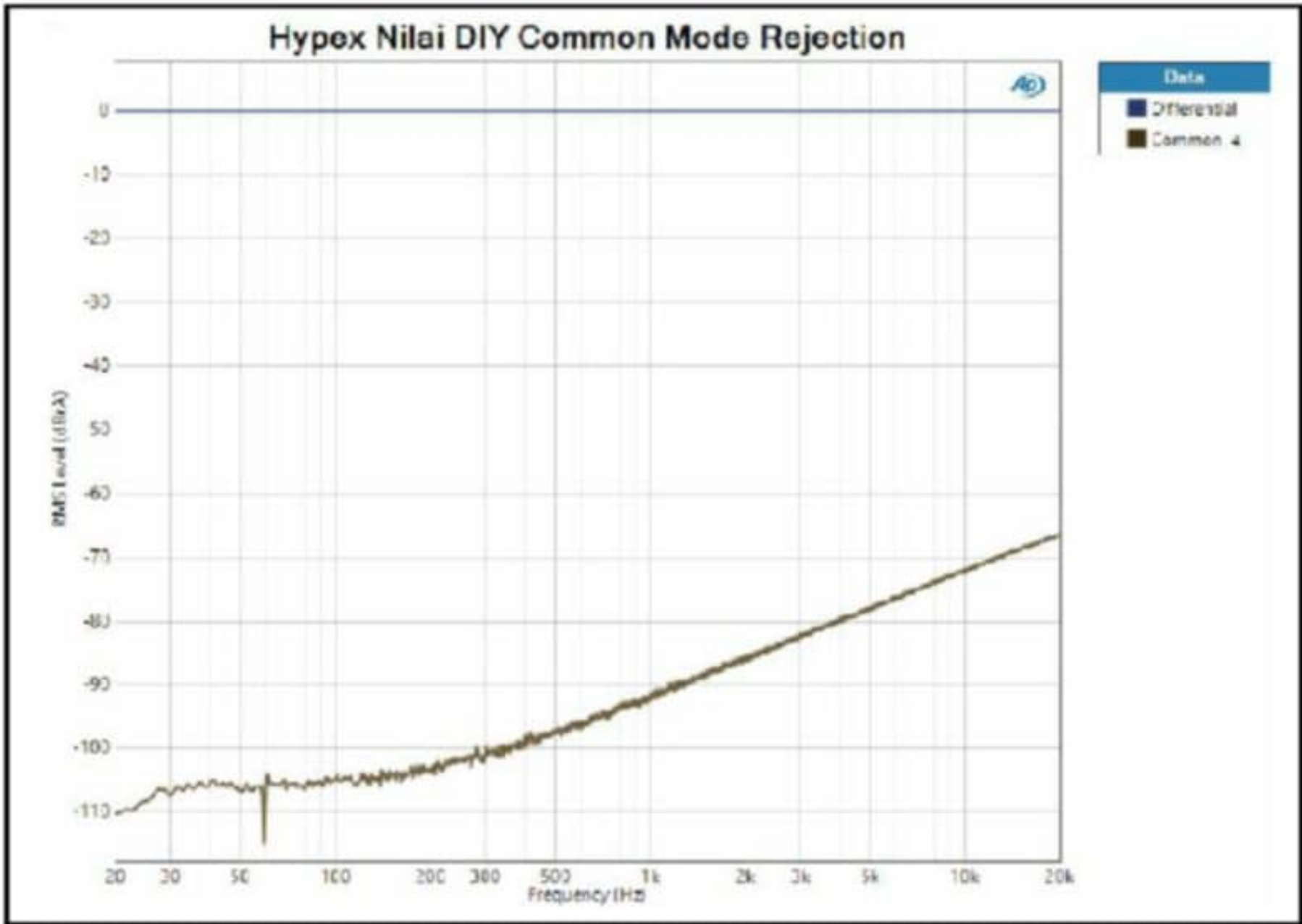


Figure 14: The common mode rejection indicates excellent noise immunity, especially at hum frequencies.

and USB inputs. I really wish that the power amp power switch was more visible. And... well... that’s about it.

Hypex continues to bring out superb products showing smart engineering. When they claim “load independent,” they’re not kidding. The amplifiers handled everything I threw at them flawlessly including torturous reactive loads. The preamplifier matched the amp’s performance and added a very user friendly (and likely highly reliable) user interface. The Nilai line is killer good. Highly recommended. 

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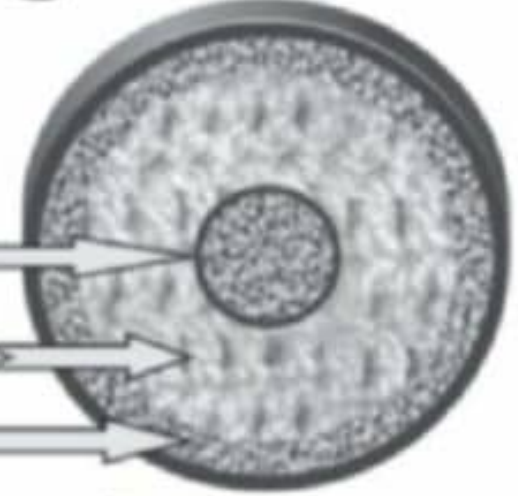
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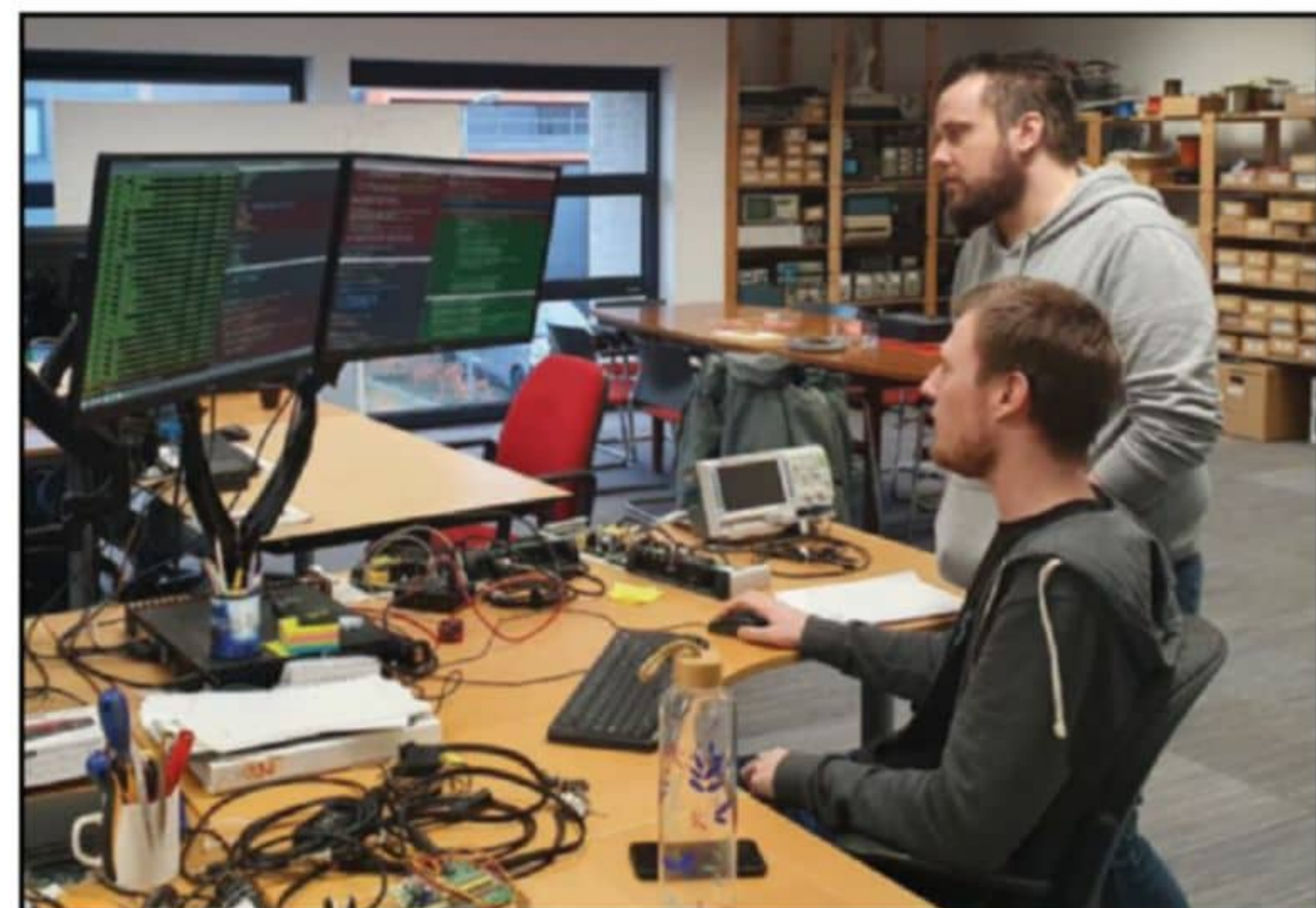
Grimm Audio

Building the Dream

By
Jan Didden
(Technical Editor)

As a relatively young company, Grimm Audio has consistently put its mark on both the pro audio and hi-fi scene. *audioXpress* thought it was time to meet the people behind this company, its products, and philosophy. Jan Didden went to visit them in a suburb of the city of Eindhoven in the south of The Netherlands.

Eelco Grimm, a Dutch audio and recording engineer and lecturer, lent his name to the company that was formed in 2003, but there is a solid partnership behind that name. Beside Eelco, there's Guido Tent, whom users may know as the founder of Tentlabs, which Guido still manages. Another founder of Grimm Audio is Peter van Willenswaard. Although nominally retired, he still maintains an active interest in the company and frequently contributes project designs. Peter van Willenswaard is the owner of AudioMagic (www.audiomagic.nl), which specializes in modification, upgrade, and design of tube audio equipment. Rounding out the founding quartet is Bruno Putzeys, another household name in high-performance audio. Grimm, Tent, and van Willenswaard go back to the 1980s of last century where they discovered high-end audio through common projects and even running their own publications. Already at that time, the interest was in comparing the results of listening tests and measurements and trying to



Software engineers Simon Jaasma and Michiel Schriever at work.



Grimm founders Eelco Grimm (L) and Guido Tent (R) with Grimm's acclaimed LS1b speaker.

find out what made a piece of equipment sound good and how that correlated with measurements. All those questions haven't been fully answered yet, but a lot of insight was gained and subsequently put into practice.

Eelco Grimm cut his audio teeth recording music with his company "Fairytapes," which used equipment designed and built by his friends Peter and Guido. After hearing an Audio Engineering Society (AES) presentation by Bruno Putzeys about a novel ADC, they asked him if he would license his design. Instead, Bruno joined the partnership. Eventually, Bruno Putzeys left the company to explore other paths. Peter van Willenswaard has nominally retired but is still very much involved in the company's designs.

With Eelco Grimm initially focused on studio work and Peter van Willenswaard and Guido Tent on home hi-fi, they were in the unique position to benefit from experiences and developments in both areas. This has been a common theme for Grimm products to this day. Although not often realized, the goals and requirements for equipment in both recording and reproduction contexts have many common elements, probably the most important being the need for neutrality, while still being able to convey the emotional content of the music.

That may seem contradictory, but as Eelco Grimm explains, neutrality is a multi-dimensional property. A speaker can be neutral with a flat frequency response, or in its dynamic behavior. An amp can be neutral as in very low distortions but can still sound different to another amplifier which also has low distortion. So, it is a whole complex of "neutralities" that must be balanced without compromising the reproduction of the emotional content of the recording. Not easy to put into hard numbers, but Grimm Audio's success heavily leans on decades of experience and research into these factors by the diverse group of partners.

Grimm Audio employs about a dozen staff and, uniquely, strives to keep all significant activities in-house. "We don't outsource

assembly for instance, but build all our products in-house, under our own control. We also do all our R&D in house, and in fact our R&D staff far outnumbers the assembly and logistics manpower. We write our own software, and we go so far as to write our own software libraries—reliability and long-term support are very important to us!,” explains Guido.

Pure Nyquist

As an illustration, Eelco Grimm talks about how the team approaches D-A conversion.

“We all know that the ‘Perfect Sound Forever’ adage from the early CD days really missed the mark. There’s a lot that can still go wrong in a bit perfect transfer. For a start that’s because digital conversion is largely an analog affair.

In a DA converter, the power supplies and the output circuitry are clearly analog. But also, the final stage of conversion within the DA chip, where digital bits are turned into analog voltages, is more analog than digital, for example in the jitter on its clock. This final stage on the edge of digital and analog appears to be one of the most sensitive ‘analog’ parts of the whole audio system. That means that for example power supply noise is very critical.

But it’s not just the analog side that matters. We figured that if jitter is a correct conversion at a slightly wrong time, an amplitude conversion error is a slightly wrong amplitude at the correct time. Both potentially have a similar impact on the overall reproduction quality (Figure 1). Mind that amplitude errors are not limited to classic multibit converter designs. Rounding errors like quantization noise are also a form of amplitude error. Important to note is that these errors already have impact at a very small timescale, so the usual solution of applying dither has only limited effect. Dither solves the problem statistically over time, but not instantly.

And, if you think about these very short timescale effects, what is then the effect of all that processing we do in filtering, decimation, and such? Do we really have enough resolution to avoid these very small (timescale) effects? To gain more insight, we performed several experiments that were real eyeopeners. If you dig deep enough, all those things such as clock circuits, oscillators, power supplies, mechanical vibrations, EMI, rounding errors in decimation filters, all those so-called ‘minor issues,’ have their effect, and are often audible.

And so, you end up with shunt regulated power supplies with 130dB noise and ripple suppression, which continues to 1MHz ... And you start to understand the origins of those “magical” stories,



for instance, that painting the edges of CDs green had an audible effect. The resulting absorption of scattered light inside the disc very slightly improved the readability, which was measurable in the positioning servo activity of the CD drive. Since the servo and the oscillator often share a common power supply or ground, this resulted in a measurable (and audible) change in jitter. Ultimately, this is an example of bad engineering. The power supply of an oscillator needs to be as clean as possible, and currently we are getting down to 2nV/RtHz.

Back to the impact of rounding errors in decimation filters. In this context, it is interesting to reflect on the popularity of non-oversampling DACs in some quarters. If you expect that your replay electronics are impervious to a load of high-frequency hash and lots of aliased signals, then leaving out the filter means you will not make any processing errors in that area. Pick your poison. But of course, the real solution is to remove all the images with well calculated filters, which means: with the smallest calculation error. The actual filter magnitude shape, the roll-off, is maybe the least important factor. For the MU2 (discussed later) we have developed a filter that is pretty much ideal in both the time and the frequency domain, and therefore we do not offer switchable filters. We don’t want to bother our customers with the burden of having to listen to a sizable portion of their music collection to determine what sounds best, only to find that on another day or with other music they would prefer another setting. It is subtle, and

The CC1 (a) or CC2 (not shown) World Clock and the UC1 Universal Converter (b) are aimed at pro recording setups, although a CC1 would equally be at home in a residential hi-fi setup.

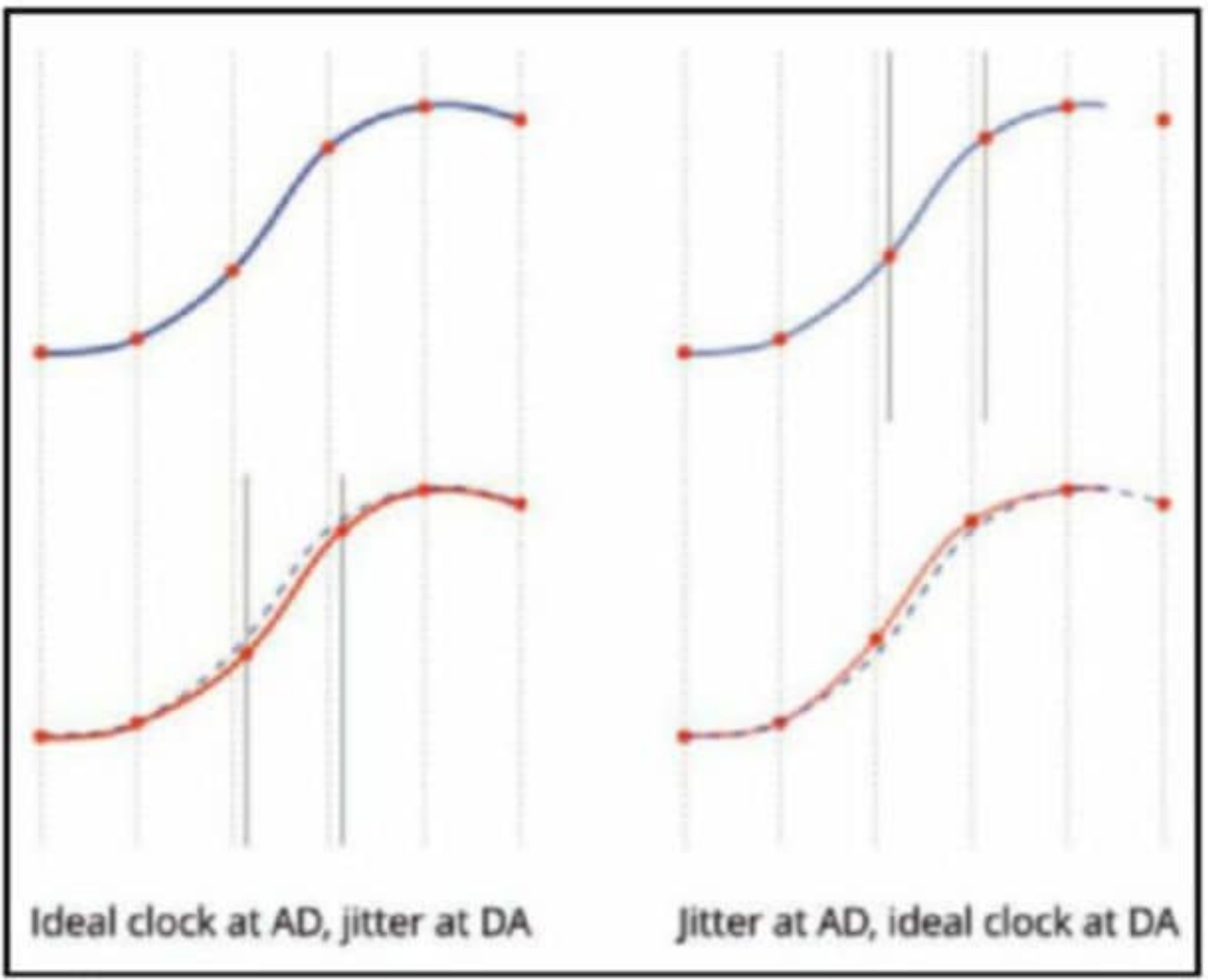


Figure 1: On the top left side an analog signal (in blue) is acquired by an AD with a “perfect” clock. The lower left graph shows what happens to the reproduced signal (in red) if the DA converter has a jittery clock. The right side of the picture shows that a correct conversion but at a slightly wrong time leads to similar errors.



Grimm's MU1 Music Player offers total control over the whole process and is both a Roon Core and Roon End Point.

dependent on the conditions and music. We really want our customers to listen to their music and enjoy it, period! We get a lot of feedback from people telling us they stopped their cable neurosis and whatnot and just play and enjoy their music. And, in the end, that's what makes us happy, that's what we are after. That's the dream we are building, both for us and our listeners."

Grimm's team found that getting the amplitude errors low enough required much more processing power than was available in typical converters, which led to the use of a powerful FPGA calculation engine in the MU1. It offloads the first (and most demanding) upsampling stage from the DAC chip to the FPGA. It was decided to output the signal in AES or S/PDIF form to the DAC so the MU1 clock quality could be transferred to the DAC (remember

that clock and amplitude errors have similar impact). That is the reason why the MU1 does not offer USB audio—the clock in the DAC would be another unknown. By offering the signal in AES or S/PDIF format with its embedded clock, Grimm makes certain that the DAC is benefiting both from the MU1 processing and its high-quality clock.

Grimm Audio's Philosophy

A typical audio chain has many different pieces of equipment that rely on each other to get the signal from the media out into the air. And you can design the very best piece of audio electronics, but if the parts of the chain before or after are not up to the same high standard, your efforts do not bear fruit! So, Grimm tries to keep the whole chain under their own control. On the recording side, they offer everything from the converter, as well as their own cable designs, to have full control from sound pick-up to the digital audio file and guaranteeing pristine performance: the CC1 and CC2 World Clocks, and the UC1 converter.

From MU1 to LS1

The MU1 is the heart of Grimm Audio's music system, and it is much more than "just" a streamer. It has not just a Roon End Point but also a Roon Core on board. And it has three inputs for physical digital sources. There are three XLR and RCA outputs providing several AES and S/PDIF signal sources, which can be used for stereo DACs but also support surround playback. The (AES digital) Cat5 output of the MU1 can be directly coupled to their acclaimed speaker, the LS1 series. This is the ideal situation because everything, from source to the acoustic output, is now under Grimm Audio's control.

The LS1s do look quite different from the usual speaker shape, whether Pro or Home Hi-Fi oriented. Calling the LS1x a speaker is missing the point. This "speaker" is a complete audio system with digital and analog inputs, power supplies, several channels of Class-D amplification and a powerful DSP system that ties it all together. The form factor is unusual for a speaker. Much thought has gone into it.

The LS1 has a relatively wide baffle but is quite shallow, and has extremely rounded sides, that double as stands. This shape is based on a solid engineering approach, that started with the realization that in principle, we can never exactly reproduce the original sound field as shown in **Figure 2**.

Apart from that basic issue, the room adds its own reflections and reverberations as a function of the speakers' directional pattern over frequency. Wrangling such problems into an acceptable



The flagship LS1be sports a beryllium tweeter and fully integrated reproduction electronics.

compromise is not done by starting with what you want to achieve, but rather by looking at and prioritizing what you want to avoid. [1]

A flat, low distortion speaker sounds great in an anechoic room, but in a real room you'll have to deal with off axis radiation; too wide makes that trumpet sound as coming from all directions; strong frequency dependent dispersion makes the direct-indirect sound balance quite frequency dependent as well.

As Eelco explains, an omni dispersion at low frequency is acceptable, but we need to keep mid to high frequency directionality under control. A transition to omnidirectionality below 250Hz is fine, and if we can get reasonably constant directivity above that, we have a design goal we can work to! Wide baffles have a bad rep but that is not because of the wide shape, but because of the strong diffraction at the edges. Rounding off the edges generously takes care of that, and this is immediately apparent in the LS1 series, where the rounded sides double as stands (Figure 2).

"With the tweeter above the woofer, as is customary, you would also need to round the speaker top similarly, but that looks plain ugly, so we chose to place the tweeter below the woofer and round off the bottom of the baffle. The edge diffraction on the woofer side is largely masked by the influence of the woofer cavity so the woofer side edge may stay hard."

The shallow box also means it can be placed close to the rear wall with relatively short path lengths for wall reflections, which helps to maintain flat dispersion at low frequency in this position.

Finally, this shape is ideal for adding a subwoofer: just put it on the floor between the legs! That is an ideal place as it largely avoids low-frequency reflections on the floor between speaker and listener. The (separately available) subwoofer contains a high-power class D amplifier with a digital motional feedback loop. The system can be switched from two-way to three-way in the LS1 control software, which engages a second linear phase crossover so the phase coherency of the system stays perfect.

From the base model LS1a to the flagship LS1be with the beryllium tweeter, they all offer a fully integrated reproduction system. Although they accept both analog and digital inputs from any source, the ideal match would be the MU1.

Again (and yes it might get boring by repetition, but it is so important I mention it again), everything is under Grimm's control, from music source to acoustic output. This tight integration between the speaker drivers, the amplifiers, the processing, and the power supplies is an invaluable opportunity to get it all working together for the best results. It allows them for instance to select drivers that you would not be able to use in a traditional passive system, but that offer additional degrees of freedom for optimization of the whole system. Don't worry about having the right cable, or an amplifier with the right characteristics; get a MU1, an LS1 speaker and you'll have one of the best audio reproduction systems that money can buy.

To give readers an idea of the length Grimm is willing to go to guarantee the best possible performance, consider the company's calibration process. They developed an algorithm that simulates thousands of possible EQ combinations to achieve the "target curve" with the smallest amount of EQs and the least

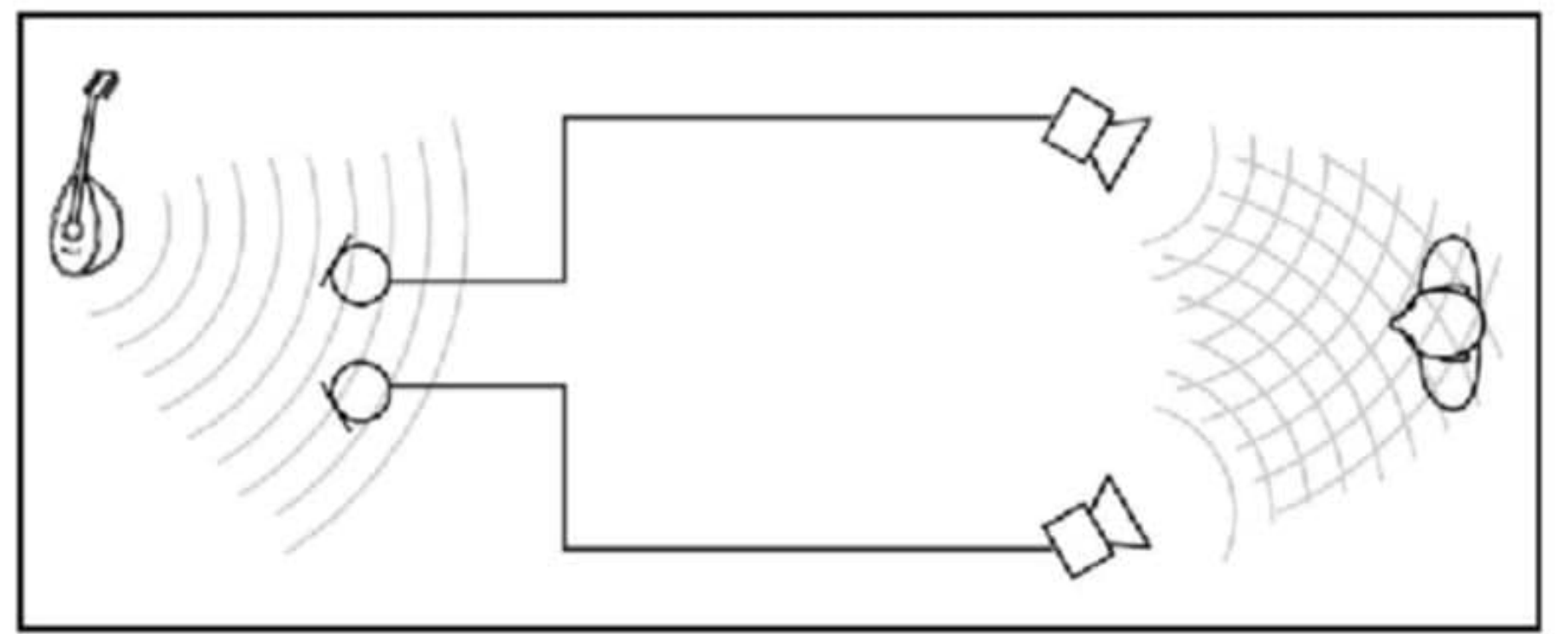


Figure 2: An exact reproduction of the source sound field through a stereo system is simply not possible. (Image Source: Grimm Audio)

extreme settings for each individual speaker. The algorithm takes the acoustic sum of tweeter and woofer into account and aims for an "acoustic optimal fourth-order Linkwitz-Riley crossover" between the two. During the whole process an operator evaluates intermediate results and directs the algorithm. Computer-aided calibration is not only a time saver but often presents surprisingly minimalistic EQ settings with a better curve matching than even the most experienced operator would manually achieve.

Where Does Grimm Audio Go from Here?

I asked Guido Tent about future products. This is always a sensitive question, but he was willing to discuss two ideas. First, there will be a MU2 in Grimm's future. Where the MU1 has strictly digital (AES or S/PDIF) outputs to interface with the LS1 series and other active systems, the MU2 will have its own internal DAC, and will interface with analog amplification and reproduction systems. With the MU1, Grimm can "control" everything up to the digital output connectors. The choice for internal upsampling and AES and S/PDIF outputs unburdens the downstream DAC from those tasks, but after that, it's out of their hands. The coming MU2 will include a discrete DAC of their own design and output an analog signal, so this control will extend to the analog interface with analog (pre)amplifiers, eliminating DAC variables in the quest for the best possible sound.

Second, a phono preamp is in the making. As always, Grimm follows the theory: First think about what you are trying to do in a fundamental way. In this case, rethink the reproduction requirements for a phono signal. Many designs combine the lowest noise and lowest distortion circuitry they can come up with, coupled with the required RIAA correction.

But if you think about it in a fundamental way, you can question whether with vinyl you really need -120dB harmonic distortion and/or -100dB noise level. What other degrees of freedom does that give the designer to optimize the audio path for neutrality and "emotional reproduction"? Another adventure embarked upon, which will eventually lead to another potentially groundbreaking product.

After an extended listening session, which included a surround setup with five LS1s, we went to a local restaurant to continue our discussions on audio design and reproduction. A discussion that will never end, but to which Grimm intends to make significant contributions while building its dream! 

Resources

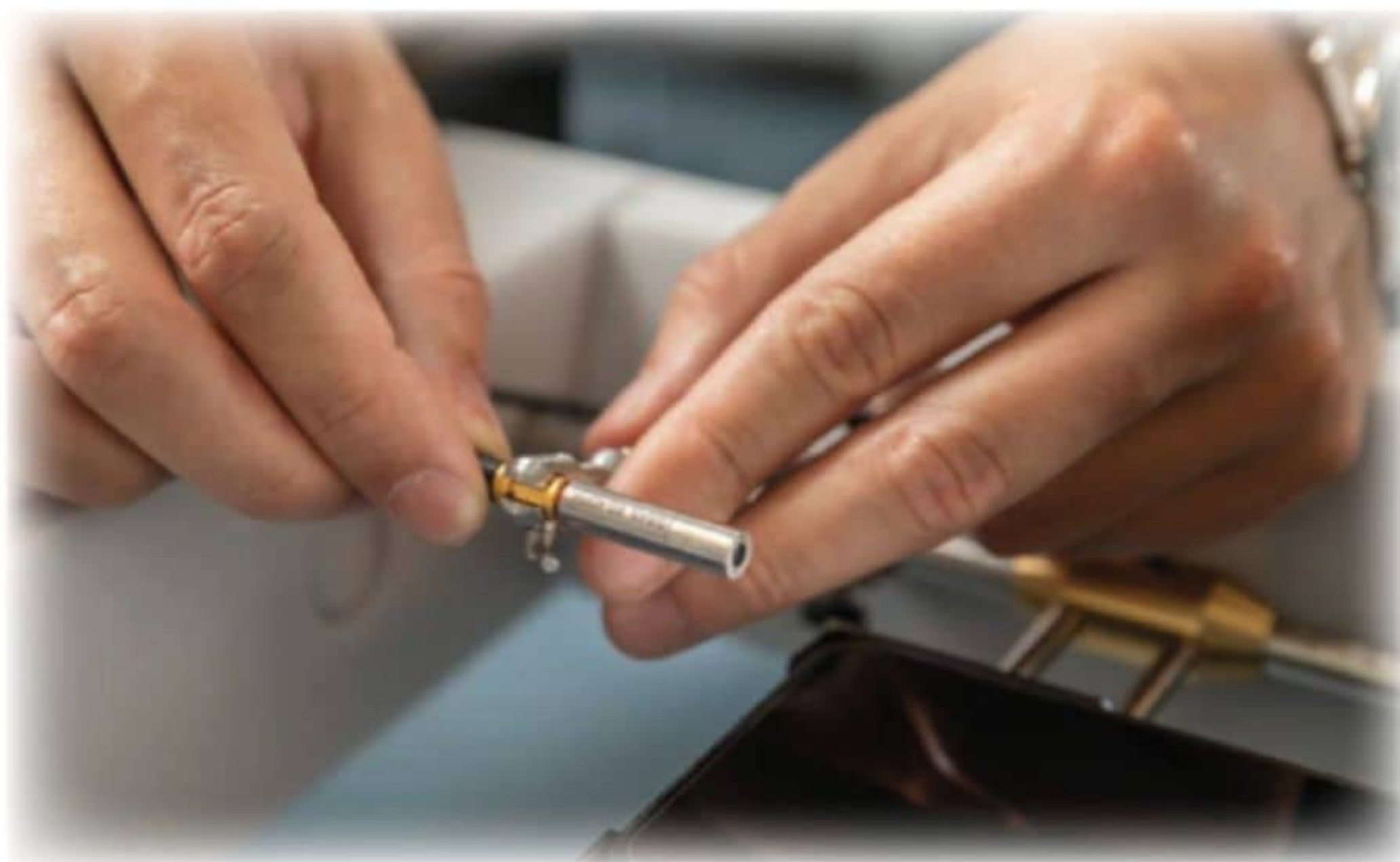
[1] "Speakers White Paper," Grimm Audio, June 2010, www.grimmaudio.com/publications/speakers-white-paper

GRAS EQset “Fire and Forget” Measurement Microphone Series



The new EQset measurement microphone series from GRAS Sound & Vibration in Denmark promises an improved workflow with less scope for errors and cost savings for production line acoustics measurements. *audioXpress* interviewed Rémi Guastavino, Director of Product Management at GRAS Sound & Vibration, to uncover the details.

By
Rens Tellers
(SonicDesigns)



Rens Tellers: Rémi, thank you so much for taking the time to talk to us! I am familiar with GRAS microphone products; my latest acquisition was a wideband 140kHz mic for specialized acoustic measurements. So, when I saw the announcement for the new EQset series I was intrigued. Can you tell us what use these microphones are targeted at?

Rémi Guastavino: My pleasure! This microphone was designed to be used on production lines. So, this means from small audio devices upwards; mobile phones/microspeakers/tablet/headphones. For improved

accuracy when measuring smaller devices (speakers and microphones), it is good practice to integrate your sound sensor in a receptive area of similar size as the object to be measured, and close-mic the measurement. This microphone contains a small optimized acoustic sound port to match the small devices being tested. The other important parameters are ease of use, cost effectiveness, no calibration required, and environmentally friendly. These are not necessarily needed for lab environments but make this microphone very usable in demanding environments!

To give you some more background: Measurement microphones have a big spread in frequency response and sensitivity. Production microphones are an order of magnitude cheaper than measurement microphones. The main limitation is the capsule, which should be low cost. But note that environmental stability is also a real issue since the sensitivity of production microphones is varying roughly $\pm 1\text{dB}$ due to variations in humidity and temperature. That's when I got the idea of using a new cartridge design that is not affected by environmental factors, with internal digital adjustment and sent them out fully calibrated.

So that lead to the project's main targets: reduce spread in specifications compared to more traditional microphones; powered either by phantom or CCP [constant current power] supply and, preferably manufactured in Denmark!

Rémi Guastavino,
Director of Product
Management at
GRAS Sound &
Vibration



Tellers: Did you consider a digitally connected microphone? USB-connected microphones for measurement are found more and more it seems.

Guastavino: That’s a very good point and my first design ideas did consider that! However, market research showed that GRAS’ potential customers (mostly based in the measurement industry) wanted analog microphones with the same dimensions as their existing traditional microphones. This would facilitate the quick and direct replacement of microphones within existing production lines.

So that ruled out a USB microphone. But there is also one positive side to this. Consider clock jitter; since the same internal mic’s clock is used for AD as well as DA conversion and the digital domain is separated from the measurement equipment, there is no additional jitter introduced by different clocks, which could happen as soon as you start connecting multiple USB microphones with individually slightly different clocks to the same computer.

Tellers: Can we take a closer look into the microphone details? How is the calibration of each individual microphone carried out?

Guastavino: Sure. I will have to omit some details, which are company confidential, but the basics are as follows. Usually, measurement microphones are calibrated with an electrostatically calibration standard, but that’s not possible with the EQset due to the sound port design protecting the sensing element on this microphone.

A custom test box is used for the calibration process, consisting of a special speaker and reference microphone (**Figure 1**). The calibration process for the microphone-under-test is comparing it to a GRAS 1/4” pressure measurement microphone (Patent Pending). The setup is connected to an Audio Precision analyzer after which a company-designed comparison method is used to tighten the microphones tolerances to a minimum.

A programming interface is connected to the microphone to apply the compensation, The settings and calibration are made by GRAS in their own labs and it’s not possible or necessary for users to do their own calibrations.

Tellers: So, in short, the calibration is using a kind of in-house Golden Standard.

Guastavino: Yes, the calibration flow is as follows. The test starts with a 30-second stabilization time after power on after which a first measurement is carried out. Compensation based upon the



The new GRAS EQ 40PM miniature production line microphone was the first to use the EQset technology designed to reduce the risk of measurement errors. A short 20kHz pressure microphone with a dynamic range from 30dB (A) to 120dB, and the same dimensions as traditional production line microphones, the EQ 40PM is aimed at increasing efficiency and improving test accuracy in the consumer audio and telecom space.

measurement is calculated and applied to the microphones internal to match the desired frequency response and the gain. Before finally boxing the product, a verification is done to ensure that all corrections were applied correctly.

So, with all of this in mind it was time to rethink the design. That got us to the following setup, a new pressure microphone called EQ 40PM with the following specs:

- A 20Hz to 20kHz pressure microphone with a dynamic range from 30 to 120dB (A);
- A new capsule design for environmental stability
- Internal A-D and D-A conversion
- Factory frequency response compensation
- Factory gain adjustment compensation
- Both phantom as well as CCP powered preamps are supported
- Staying within current production microphone price ranges

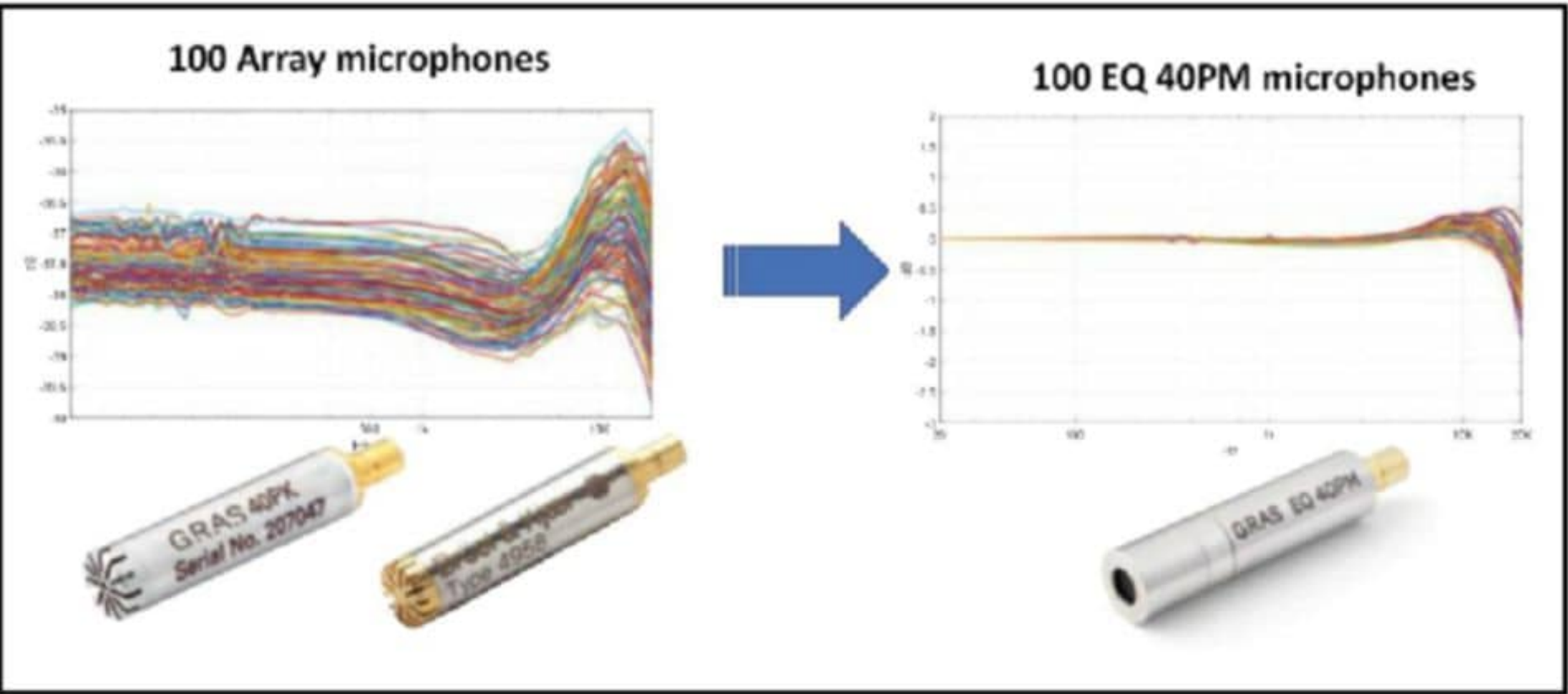


Figure 1: The consistency of EQset fixed sensitivity and flat curve reduces measurement uncertainty and simplifies measurement microphone setup, monitoring, replacement, and corrections, without any individual microphone adjustments or corrections. The simplification of setup and other tasks saves time in the setup itself and operator training time. There is also one additional benefit that outshines the time savings: EQset all but eliminates the inherent risk of false passes on production lines or correction error for frequencies outside the calibrated frequency in R&D—common issues with other microphones in the market.

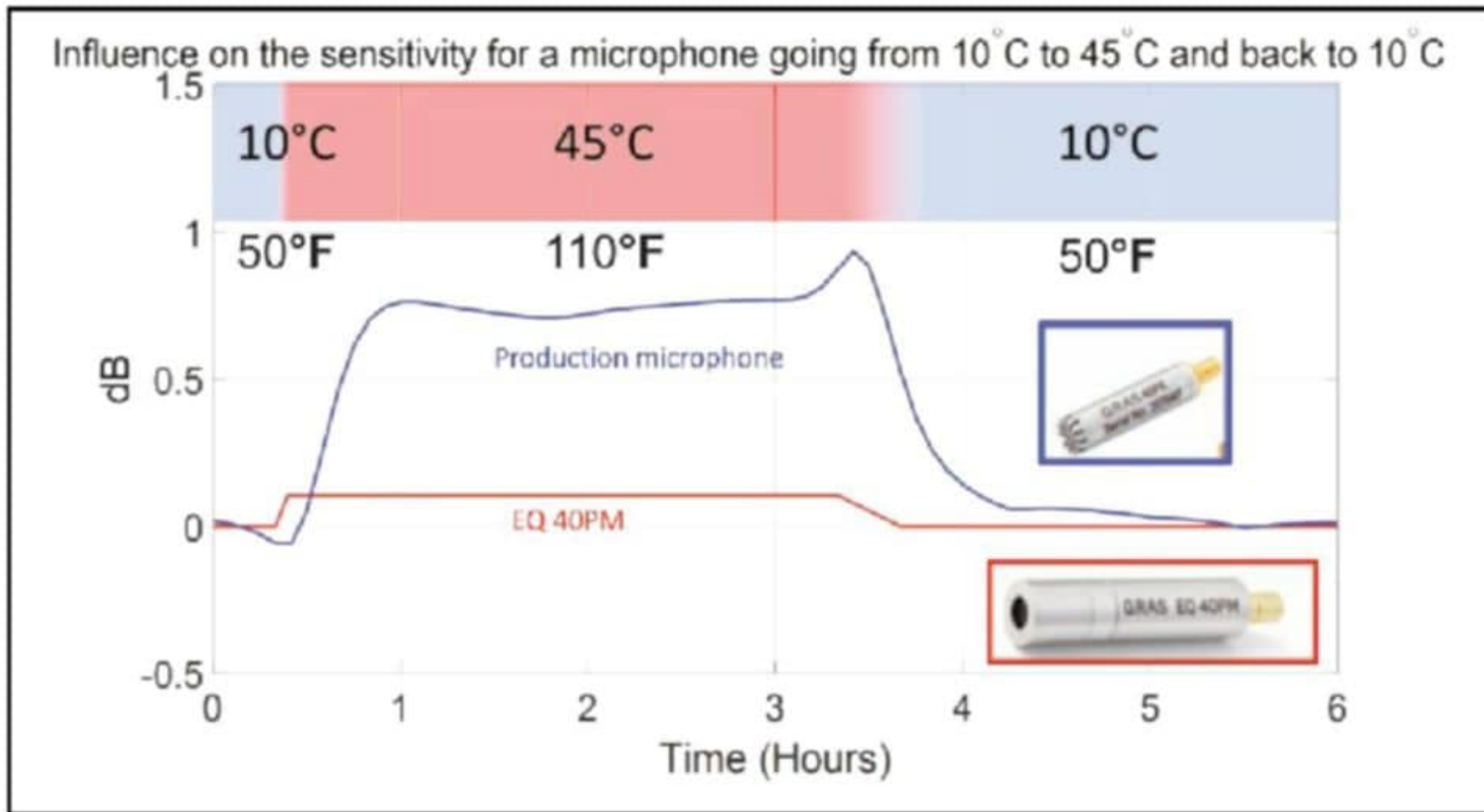


Figure 2: Another benefit of the uniformly fixed sensitivity and flat response curve of each EQset microphone is that because there is no need to make individual corrections for each measurement microphone, there is, therefore, no need to store that correction information on the microphones via TEDS. This enables EQset microphones to be used effectively with systems based on audio sound cards using 48V phantom power as well as any CCP-based system already in use, so there is no need for price-prohibitive new equipment or retooling of existing CCP-based systems.

Inclusion of all these techniques in the traditional dimensions of 34mm x 7mm requires some extensive engineering skills, but we managed to squeeze all of it in this little pressure-based microphone!

But, to be honest, using a digital domain inside a microphone also causes some limitations. SPLmax is a bit lower and SPLmin a bit higher than what we are used to in a traditional microphone. The DA converter determines the dynamic range to about 90dB, which can be placed where it is most useful. Considering our customers' demands, we set the range at 30dB to 120dB, which indeed covers most of the normal measurement conditions.

Tellers: You also mentioned a novel pressure sensing element. Can you give some more details here?

Guastavino: The new sensing element (capsule) is designed to be more stable towards humidity and temperature. Therefore, this gives a major advantage in environmental terms over traditional capsules, while remaining in a friendly price range.

So, separately from the DSP use, GRAS also managed to improve environmental stability.

So, let's show some data here. **Figure 2** shows a comparison between a traditional and the new EQ 40M in terms of temperature stability; I believe the graph speaks for itself! With regard to humidity sensitivity, it's even worse; including hysteresis, it can take days for a production microphone to go back to the original sensitivity!

It's all published in the EQ 40M datasheet, of course. One spec you won't find is for the AD and DA process; that is internal and transparent to the user.

Tellers: One other thing not so often seen in the datasheets is the microphone's phase response. Are you publishing that?

Guastavino: Good point! GRAS evaluated the phase response and concluded it's extremely consistent. The additional phase induced by the DSP is constant and thus similar to the phase shift you'll get by slightly changing the microphone position. So, no correction is made to the phase response since it doesn't seem to be a necessity. But, as with all microphones, the devil is in the positioning!

Tellers: You designed this product in close coordination with customer requirements. How do you make sure those requirements are known and taken into account?

Guastavino: That's an important issue for us and has been from the beginning. To start, GRAS has a customization program: if customers want to have something that is not in the standard catalog, we will evaluate such requirements and come up with a suitable solution. It's in our DNA.

An important part of this is GRAS actively going out to customers and taking part in the customers' processes. We have an internal customization committee which will look over it and decide whether new ideas and/or projects are feasible within the current portfolio or should be considered for future product development. And it is actually a two-way street as exemplified by this new product offering. A lot of the input for this microphone's technique came from the customers, not only the used techniques but also priorities (like what's more important frequency or dynamical range or USB). Based upon the customers' feedback, GRAS changed the product to contain an analog output instead of the USB connection initially intended. Customers really appreciate being part of the design process and seeing this new product being aligned to their wishes.

About the Author

Rens Tellers runs a company on electrical automation, with more than 15 years of experience in electrical engineering that he translates into the audio world. He currently focuses on improving audio amplification and measurements thereof, which has earned him member status of the Audio Engineering Society (AES). Rens has been an active musician (bass guitar and piano) and music enthusiast his whole life, with extensive work in the recording and mixing of music. One day, he hopes to dedicate his entire time to making music sound better in all aspects.



As a side note, let me mention that during the recent lock downs we really missed not always being able to go out to the customers and get all their internal notes, not only during meetings, but also the important discussions after the meeting at the coffee machines...

Tellers: What sort of customers do you serve?

Guastavino: GRAS' customers can be found in all the various fields of product measurements. On the high side of the segment, you'll find customers with their own test equipment—high-end brands which use high-end test equipment including high-end miking and using conditioned spaces for testing. On the lower side of the segment you'll find production lines where simple sound cards are used requiring quick measurements results. And at the bottom of the segment, there is also a group which doesn't test because it is simply too expensive for the product (think of cheap earbuds for 10 euros at the airport).

Tellers: Since this microphone is priced in the lower measurement microphone price range, one could expect more of the mid- and lower segment users to jump on-board with this product. What help is available for less experienced users?

Guastavino: As a customer, you get the microphone with a stated sensitivity. There are no manuals to wrestle through, as the microphone can just be connected and used with no adjustments or TEDS calibration files.

Unpack, plug in, and use, that's it. If you really need it, there's a QR code on the package that takes you to our website, to the microphones landing page, containing all the product information, and general topics such as on/off-grid usage and free-field vs pressure usage. And we make it a point that these microphones do not need any calibration, not for the microphone, and not for the used cable. That said, it is probably a good idea to check your whole measurement chain once in a while.

Tellers: It seems to me that this is the first step toward a new series of intelligent microphones. So what is the future?

Guastavino: It is in the plans to evolve the same technology to higher performance measurement microphone products, aiming for higher frequency bandwidths. There are a few hurdles to take here. The biggest limitation of dynamic range is due to the DSP and DA conversion, which currently resides



Online interview with Rémi Guastavino, with Rens Tellers, Nick Dimes and Caroline Gabbers from Axiometrix, and *audioXpress* Technical Editor Jan Didden.

at about 90dB. Traditional production microphones can reach around 120-130dB. GRAS is working on extending this range.

Tellers: Maybe in a future release, you could add some user setup functions, like moving the dynamical range to a specific position? Or maybe compensation for the grid in front of the capsule?

Guastavino: This is an item for the future, as for the 20Hz to 20kHz range the ¼" grid is small enough to avoid influencing the frequency response within this range. 🚫

About GRAS Sound & Vibration

Founded in 1994 by Danish acoustics pioneer Gunnar Rasmussen, GRAS has since grown to be a worldwide leader in the sound and vibration industry. The company develops and manufactures state-of-the-art measurement microphones and related equipment. As part of Axiometrix Solutions, they complete the group as being a solution provider for testing environments. This includes applications and solutions for customers within the fields of aerospace, automotive, audiology, consumer electronics, and other highly demanding industries.

www.grasacoustics.com

About Rémi Guastavino

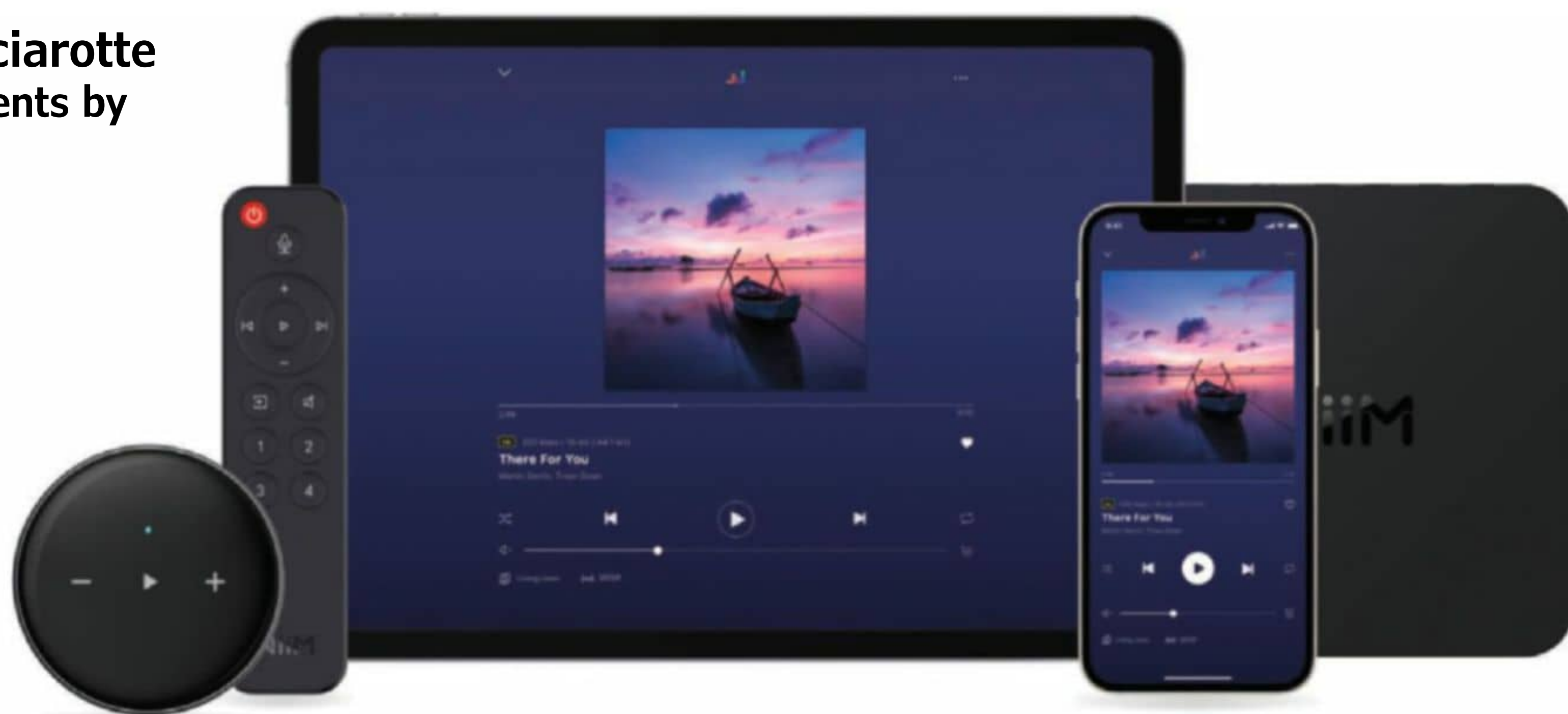
Rémi is currently working as Director of Product Management cooperating closely with business development and has a technical background with a PhD in Acoustic Measurement Methodology. Born and raised in France, Rémi Guastavino relocated to Sweden to study at the Royal Institute of Technology (KTH) where he earned an MSc in Engineering Physics and his PhD in acoustics. Rémi then moved to Denmark where he focused on user-centric acoustic solutions working in both technical and strategical positions. He started his industrial career path at Bruel & Kjaer in 2008 after which he changed to GRAS Acoustics in 2020. As director of product management at GRAS Sound & Vibration, Rémi Guastavino defines the new product strategy, which industries their products should be aimed at, and how they can be improved to address the needs of the customer. He has focused on both the technical and user-related aspects of next-generation acoustic transducers.

WiiM Pro Music Streamer

Smart Network Streaming System

Linkplay Technology is recognized for its widely adopted wireless audio streaming modules, smart home, and connected AI products. The new WiiM Pro is a versatile music streamer that combines all the latest technologies and protocols, making it the ultimate (and most affordable) home audio hub available. *audioxpress* received a pre-launch unit to review subjectively and objectively.

By
Oliver A. Masciarotte
with Measurements by
Stuart Yaniger



The WiiM Pro supports many popular music streaming services in the WiiM Home app, or users can connect their NAS for local files. A multiroom sound system can be created with existing AirPlay 2 speakers, Amazon Echo, and Google Home devices or using multiple WiiM Pros, WiM Minis, or LinkPlay-powered devices.

Around my house, a higher energy veneer overlays everything around us, an invisible one of modulated RF. It is this all but ubiquitous 2.4GHz and 5GHz WLAN network that Linkplay Technology taps into for its convenient and inexpensive WiiM Pro music streamer.

The WiiM Pro, a lightweight rectangle of black plastic that is smaller than a mac mini, is powered by the soon-to-be-universal USB-C standard. A bit surprising at this price, a USB-C wall wart is included in the package, as is a 1m POF interconnect for AES-3 optical and a two-channel RCA-to-RCA interconnect for analog. The brain of the product is a versatile and efficient Arm Cortex-A53 64-bit SoC with access to 512k each of DRAM and flash memory for buffered gapless playback.

WiiM Pro supports all common streaming audio standards and services, including Spotify, iHeartRadio, Tidal, Amazon Music, SoundCloud, Qobuz, Pandora, Radio Paradise, vTuner, Deezer, and Napster. Speaking of Apple and Android, configuration and operation for the WiiM Pro is mediated via a cross-platform app, WiiM Home, that acts as a full featured remote. It also supplies unified search across all supported music services. Wireline I/O

encompasses stereo line in and out on RCA, plus digital audio out on AES-3 optical and coax. A 12V trigger input allows for legacy amplifier control, and an RJ45 completes the mix for old skool wired networking. A multicolor LED supplies status feedback and capacitive transport control and volume buttons provide rudimentary command directly from the device. Incoming analog is digitized for broadcast at up to 96kHz, while digital from the AES inputs or the app is supported at up to 4x or 96k. Streaming protocols include AirPlay 2, Chromecast Audio, DLNA, Spotify Connect, TIDAL Connect, Amazon Music Casting, and Roon.

Setup

The WiiM Home app certainly is full featured and makes the setup experience quite painless. For this review, I received a pre-production model in non-branded packaging. The first WiiM Home version I installed on my ancient and trusty iPhone SE was like many other “convenience” apps for remote piloting a consumer appliance. It was a bit hard to use, there were gaps in the functionality, and some of the labeling and explanatory text was awkward or incomplete. I fired off a note to the folks at Linkplay

with my observations, and all my complaints were addressed in the next revision.

The WiiM Pro uses Linkplay’s A98MG SoM (System on Module), powered by an Amlogic A113X 1.5GHz quad core ARM processor, which supports IEEE 802.11 b/g/n/ac single and dual band (2.4GHz and 5GHz, A98), along with Bluetooth connectivity (BT5.0 with EDR and BLE). The built-in DAC featuring Texas Instrument’s high precision, low jitter Burr-Brown PCM5121 supports up to 384kHz/32-bit audio with a 106dB SNR and 0.005% THD+N. Along with 23 factory graphic EQ presets plus one custom memory register for a user setting, there is automatic and manual sync for A/V or multiroom installations. The WiiM ecosystem can achieve less than 150µs synchronization accuracy among physically separate speakers if the speakers are on the same wireless network.

Connected receivers or speakers may have different audio path latency due to different hardware, software, and DSP processing. To maintain the synchronization of each system, WiiM measures the audio path delay automatically. By leveraging a built-in microphone and proprietary algorithm, the entire audio path latency is measured and compensated automatically. The latency value is used to maintain audio sync in wireless multi-room playback with Apple HomePod, Amazon Echo, Google Home, and other WiiM devices. Manual audio compensation can also be dialed in for custom delay.

Individual EQ settings can be assigned to each source, such as Bass Boost for the analog in or Bluetooth. Though I have yet to personally see a need, the app does allow for voice command with Siri, Alexa, and Google’s Voice. The app has 12 music preset slots, allowing you to save your favorite station, playlist, album, etc., and access them with the preset button on the Pro or remote, or by using the app.

Although a printed user guide is included, the downloadable PDF manual states that “...WiiM Pro can be grouped with (Apple) Homepod, (Amazon) Echo, Google Home, AirPlay 2 enabled devices... Alexa-compatible devices, or another WiiM Mini (\$79) or Pro to stream music throughout your home. Or play different songs in different rooms. With its digital and analog inputs and outputs, WiiM Pro works with your existing stereo system.” The WiiM Pro also supports QPlay 2.0, which is used specifically in China by Tencent QQ Music. That’s where the “Q” comes from, and is an “improved” DLNA protocol, according to LinkPlay. I had no trouble modernizing my bedroom TV’s audio rig, an old Pioneer AVR and even more elderly a/d/s L200 speakers I had built up while I worked there.



The WiiM Pro turns any analog audio equipment into a smart, network connected system with Wi-Fi or Ethernet, allowing users to modernize their favorite equipment for a fraction of the cost of buying a new setup, while also enjoying hi-res audio sources.

The stereo replaces the super crappy in-built audio that’s part of our small Samsung smart TV. With WiiM Pro on duty, I was able to stream pleasantly EQ’d Qobuz while tidying up, reading in bed or as background noise while snoozing. Same goes for our ward Max, who currently has the only vinyl rig in the house. That setup consists of an ancient Onkyo receiver paired with another, larger pair of a/d/s speakers. In this instance, they are a set of custom three-ways employing hand-selected examples of their 800 series drivers and crossover. Both systems are capable and pleasant but lack the modern conveniences of streaming and Bluetooth access. WiiM to the rescue!

WiiM Pro also offers an endpoint for Bluetooth 5.1 streaming. LinkPlay stealthily powers another lifestyle audio fixture in my home, Cleer’s stylish



The WiiM Pro reflects the company’s efforts to combine all the latest and best audio technologies available in a single design that provides users with an easy and affordable way to connect existing audio systems.

About the Author

Oliver A. Masciarotte has spent more than 40 years immersed in tech, working on manufacturing, marketing, and product development for many pro and CE audio manufacturers including dbx, a/d/s, Lexicon, Sonic Solutions, and Minnetonka Audio. His client roster is as diverse as Apple, Harper Collins, NASA Johnson, NPR, and Universal. His writings include a book covering file-based music for the home and more than 100 articles for sundry trade publications. A member of the Audio Engineering Society (AES), Society of Motion Picture and Television Engineers (SMPTE), and Digital Cinema Society (DCS), he is currently co-founder and CXO of MAAT, a digital audio software manufacturer.





The WiiM Pro supports AirPlay 2 to connect to an Apple TV, Mac computers, or any Apple device, and also delivers an elegant and complete iOS and Android app for users to easily manage their entertainment systems all over the home. By connecting existing audio equipment to Wi-Fi or Ethernet, it brings a whole new life to home entertainment. That includes working with Siri, Alexa, and Google voice assistants to control music selection, volume, and playback. And from the app, users can adjust any part of the home system with 10-band digital equalization and a manual graphic EQ.

and good sounding CRESCENT smart speaker. Since the day I received it, I have been unable to convince CRESCENT to play from my preferred streaming service via Bluetooth. After way too many attempts at configuring Qobuz with the “aid” of Cleer’s tech support, I simply gave up using Bluetooth. During this WiiM review, the user manual suggested that, for the best AirPlay 2 experience, Apple users employ a simple HomeKit-based setup. After adding Apple’s Home app for iOS to my phone, CRESCENT miraculously appeared as a Bluetooth target. Frankly, I was amazed but, on reflection, really shouldn’t have been as Linkplay’s expertise in streaming audio software and hardware, provides turnkey solutions for big brands such as Harman, JBL, Yamaha, Marshall, and Edifier.



Small but extremely powerful, the WiiM Pro delivers all the options needed to listen to Tidal or Apple Music or any other music service anywhere in the house, also serving as the hub to a multiroom wireless system.

About WiiM Pro

LinkPlay CEO Lifeng Zhao stresses their focus on solutions, not just products. “Linkplay’s traditional business model is and will continue to be a turnkey solutions provider. The WiiM line of products are being developed to take advantage of the various technologies we deploy into solutions we can control and bring to market.”

Making it easy to use for everyone is very challenging. Being a network-first product, it always needs to work with the user’s network environment, regardless of configuration. “Ultimately, WiiM Pro’s performance will be heavily affected by the end user’s network environment.” Another aspect I appreciate is their Continuous Quality Improvement (CQI) approach to their offerings, such as periodic Over-the-Air (OTA) updates. “All products get updated based on our continued and never-ending push for the best UX in the market.” Amen.

At \$149, the WiiM Pro is neither the least nor most expensive entry-level streamer on the market. What does distinguish it from the competition is that the price is low while, overall, the UX and functionality are excellent. Linkplay designs its devices to interoperate seamlessly with a customer’s existing gear, bringing the best possible user experience. Linkplay’s design anticipates that, when things do go wrong, the product performs self-diagnostics and provides suggestions for the user to improve their network environment automatically.

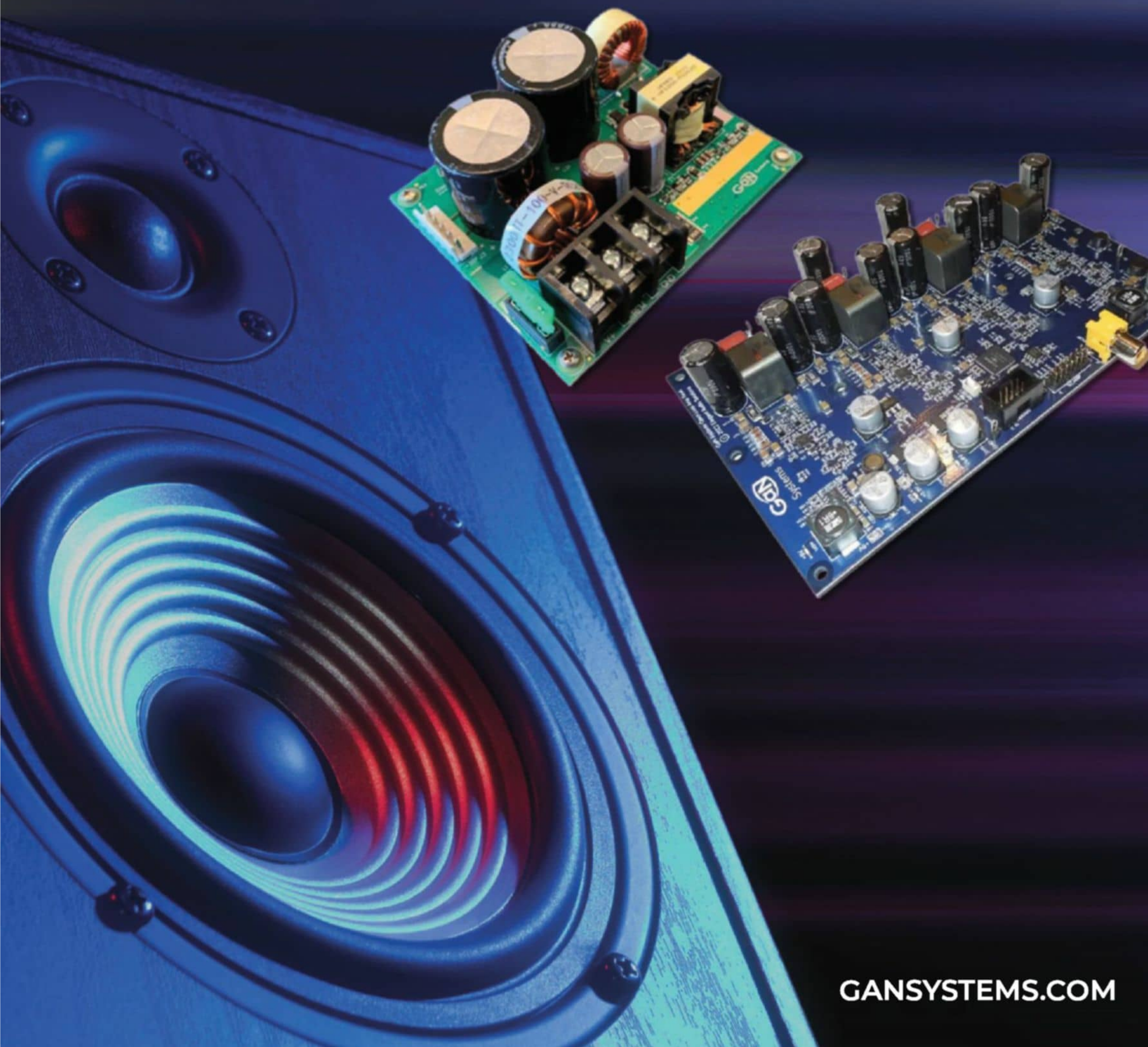
The folks at Linkplay have really done their homework to make what could be a frustrating techno nightmare simple and straightforward. The app even warns you when Wi-Fi signal strength is weak, politely suggesting you “...please keep it close to the router.” WiiM Pro even updates automatically, just after midnight, when connected to your Wi-Fi network.

For \$50 less, Linkplay offers the WiiM Mini, which lacks a few features including Ethernet, AES-3 optical in, and Google integration. Less onboard memory and a wimpier SoM contribute to the cost reductions. Still, WiiM Mini is an excellent and crazy cheap way to smarten up your audio game while its still low cost and feature-packed WiiM Pro big brother is an excellent way to either unify existing systems and smart speakers around your home or repurpose an old AVR, receiver or integrated amp plus speaker pair you probably have gathering dust.

Trust Linkplay to keep its products fresh. As the company says, “All the music services we support are always evolving...(along with) the feedback from users. We make continual changes to improve the user experience.” For more information, visit: <https://wiimhome.com/WiiMPro/Overview>.



LEADING CHANGE IN CLASS D AUDIO



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WiiM Pro Music Streamer — The Measurements

By
Stuart Yaniger

The kind folks at WiiM sent a WiiM Pro unit for me to run electrical tests in parallel with Oliver Masciarotte’s use and listening review. When I unpacked the unit, I was somewhat surprised at its light weight and, frankly, cheap feel. It’s visually attractive despite the all-plastic exterior, but it’s not audiophile eye-candy by any means. Still, there’s a lot of functionality per dollar here, so visual and tactile esthetics can be forgiven.

I installed the WiiM Home app on my iPhone, then used it to connect the WiiM Pro to my home network. The installation went exactly as described in the instructions and everything operated smoothly with my usual streaming source (Amazon HD) and my Network Attached Storage. The front touch panel switches were reliable and responsive. The streamer functionality in my system was flawless, and the app was intuitive and slick, easily the equal of other much more expensive systems I’ve used.

After using the WiiM Pro for a few weeks, I put it on the bench for basic measurements, which were a mixed bag. The data here was acquired using an Audio Precision APx525 analyzer.

Given my positive user experience, it was unsurprising that

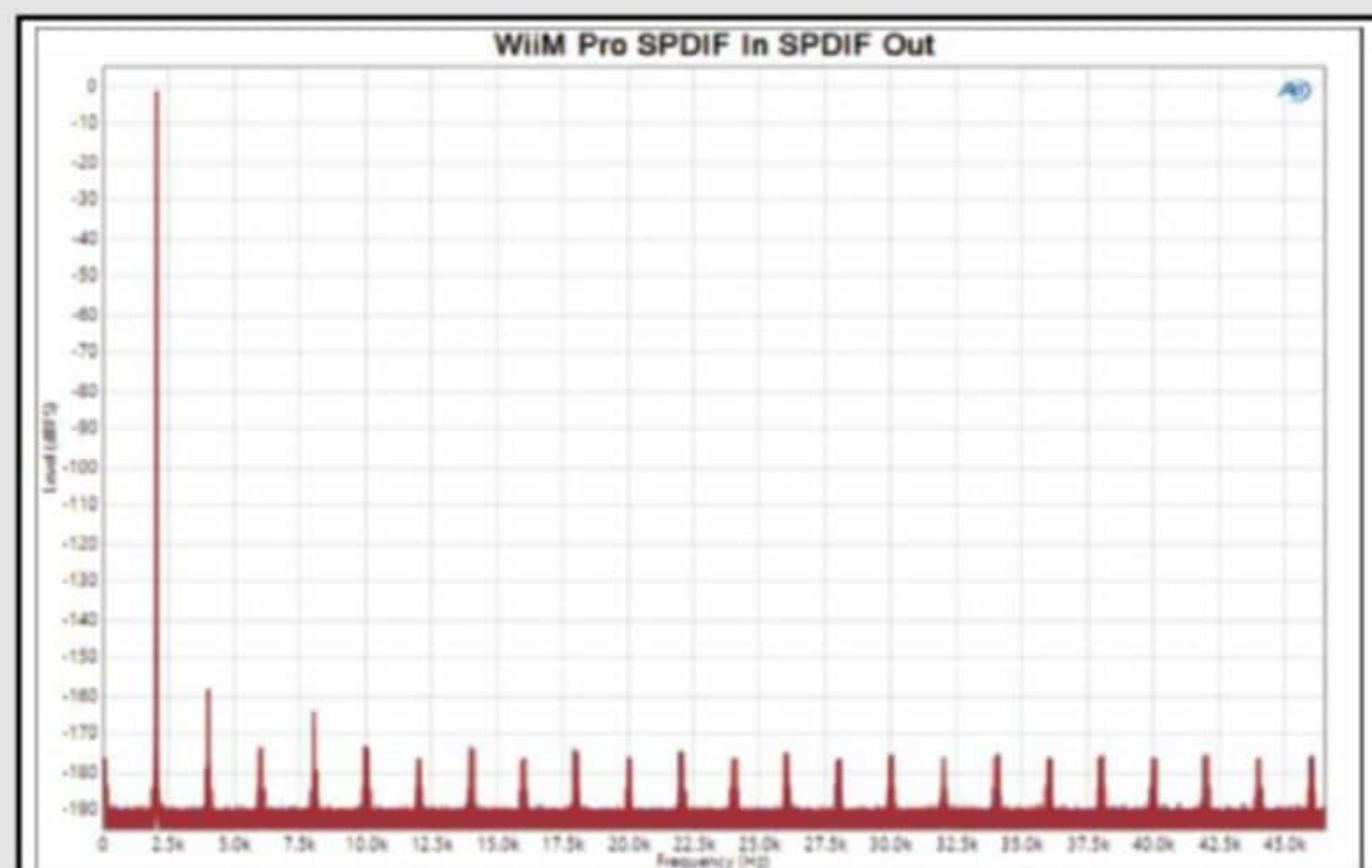


Figure 1: The spectrum of a 2kHz sine wave with digital input and output showed essentially perfect performance.

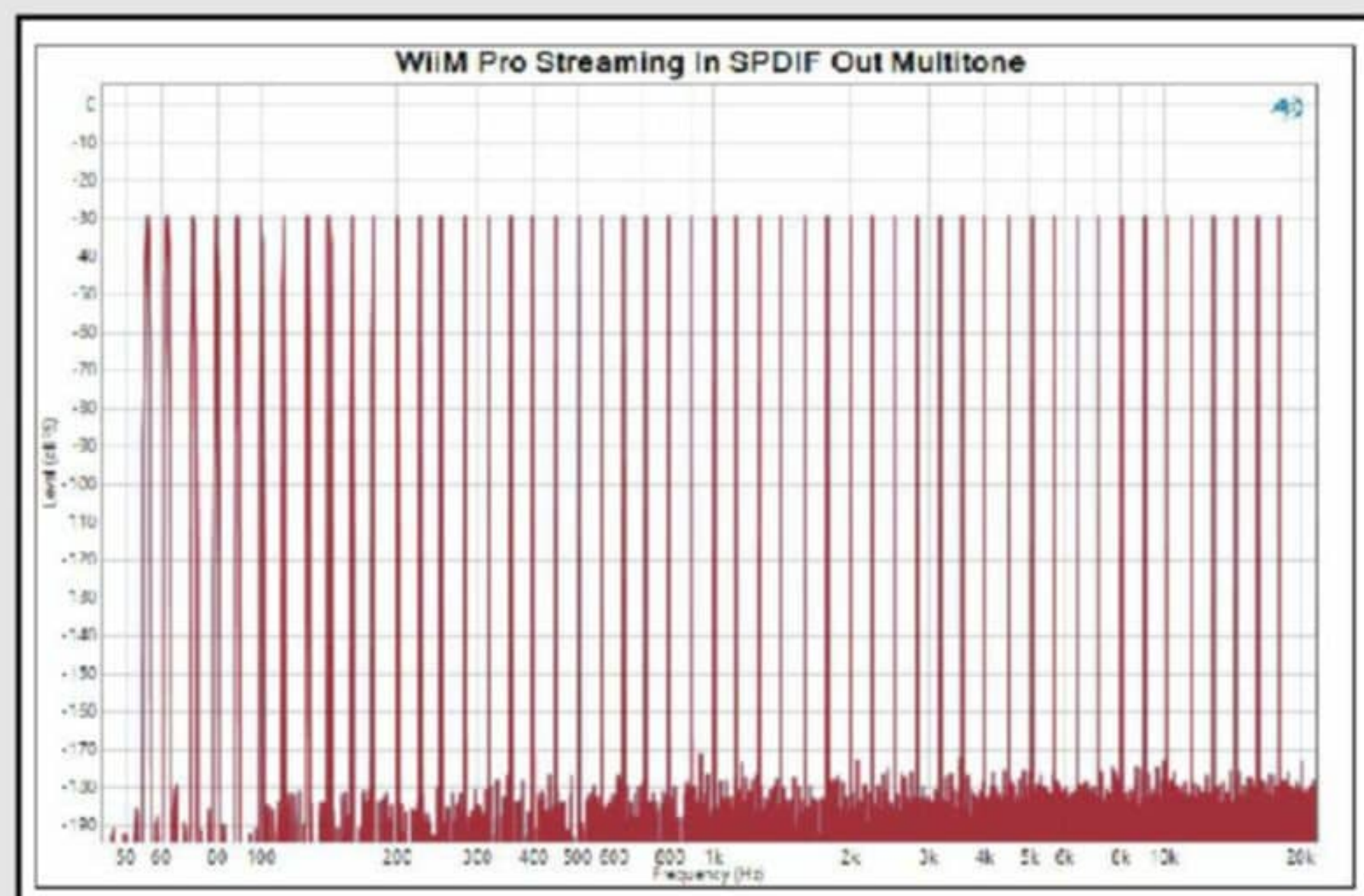


Figure 2: A multitone signal streamed from an NAS source gave a textbook digital output.

the SPDIF and streaming sections measured quite well. For SPDIF I/O, I used optical cable. The spectrum of a 2kHz sine wave at 0dBFS with both optical in and optical out is shown in **Figure 1**. The sample rate for this signal was 96k. Using the streaming function, I input a multitone signal and looked at the SPDIF output (**Figure 2**), which was essentially perfect both with respect to distortion and flatness of frequency response.

Moving to the analog outputs, a 0dBFS SPDIF input gave a 2V output from the line out, which may be adequate for directly driving some but not all power amplifiers. For this use case, the user has a choice between a higher gain power amp or inserting a preamp or gain stage between the WiiM Pro’s output and the power amp—this of course comes at a cost.

With an SPDIF input at 0dBFS and analog output with volume turned up to full, a 2kHz sine wave gives the spectrum shown in **Figure 3**. The most significant harmonic component is the second at -91dB in the worse channel and -96dB in the better

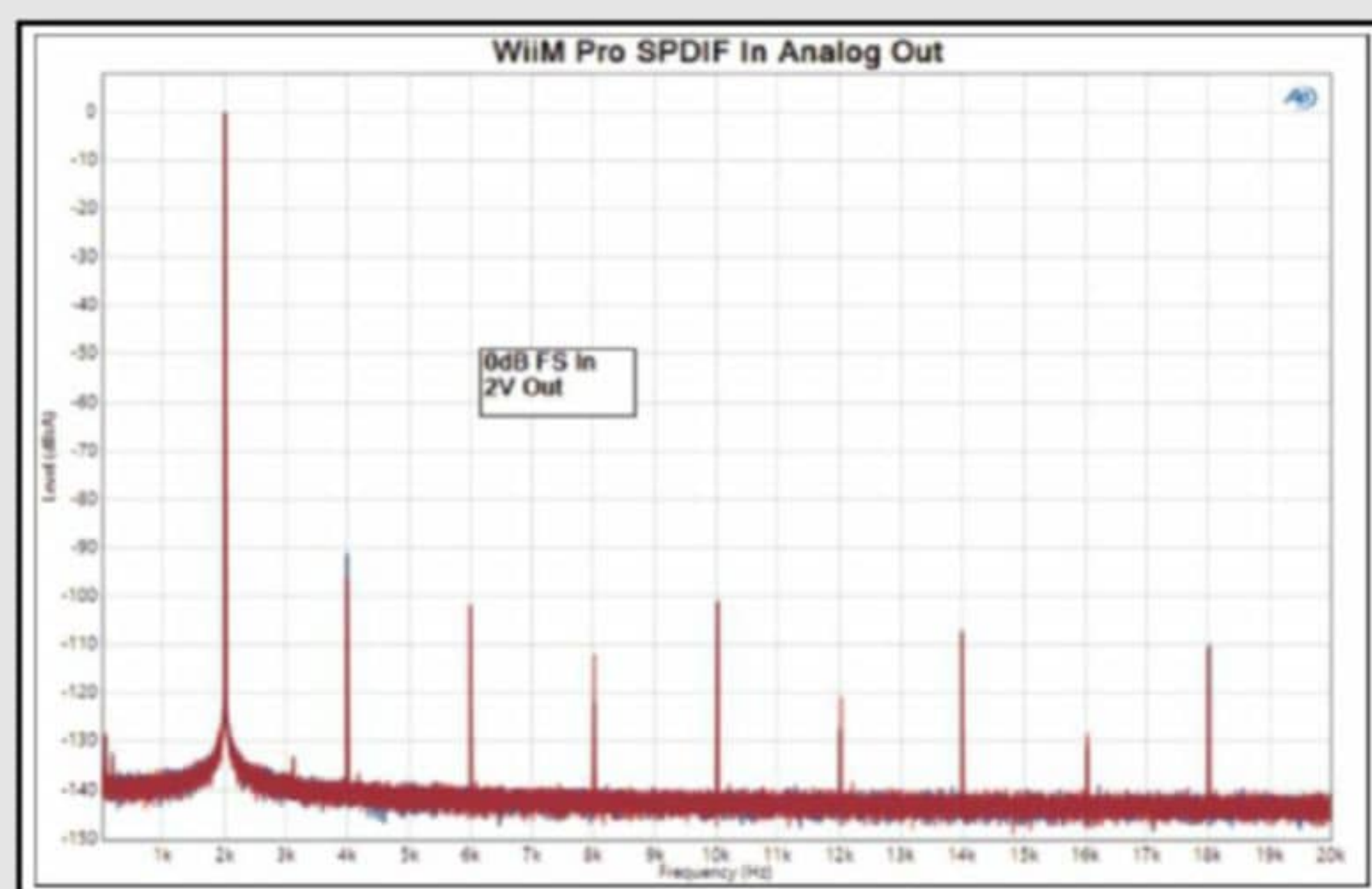


Figure 3: The spectrum of a 2kHz sine wave with SPDIF in and measured at the line outputs was relatively clean, with the dominant harmonic being second order at less than 0.002%.

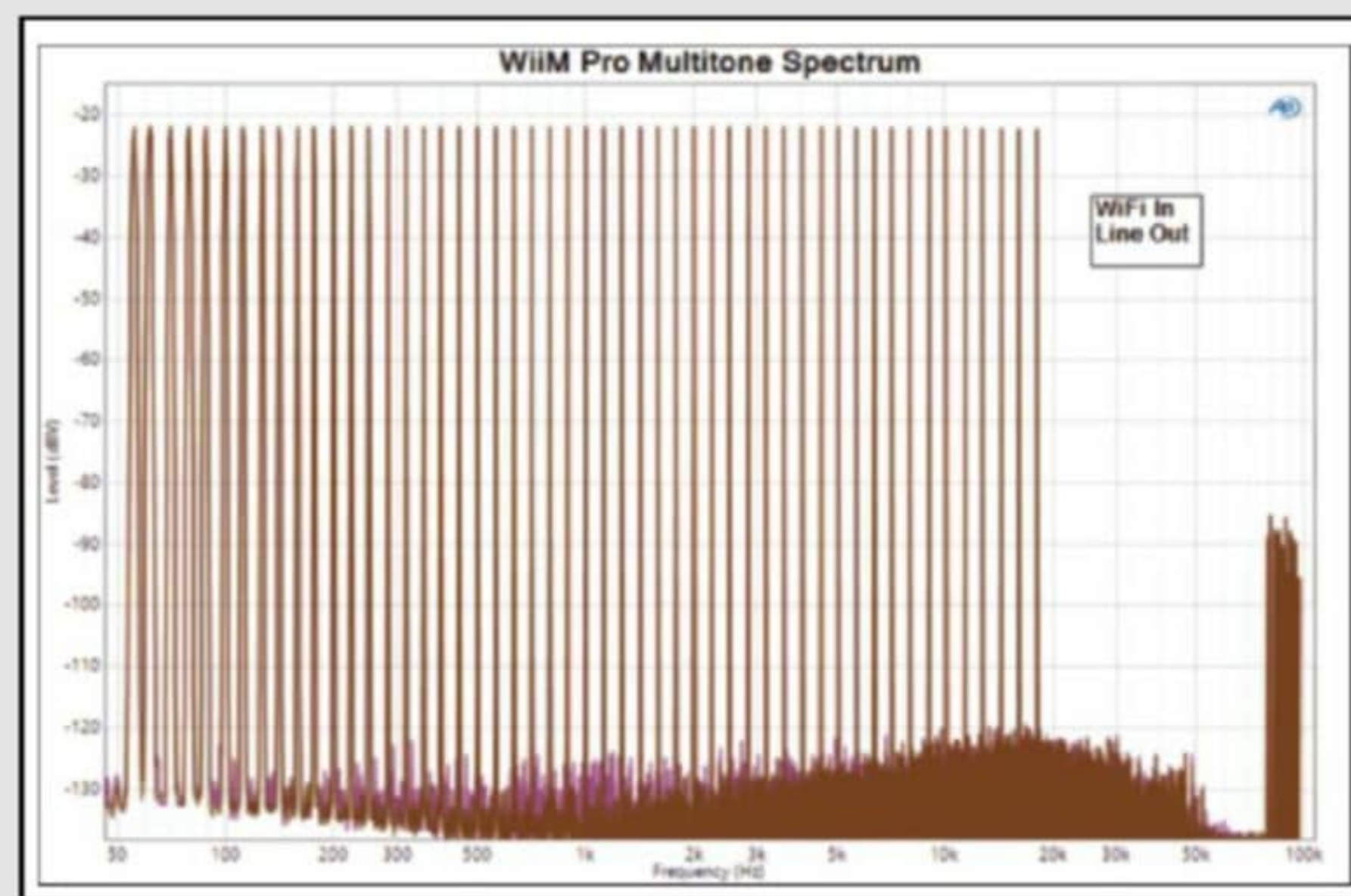


Figure 4: The WiiM Pro’s response to a multitone spectrum streamed from an NAS source and measured at the line (analog) outputs showed low distortion and flat frequency response with a bit of ultrasonic hash.

one. Higher-order harmonics remain below -100dB. This is not state-of-the-art performance, but more than adequate to achieve sonic transparency. A more complex multitone signal streamed to the WiiM Pro via WiFi gives the spectrum shown in **Figure 4**. Distortion in the audio band between the frequency components is below -120dB, which is quite good. Interestingly, the analog outputs show some ultrasonic hash near 100kHz at -85dB. This is not likely to cause any issues in actual use.

Figure 5 shows the dependence of distortion on frequency with optical SPDIF input and line level analog output, not state of the art for 2023 but is low enough to cause any audible issues.

Other than low output voltage, the performance as a streamer or with SPDIF inputs is very good and the WiiM Pro will not be the audibly limiting factor for any system using it in this manner.

All is not as delightful using the analog inputs. Gain from Line In to Line Out measured slightly less than 6dB at the maximum volume setting. **Figure 6** shows distortion versus input level; the WiiM Pro goes into severe clipping at input levels above 1V, corresponding to output levels of slightly under 2V. This will be a problem with most modern sources like DACs, which will typically output 2V to 4V at full scale. If a source has a variable output, it needs to be turned down accordingly to avoid overload. For sources with fixed

outputs, overload and clipping may very well be a problem unless a pad or attenuator is added to the signal path.

Another issue with the analog I/O is a little more difficult to fix and can be seen in **Figure 7**—there is an interchannel phase error, which is actually a constant time delay of about 25µs. This does not happen with the SPDIF or streaming inputs (**Figure 8**), and it was something I didn’t hear in my use of the WiiM Pro as a streaming device, so it’s clearly caused by something in the ADC section. In theory, the user can compensate for this flaw by moving one speaker a few inches closer, but there’s really no reason that something like this should be seen in a product in the modern era.

Overall Thoughts

From the measurements, I would be happy to use the WiiM Pro with digital sources or streaming, where its performance ranged from very good to excellent. My only caveat there would be the somewhat anemic 2V maximum from the analog line output. The issues with the analog inputs are concerning enough that I would be hesitant to use it in those scenarios. Considering the very good performance with digital inputs, this was a bit surprising and disappointing. 🤦

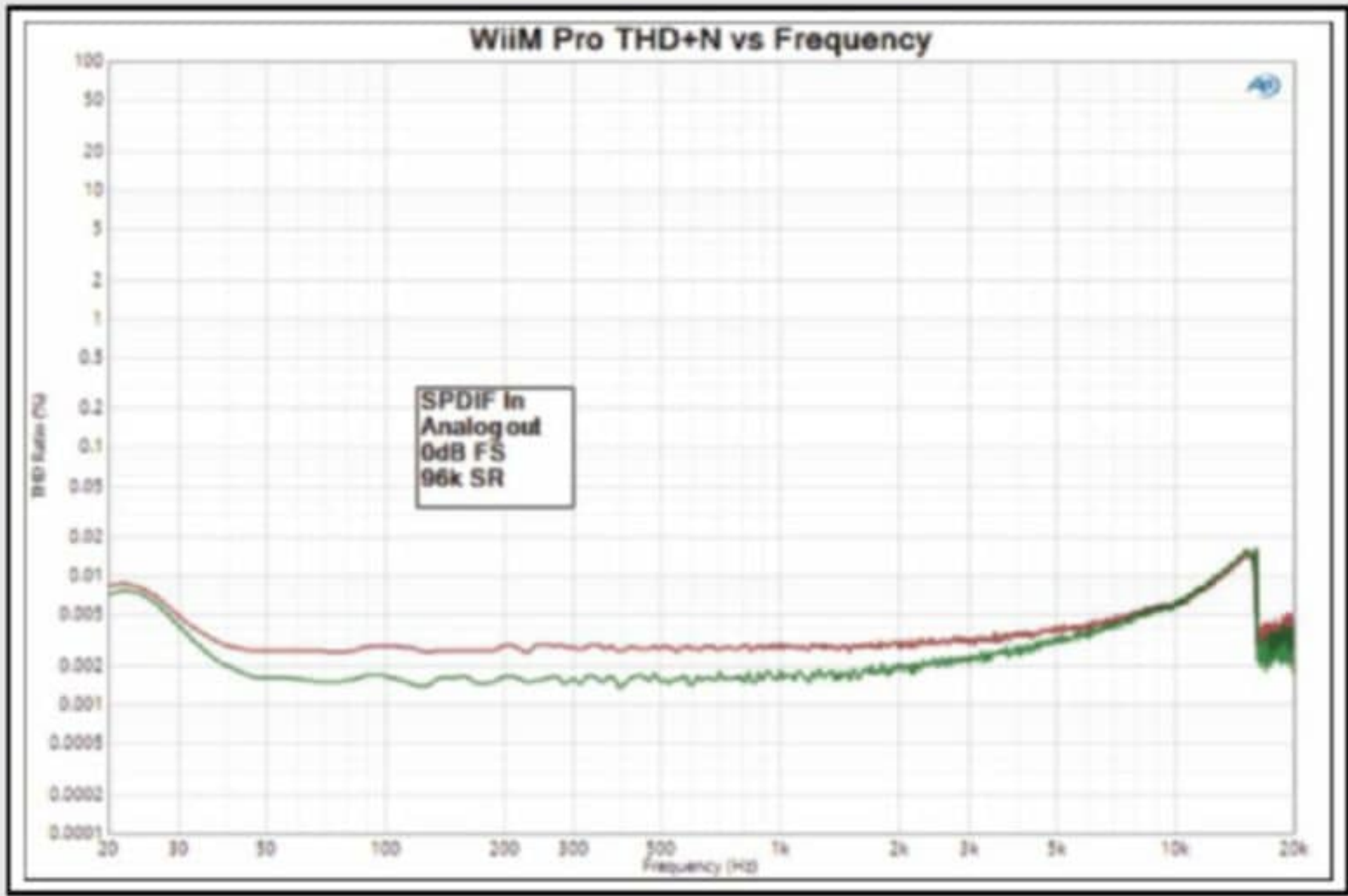


Figure 5: With an SPDIF input, the Line (analog) output maintained relatively low distortion across the audio band.

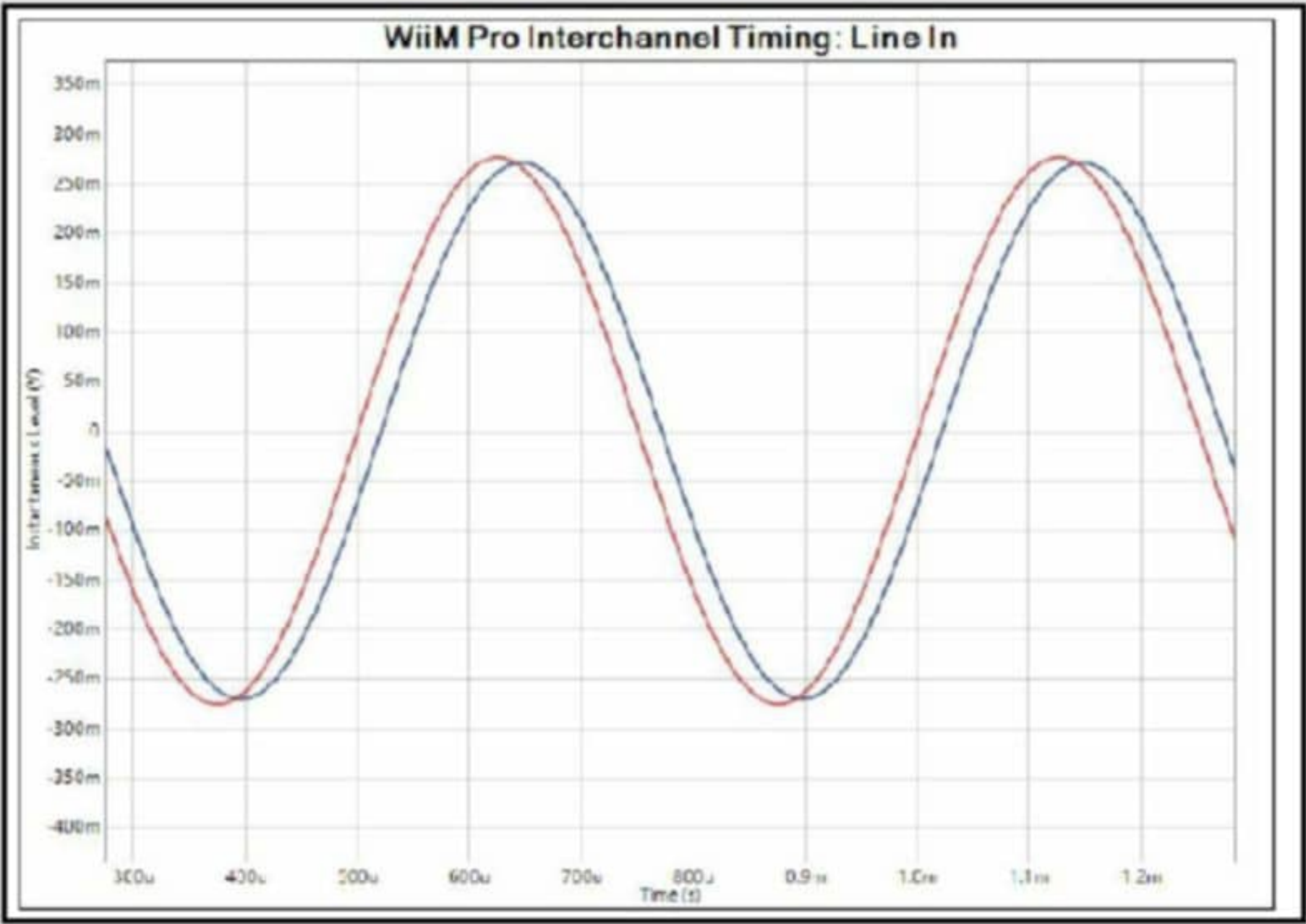


Figure 7: The analog Line In input had a timing error between the two channels.

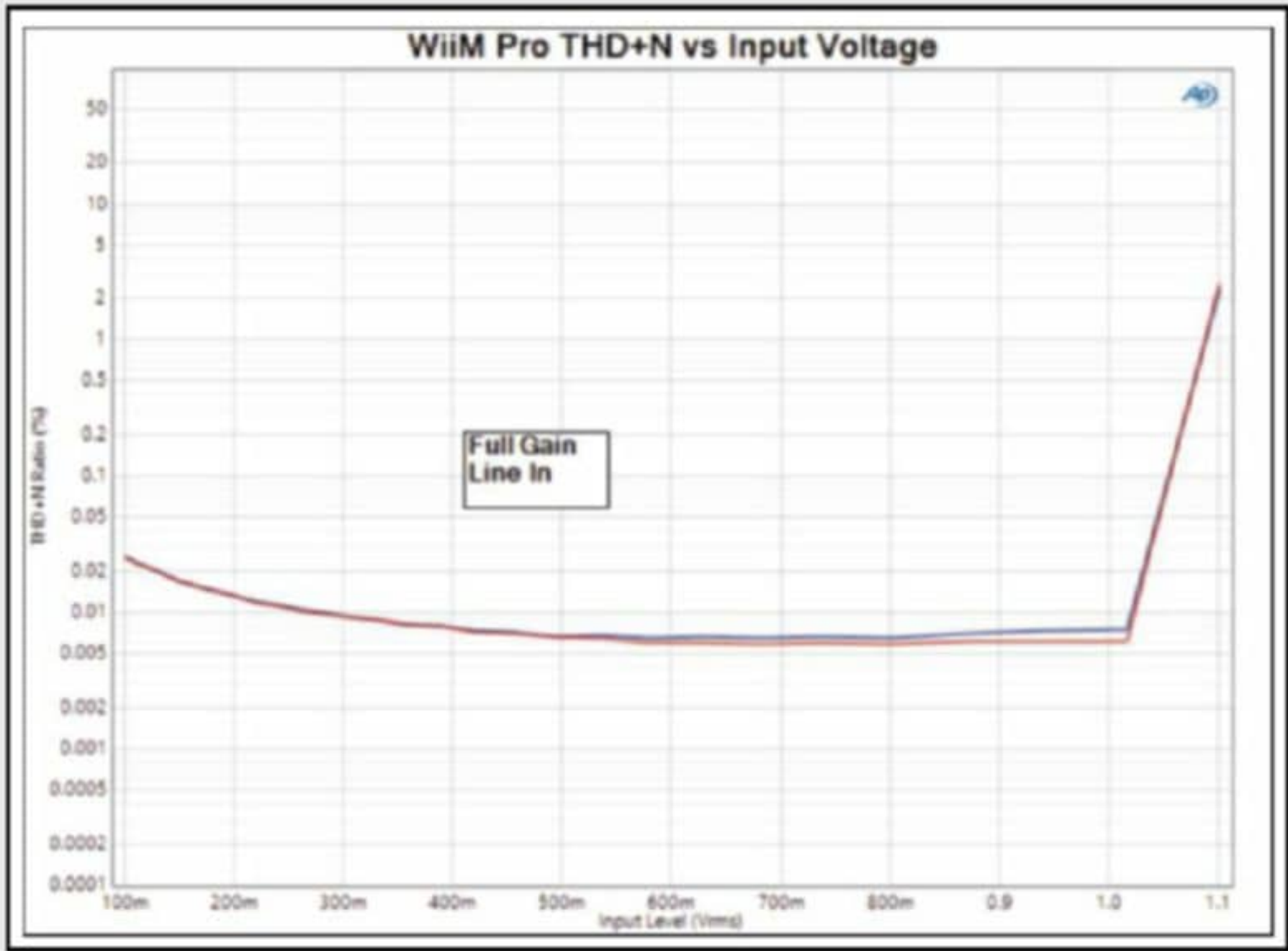


Figure 6: The WiiM Pro’s Line (analog) input showed sharp clipping at 1V.

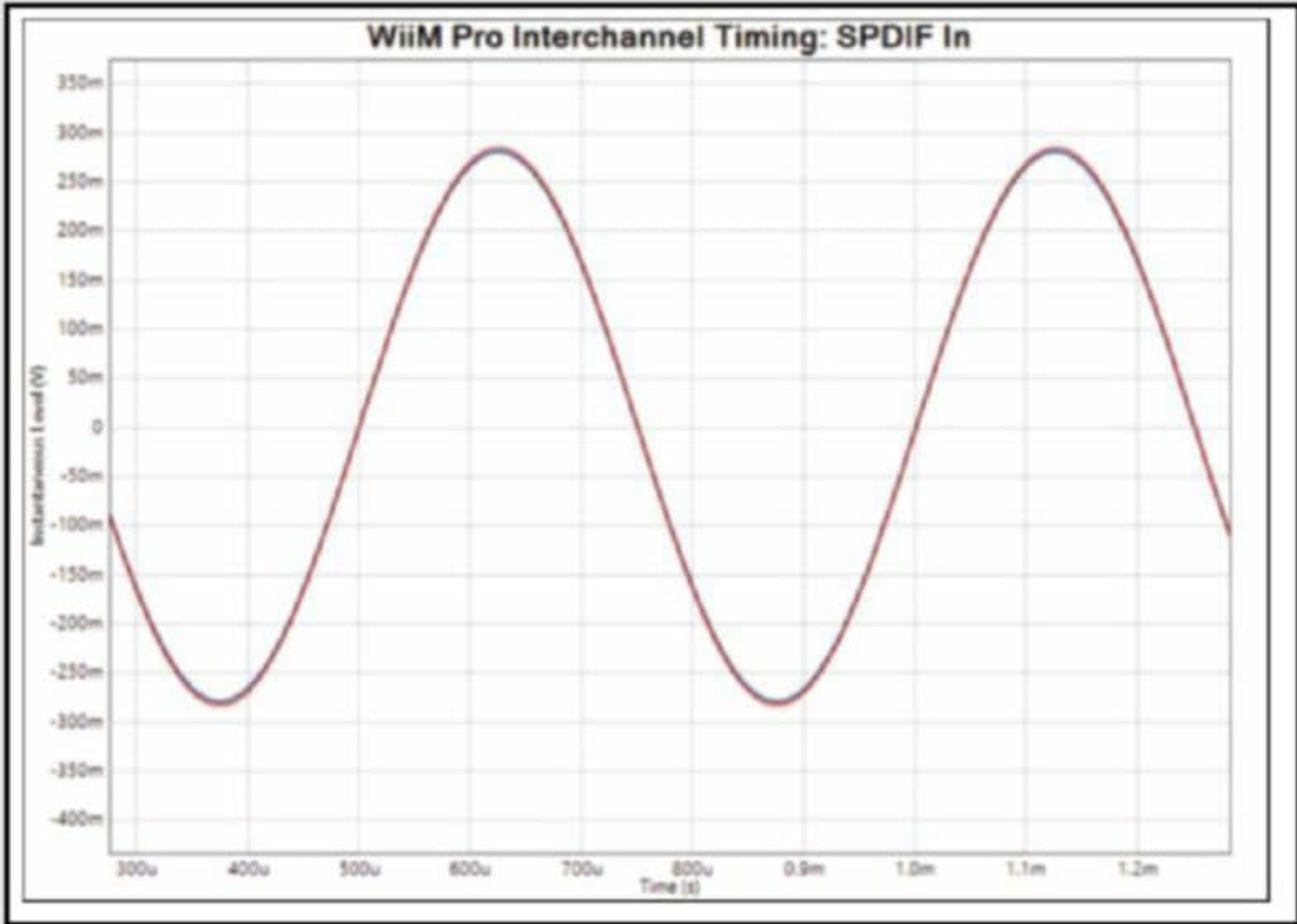


Figure 8: Interchannel timing differences are not an issue with SPDIF or streaming inputs.

Room Gain

Room acoustics is often discussed in the hi-fi community, since the room where you put your loudspeakers will heavily influence the sound that reaches your ears, both at low and high frequencies. In these discussions the term “room gain” sometimes comes up. This article was inspired by Erik Wiederholtz (CTO of Perlisten Audio) and his interest in the topic, and he was kind enough to provide measurements and photos from his own listening room for a sealed subwoofer as well as for a vented subwoofer.

By
René Christensen

Room gain is not a term used in acoustics textbooks as such, it lives in the world of hi-fi without a clear definition. Sometimes it describes how the sound field from a loudspeaker is generally affected by the room, but it is seemingly often associated specifically with the sound pressure in a room around and below the first mode of the room, and that is the definition we will explore here.

What is observed in some measurements is an in-room sound pressure that is not easily explained by the anechoic response, as in the room the level extends to much lower frequencies than the characteristic frequency of the loudspeaker. This effect is a surprise to many people, and since it is not much explored in the literature, it will be approached here in steps.

Boundary Gain

Let us first build some intuition about different aspects related to sound sources in constrained spaces. Imagine first a monopole sound source S radiating into free space. For a given volume velocity the resulting sound pressure can be calculated or measured, and a certain sound power is radiated from the source. Next, imagine the source placed up against it in an infinitely large hard boundary, where hard means that it is not moving and perfectly reflects all sound. An equivalent situation can be represented by removing the wall and adding a mirror source on

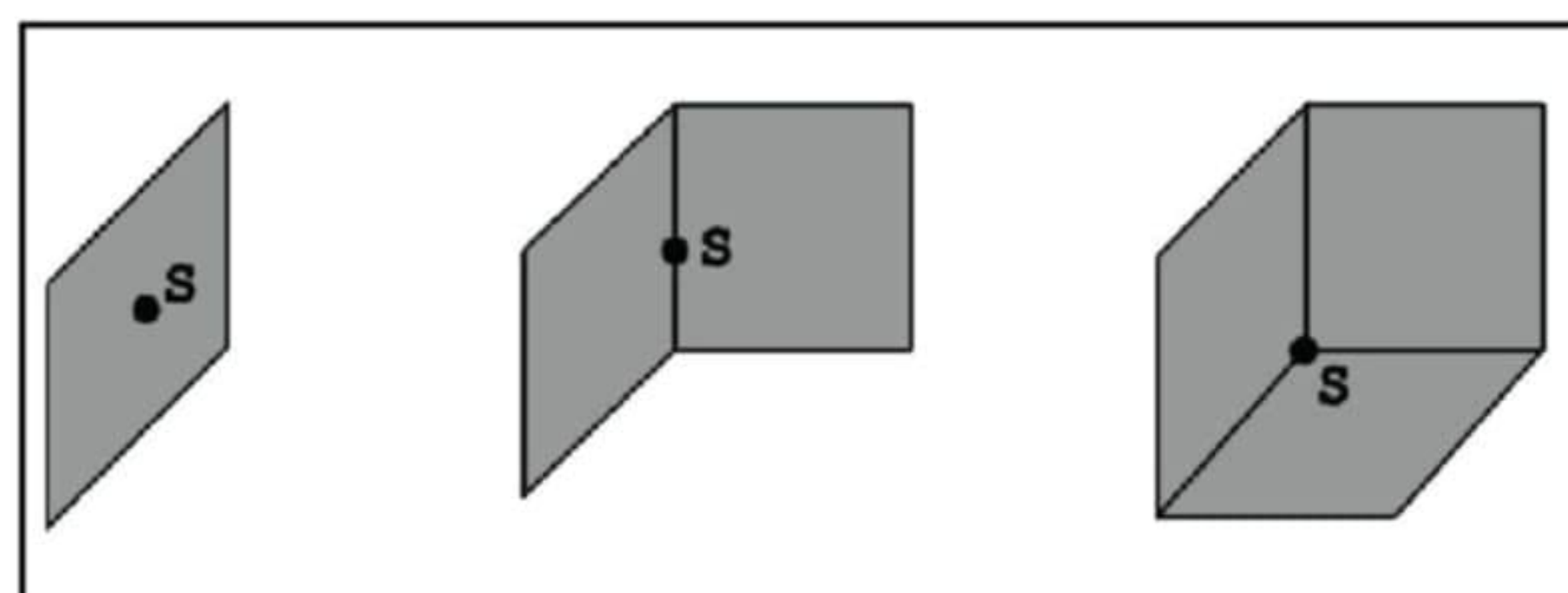


Figure 1: Three infinite boundary situations causing a boundary gain of 2, 4, and 8, respectively, for a source (S).



top of the original source with the same source strength, and so the sound pressure is effectively doubled as the two sources are correlated. Boundary gain thus describes how restricting the solid angle that a sound source radiates into will increase the acoustic output. In **Figure 1**, three typical boundary gain situations are depicted. There is an underlying assumption that the source is omni-directional, so above a certain frequency where the source becomes more directional, the effect will be less relevant.

You will sometimes see a switch on the back of some equipment where you can toggle between situations regarding the placement near a wall, and it will apply appropriate filters to lower the gain at lower frequencies because of this boundary gain. In some instances, the filters may even compensate somewhat for the speaker-boundary interferences response, which is our next topic.

Speaker-Boundary Interference Response

When the source is not placed directly at the infinite boundary but instead at a distance from the boundary, there are also reflections to consider, oftentimes categorized as speaker-boundary interference response (SBIR) effects in the hi-fi community. The distance from the source to the boundary as well as the frequency will affect the sound pressure in any point. The mirror source analogy can be used again to view the reflection as an added source as shown in **Figure 2**, and so for all frequencies one must sum over

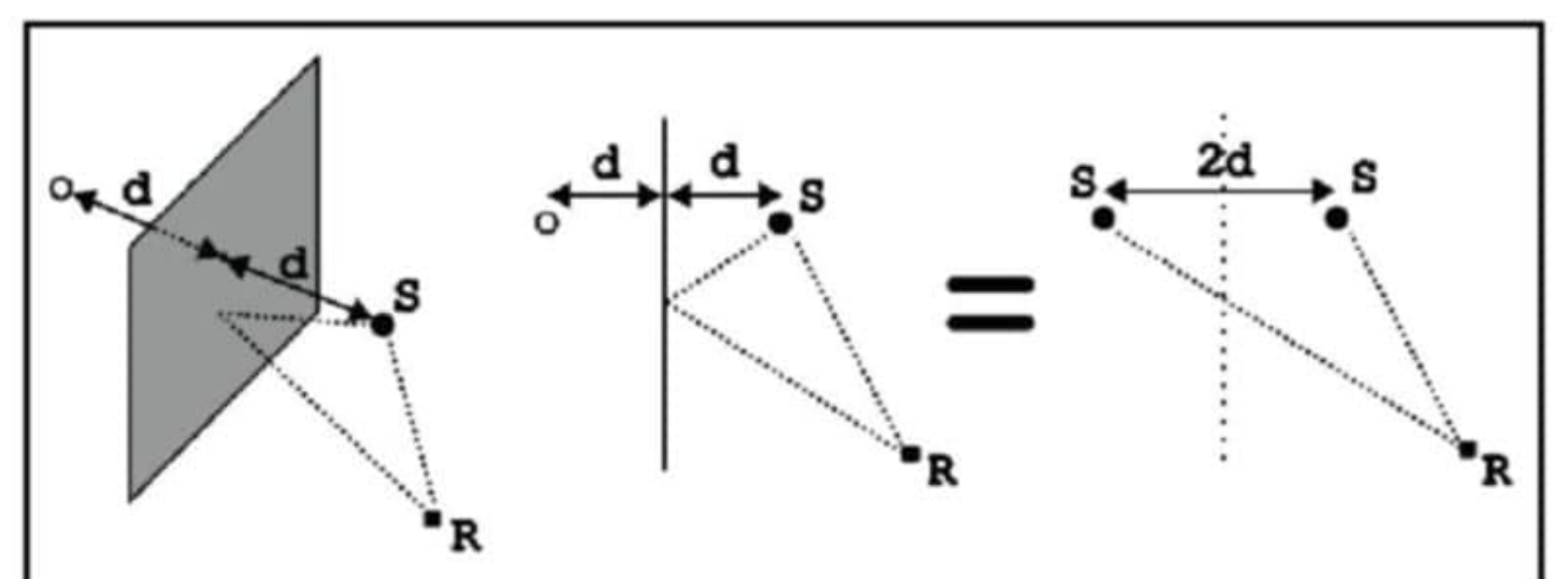


Figure 2: A single source (S) near a boundary is equivalent to two sources resulting in the same sound field at a receiver point (R).

the two sources causing constructive and destructive interference. For example, for a frequency with a wavelength of a quarter of the distance d from the source to the wall, there will be half a wavelength extra distance from the mirror source to a listening position on-axis, and so there will be a complete cancellation for the listener at this frequency, assuming hard walls.

If there are several boundaries present, there are several first reflections, and if more than first reflections are to be considered then mirror sources of mirror sources must be included. Again, the directivity of the source is important, and this directivity will depend on the frequency.

Modal Behavior

When there are enough boundaries to form a closed room, modal aspects must be considered. While the mirror analogy still holds, it becomes inconvenient to work with in practice. Instead, a modal approach is utilized. A mode, or "eigenmode," consists of two characteristics: An eigenfrequency and an eigenfunction. The eigenfrequency is the (angular) frequency associated with the mode, and the eigenfunction expresses the shape of the sound field associated with the mode. For a cuboid shaped room with hard walls the eigenfrequencies can be calculated [1]:

$$\omega_{n_x, n_y, n_z} = c_0 k_{n_x, n_y, n_z} = c_0 \sqrt{\left(\frac{n_x \pi}{l_x}\right)^2 + \left(\frac{n_y \pi}{l_y}\right)^2 + \left(\frac{n_z \pi}{l_z}\right)^2}$$

where l_x , l_y , and l_z are the dimensions of the room in question, with sound speed c_0 and indices n_x , n_y , and n_z each from zero to infinity. Each eigenfunction is found via:

$$\Psi_{n_x, n_y, n_z}(x, y, z) = \Lambda_{n_x, n_y, n_z} = \cos\left(\frac{n_x \pi x}{l_x}\right) \cos\left(\frac{n_y \pi y}{l_y}\right) \cos\left(\frac{n_z \pi z}{l_z}\right)$$

with a scaling factor of:

$$\Lambda_{n_x, n_y, n_z} = \sqrt{\epsilon_{n_x} \epsilon_{n_y} \epsilon_{n_z}}; \epsilon = \begin{cases} 1 & \text{if } n_i = 0 \\ 2 & \text{if } n_i \neq 0 \end{cases}$$

The modes are calculated from the geometry of the room, the boundary conditions, and the fluid properties of the medium. The latter two are typically "hard wall" and "air" as the medium.

There will be infinitely many modes at discrete frequencies. At higher frequencies they are spaced so closely that other approaches than the modal should be applied to study the sound field, whereas below the so-called Schröder frequency the modal approach described here is convenient.

What is most important to understand when it comes to modal behavior is that modes exist independent of any source. That is how they are calculated, both analytically and numerically using Finite Element Analysis (FEA) as there is no source entered anywhere in the eigenmode setup. When sources are present in a room, all the modes have the potential to be excited, each to some degree, and so the total sound field will depend on the source type, source placement, and listener/receiver placement. In general, the sound

field in a room can be described via the pressure phasor, but it is not trivial to establish the complex coefficient A for a given situation:

$$\rho_{n_x, n_y, n_z}(x, y, z, \omega) = \sum_{n_x=0}^{\infty} \sum_{n_y=0}^{\infty} \sum_{n_z=0}^{\infty} A_{n_x, n_y, n_z}(\omega) \Lambda_{n_x, n_y, n_z} \cos\left(\frac{n_x \pi x}{l_x}\right) \cos\left(\frac{n_y \pi y}{l_y}\right) \cos\left(\frac{n_z \pi z}{l_z}\right)$$

For certain known sources, and source positions, one can calculate the sound pressure explicitly for any listening position of interest as we will discuss now.

In-Room Frequency response

For a sound source (S) placed in the room and a receiver (R) placed in the listening position, the resulting complex sound pressure can be calculated either via numerical methods or via the analytical expression:

$$\rho(R) = \frac{ic^2 Q \rho_0}{V} \sum_m \frac{\Lambda_m^2 \Psi_m(S) \Psi_m(R)}{\left(\omega - \frac{\omega_m^2}{\omega}\right) - i \frac{\omega_m}{\omega \tau_m}}$$

With m being a general index over all modes, and τ_m being a time constant to include a (small) amount of damping for each mode. As superposition holds, one can calculate the sound field from multiple subwoofers with each of their characteristics (gain, phase, characteristic frequency, quality factor Q , sealed, or ported) placed at different locations in the room by summing across several instances of the equation for a fixed listening position. This way one can also calculate dipole and quadrupole responses by combining monopoles with certain distances and phases.

Note that while the frequency response is calculated using the room modes, it is not in general possible to extract all modes from looking at a single frequency response.

Certain modes may not be significantly excited due to the placement of the source(s) and/or the receiver could be placed so that the effects of certain modes are not seen. Also note that any applicable boundary gain and SBIR effects are included in the modal calculations.

The Zeroth Mode

As we are mainly interested in understanding why there is a gain at very low frequencies, we first realize that there is a "zeroth mode" at 0Hz falling out of the above analysis. We can calculate the sound field resulting from a monopole source with volume velocity Q at very low frequencies by truncating the previous equation to index zero only:

$$\rho = \frac{ic^2 \rho_0 Q}{V} \frac{1}{\omega} = \left(\frac{\rho_0 c^2}{i \omega V}\right) Q$$

We recognize the fraction in parentheses as the acoustic impedance of a closed volume that simply acts as an acoustic compliance. The resulting pressure from a constant volume velocity source Q will hence be higher and higher for lower and lower frequencies. We now need to investigate the output from different loudspeaker types and how they relate to the above.



Photo 1: The subwoofer is shown in the lower left corner of the room with an unrelated placement of the microphone.

Loudspeaker Types

The final piece of the puzzle is to look at the displacement behavior of the low-frequency driver(s) in the loudspeaker for different loudspeakers; primarily the sealed enclosure and the vented enclosure with one or more ports. In general, a loudspeaker will be designed to output a constant acceleration as a function of frequency, as this will result in a constant sound pressure level (SPL) as a function of frequency. But this is in its normal operating frequency range above its resonance



Photo 2: The microphone placed in the upper right corner of the room

frequency. For a standard electrodynamic driver in an enclosure, there will be a constant cone acceleration above the characteristic frequency of the loudspeaker, but below it will tend toward a different characteristic, depending on the enclosure type. For a sealed enclosure this will be a constant displacement at low frequencies, which is proportional to a velocity that decreases with frequency. For a vented loudspeaker, however, the displacement is not constant at low frequencies, but instead has a second-order slope corresponding to the fourth-order roll-off in its anechoic pressure.

Room Gain

We can now see where the observed room gain originates. For a sealed subwoofer, the displacement is constant as a function of frequency below the subwoofer's characteristic frequency. So the downward slope of the acoustic impedance of the room will be countered by the velocity sloping upward with first order, resulting in a flat sound pressure response from the subwoofer below its characteristic frequency. This is what has been discussed on many hi-fi forums, namely that the room helps retain a constant sound pressure level below the subwoofer's own roll-off. The reason is simply that there is a zeroth mode at 0Hz, and although sound is associated with oscillation, there still is the same resonance phenomenon toward low frequencies as there is near any other excited modes.

Think of a syringe blocked on one end with a plunger in the other end. The static pressure change will be proportional to the displacement of the plunger, and this will hold dynamically also for an oscillatory displacement below the first axial eigenmode in the syringe. For a vented subwoofer the output from the rear of the driver(s) will short-circuit the front output, and so less and less pressure can be built toward 0Hz. The roll-off for the vented subwoofer will not be the fourth order as observed under anechoic conditions, but instead second order in-room. The syringe analogy can be used again, only now there is a small hole in the barrel, preventing quasi-static pressure. The room is assumed airtight, which is not generally the case, and so the measured results will of course vary from the above ideal case. The calculated results will now be compared to measurements.

Measurements vs. Simulations

Erik Wiederholtz provided several measurements for his room, but in this article we will limit ourselves to a sealed and a vented subwoofer, respectively, placed in the front lower left corner, for a

5.7m x 4.62m x 2.67m room as shown in **Photo 1**, with the microphone placed in the opposite rear upper right corner as shown in **Photo 2**.

The simulation software COMSOL Multiphysics was used for calculating SPLs, but only because we have also been exploring the effect of adjacent rooms and leakages. The analytical expression given earlier will fully suffice in recreating the responses shown here.

The sealed subwoofer was simulated via its anechoic transfer function with a Q of 0.71 and a characteristic frequency of 33Hz. In **Figure 3** the analytical results show the constant pressure at frequencies below the subwoofer's natural roll off, and the measurements follow the same trend at low frequencies. There are no losses in the simulation, and so the resonances are very pronounced, whereas the measured results are for a room with furniture in it, and it also has a large opening into an adjacent room, and so the two responses are naturally different.

Figure 4 shows a vented subwoofer via its corresponding anechoic transfer function with a Q of 0.71 and a characteristic frequency of 25Hz. The simulated response is seen to roll off with a second-order slope toward low frequencies, but the measured response rolls off with a fourth-order slope. This is due to a subsonic high-pass filter setting in from 17Hz and below, which was not included in the analytical expression.

Conclusion

The elusive term “room gain” has been described via a modal approach recognizing that for an airtight room there is a “zeroth mode” that affects the very low frequency response of a loudspeaker, so that its in-room response will not match its anechoic response.

The traditional modal aspects from room acoustics do indeed capture the room gain effect, and it will show up both using the standard room acoustic equations and in numerical calculations. The room acts as an acoustic compliance, so a lumped analogy can be used for arriving at the same conclusion of the pressure being constant for a constant displacement from a sealed subwoofer at low frequencies. As rooms are not in general airtight, the effect will be seen to various degrees, but it has at least been explained now how it relates to the traditional room acoustics nomenclature. 

Author Acknowledgements: Thanks to Erik Wiederholtz (CTO of Perlisten Audio) for some fruitful discussions and for delivering in-room measurements and photos of the room.

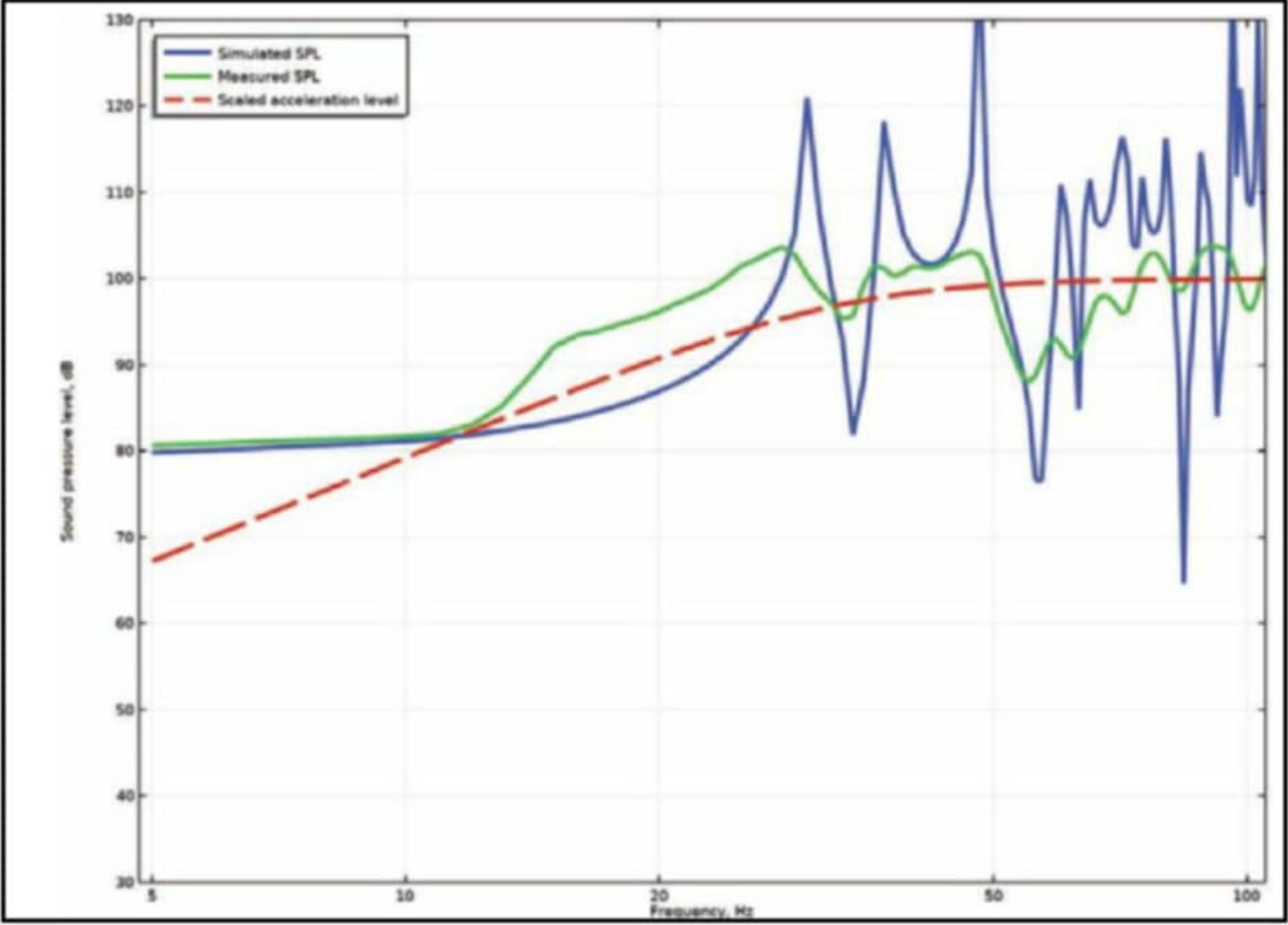


Figure 3: Simulated (blue) and measured (green) sound pressure levels for the sealed subwoofer, with scaled acceleration level (red).

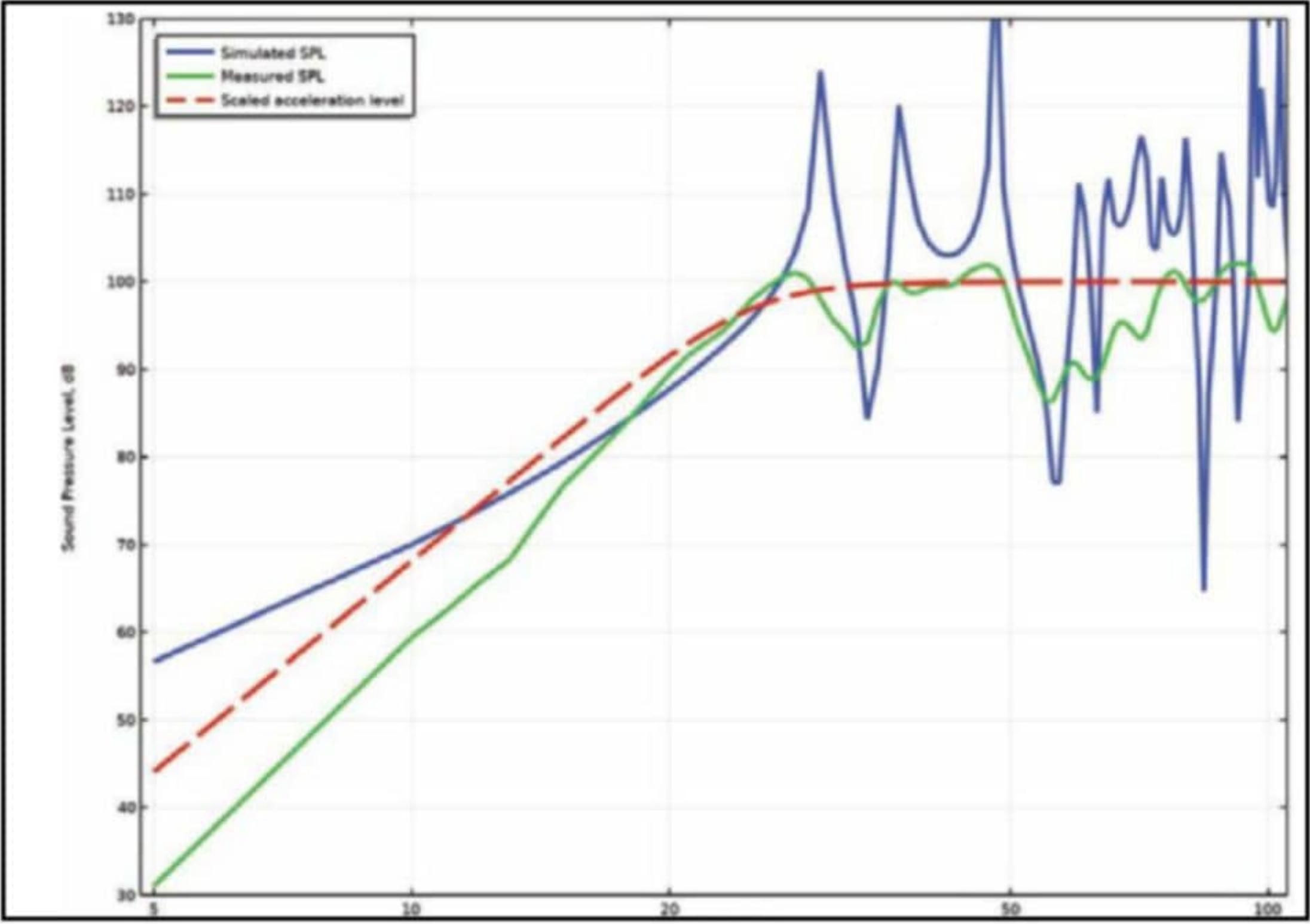


Figure 4: Simulated (blue) and measured (green) sound pressure levels for the vented subwoofer, with scaled acceleration level (red).

About the Author

René Christensen (BSEE, MSc, PhD) has been working with simulations in the loudspeaker and hearing aid industry for several years, and in 2021 he started his own company, Acculation ApS, with a focus on mathematical modelling of many different products, such as transducers. The overall aim is to use academic research results for engineering better solutions faster, training engineers in this type of virtual prototyping, and working with students across the world to progress the field.



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Acoustical Measurement and DSP Application in Automotive Audio

Frequency Bands, Sound Philosophy, and NVH

By
Philipp Paul Klose



This article is a continuation from the series “Acoustical Measurement and DSP Application in Automotive Audio,” with previous installments released in *audioXpress* (March 2023 and June 2023). While the first two articles concentrated on measuring audio signals in the cabin and the basic setup of the DSP modules in the processing pipeline, this article will go into the philosophy, fine tuning of the parameters, and noise, vibration, and harshness (NVH) issues in car cabins.

Interior Acoustic Effects

Pressure Chamber Effect and Room Modes

The relatively small volume of car cabins, which is somewhere in the range of 2.5 m³ to 4.5m³ has two effects on the playback of low frequencies. The low frequencies are boosted due to the high pressure created by the loudspeaker in the sealed environment of the car, which is called pressure chamber effect. The dimensions of the car’s periphery add a lot and dense room modes, causing the frequency response to resemble a mountain ridge instead of an elegantly curved math-function. The geometry of the formed sheet metal is made with properties in mind primed for industrialization, safety, and aerodynamics, leaving the acoustics often (but not always) in a state where they are adopted and designed around certain boundary conditions: If you cannot isolate the sound from a noise-making part, make it at least sound good. The high level of material complexity and forms makes it very time consuming and knowledge intensive to calculate the acoustic properties of the cabin with finite-element and similar numerical methods, let alone conventional geometric acoustic methods. This makes prediction in the design phase often time intensive, inconclusive, or simply not viable.

To give you an impression on the severity of the effect of the cabin acoustics on a loudspeaker’s performance, **Figure 1** compares a mid-bass driver

measured in an anechoic chamber against the same speaker mounted in a car door and measured in the interior.

Reflections

Without glass in the car, it would be very windy, or it would be hard to see anything at all. So, unfortunately, we cannot get rid of it. But we can work around it. Glass and hard materials create strong reflections in the sound field. These often reflect in measurement in the mid and high frequencies as dips and peaks, even with highly smoothed plots. Fine EQing can be used to tame problems here.

House Curves in Automotive

Flat and linear are relative terms in automotive. During the tuning process you will encounter weird and bedazzled frequency responses. That is normal and the feeling for working and interpreting graphs will set in with time through increasing experience.

To give you an overview about the problematic areas in the frequency domain, I have averaged the benchmark measurements from 10 cars with premium sound systems of different makes. Probably to your surprise, this curve is far from being flat and linear.

Figure 2 shows that the overall frequency response is tilted and loses on average approximately 0.75dBr per octave. This is good practice in cars and

helps with ear fatigue over long listening periods and overall gives a modern and powerful sound impression. More bass-heavy cars tilt stronger with ~1.5dBr, while more “tame” audio systems slope in the range of ~0.5dBr. The overall shape of the target curve goes back to Marco Binelli and Angelo Farina’s work for the loudspeaker manufacturer ASK [1] and was also described by Floyd Toole [2].

The bass area below 150Hz is over-boosted above the ±3dBr tube going to up to +6dBr above the average at approximately 60Hz. The general shape of this curve “resembles” the noise curve of the car itself and is also used as a tool, masking the car’s noise while driving. During driving, this bass boost often is even more exaggerated by a speed-dependent shelving filter while sometimes also interior microphone signals are used to counteract the increasing background noise with higher speeds through level adaptation in the DSP.

Frequency Bands and Their Challenges
Figure 3 shows the frequency bands and their approximate coverage from different speaker types.

(0) Sub-bass

The lowest reproducible frequency in the car cabin is determined by the diameter and excursion capabilities of your subwoofer and woofer chassis. Premium factory built-in systems are designed to go down to about 25Hz, standard (series) sound about to 38Hz (f-3dBr).

The lower, the better: Bass perception is for laymen as for sound enthusiast an important indicator for perceived quality. Besides the mid-bass band, this is one of the most EQ’ed frequency ranges in the whole audio spectrum.

It has been proven that several sources for low frequencies placed in different locations in the car give the best overall performance. Premium sound systems place several woofers in the doors, trunk, floor, and firewall area to spread out the acoustic work and aim for an even sound field with little variation over seat positions. Practically speaking, it is better for the overall quality impression in the cabin to have two 8” woofers and one 10” subwoofer with a proper mono bass tuning, instead of having all low frequencies handled by one big 12” subwoofer in the trunk.

(1) Bass

Car cabins and bass... This is per se not a match made in heaven. The small size of the cabin makes it hard to get an even and good bass performance: Room modes let the unoptimized frequency response look like a ridge line, such as the effect seen in the sub-bass band. The pressure chamber effect boosts

the complete bass region here like a shelving filter. Some EQ is needed to lower the peaks and even out the dips. On the other hand, too much EQ can seriously affect the impulse response and make quickly for a dull ‘over tuned’ sound impression. You want to aim for a smoothed-out frequency response here and spend time on the details.

(2) Mid-bass and Low-Mids

This is without a doubt the most difficult range to tune. Most EQ bands are needed here. Our ear draws

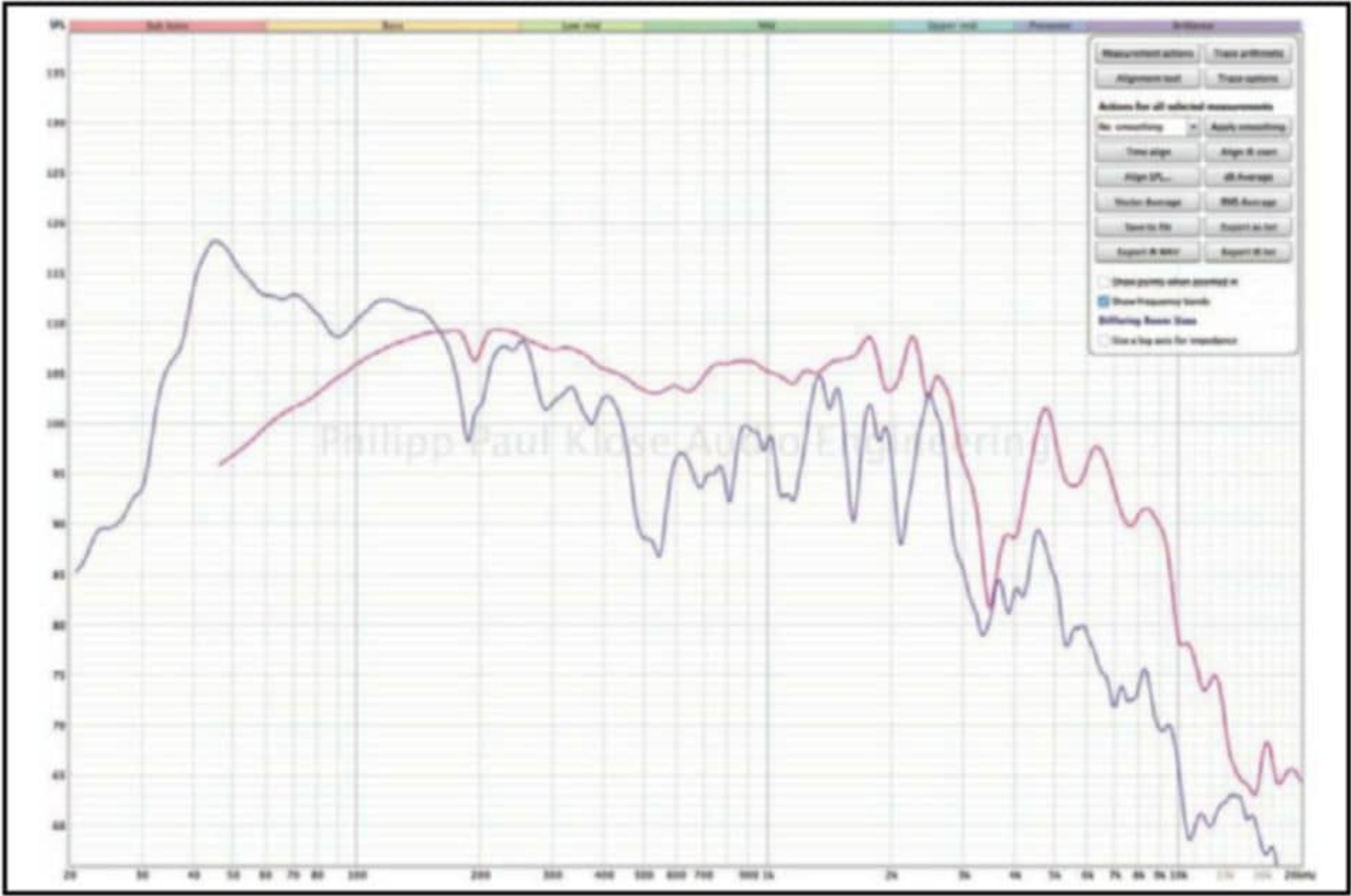


Figure 1: Midbass door driver measured in an anechoic environment (pink) and installed in a car cabin. In the car, the bass frequency regions are massively boosted due to the pressure chamber effect. The very dense room modes change the spectral performance in all frequency bands in various hardly predictable ways.



Figure 2: Average of 10 benchmark measurements of premium sound systems in cars (violet). The green lines mark ±3dBr from the blue line showing the overall average. The spectrum tilts approximately 0.75dBr per octave.

a lot of information from this frequency range since many instruments have their fundamental frequencies in this area and critical overtones that are used by the ear to discern the instruments characteristics and extract their individual melodic information. The quality of frequency response in the mid-bass

is another important factor with more experienced listeners that influences the critical acclamation.

Clean timing and phase relations in the mid-bass between the delayed speakers are necessary against acoustic cancellations.

A rather unpleasant effect in this area are null-points. In null-points neither positive nor negative EQs do have an effect. It can be compared to an acoustic black hole. Everything is sucked in. In extreme cases it can be necessary to restructure a tuning strategically around this affected area! [3]

(3) Low-Mids and Mids

In the low-mids and mids, the side effects acoustic measures can be found. Acoustic dampening material that is used to keep road noise out of the car, also affects the interior entertainment sound. This can be seen in Figure 3, as a dipping area between 350Hz and 600Hz.

Although mid-bass woofers often have the capabilities to directly cross into tweeters, they can feel breathless and stressed in this area in cars. If budget and space allow it, add midrange speakers in the doors and dashboard for clarity and relief of the mid-bass woofers. It adds to the spatial resolution. Especially midrange center speakers have an enormous stabilizing effect of the stage. Null points can also occur here, so be aware of them.

(4) Upper-Mids

In the upper-mids we are above the Schroeder frequency (room modes are random here). Most cars are easy to tune in this area. But, due to the high position dependency of the SPL in the cabin, multipoint measurements and array measurements are a must to make good tuning decisions here. In this frequency band, the best overall linearity can be achieved.

(5) Presence and High Ear Sensitivity Area

The human ear is the most sensitive in this area. Due to the individual size of pinna and ear canal, every human being has a slightly different frequency response, in-ear resonance, and perception in this frequency band. If you do the car tuning for yourself, it makes sense to adjust it to your personal liking. If you tune for a series car or for someone else, make sure to verify your impression with measurements.

Often a certain "trademark" sound associated with a brand or tuner is realized through adaptations in this range. Since every audio brand tries to distinguish itself from one another, here is one of the places where they do so. So do not be surprised if you measure irregularities in this area. Dampening a little here can enhance spatial perception and elevate the stage upwards away from the door pockets. Be



Figure 3: Challenging frequency bands and their approximate coverage from different speaker types.

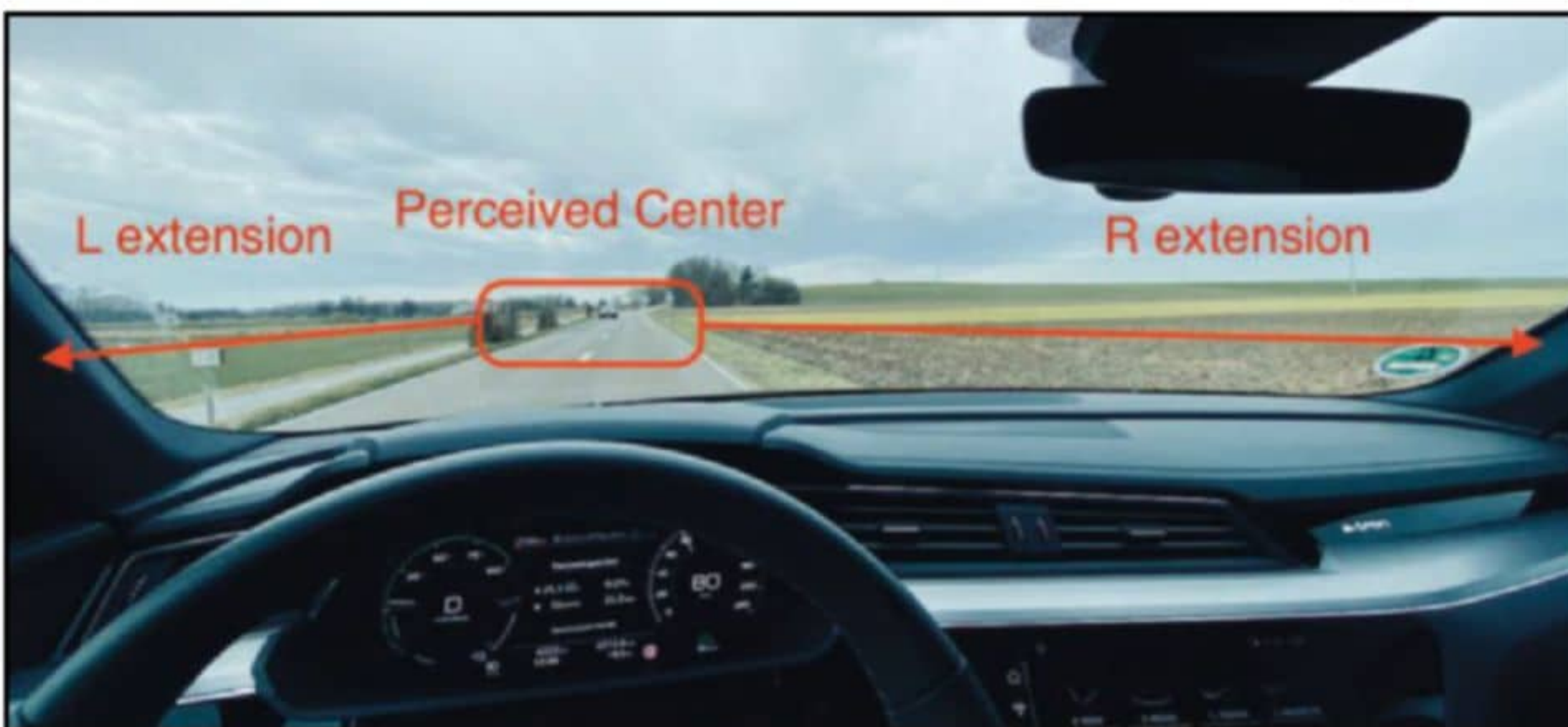


Figure 4: This is an example of driver-oriented tuning style.



Figure 5: Here is an example of middle of the dashboard tuning style.

aware that when you apply taste tuning, you might be interfering with musician’s and engineer’s vision in sound reproduction.

(6) Highs

The high frequencies above 5kHz hold challenges with tweeter resonances and off-center use of speakers. When you are using hard-dome tweeters or broadband speakers, it is likely that you will encounter strong resonances here, that make the sound harsh and edgy. This can often be countered effectively and easily with EQ application.

Often tweeters are in close vicinity of reflecting surfaces. The resulting loss of high frequencies must be compensated with high-frequency boosts, broad combing effects from glass surfaces can be countered with slight boosts.

Staging and Width
Driver-Oriented Style

The staging of the played back content in the car is not per se created “naturally,” through the physical positioning of the speakers or design of the crossover, but rather by tuning design decisions in the DSP. There are two predominant philosophies when it comes to staging.

The driver-oriented style of imaging could also be called “right behind the steering wheel” (**Figure 4**). It can often be found with

European car manufacturers. The phantom center of the stage is perceived directly in front of the driver and passenger. This sound is created in its core by delaying all sources to the driver

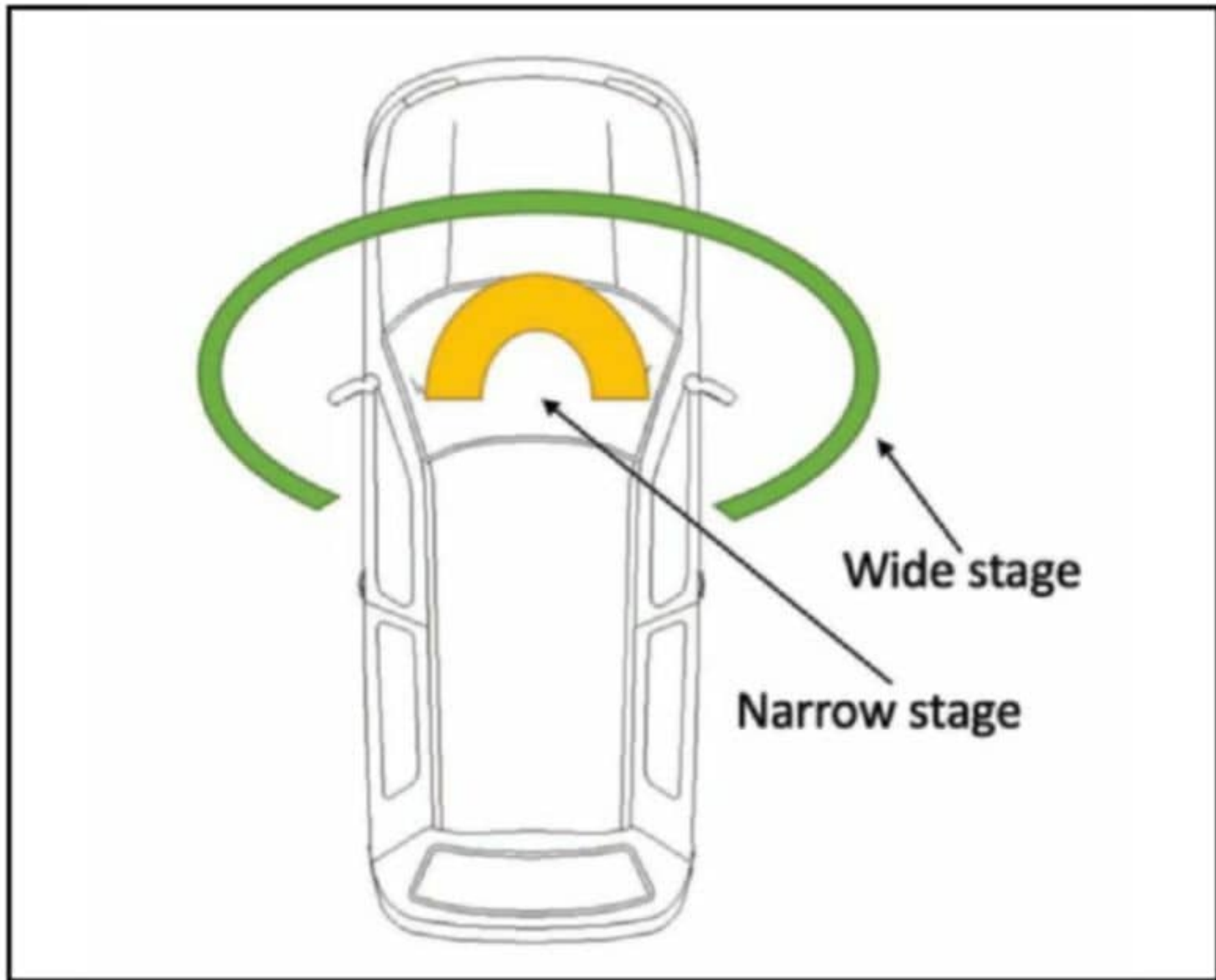


Figure 6: Using the image of this car, we can show the perceived width of a wide and narrow stage.





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position in the car. The imaging is automatically mirrored on the passenger side.

Middle of the Dashboard Style

The “middle of the dashboard” style can often be found in US and some Asian makes. The phantom center of the stage is perceived in the middle of the dashboard (**Figure 5**). This sound is created in its core by delaying all sources to the middle of the car. To shift a skewed imaging, delay and level compensation must be used. For example, to achieve a shift toward the driver side for 5cm (2”), add a delay of 0.66ms and +.4dBr of level to the L signal and dampen the R signal by -0.4dBr. If you would rather go in the other direction flip the algebraic signs and the delay to the R channel. For further shifts, linearly scale the values up.

Width of Imaging

A decision that is completely based on your personal taste and independent from the front imaging style is the width of the stereo image in

the car cabin. The choice palette ranges “from A-pillar to A-pillar” to “from B-pillar to B-Pillar”. In general, it can be said, that very wide stages require support through rear fills, surround speaker, and center speakers (**Figure 6**). If these speakers are not present, a narrower staging is the better choice. If you have plenty of speakers, you can widen your image more easily. Through a center speaker the image in front can be effectively stabilized.

Using polarity flips, mid/side processing, phase shifts, and allpass filters, the perceived width of the stereo image can be changed.

Noise of the Car

Driving Noise

To get the impression of how much and in which frequency region a car produces noise, I have recorded an upper middle-class SUV of German origin with an inline six-cylinder Diesel engine with an artificial head at 30km/h, 80km/h, and 130km/h and analyzed the recordings in the frequency domain (**Figure 7**). During the measurement runs, the entertainment system and air conditioning were switched off, so only engine, powertrain, tire, and wind noise was recorded.

The energy is mostly contained in the low frequency domain and drops sharply to higher frequencies. The broadband dip centered at around 500Hz can be attributed to the acoustic dampening foam used as measure to improve the ride comfort. The slope of this curve is a relevant factor for the form of the previously shown house curves.

Quite significant amounts of noise are produced by the wind, tires, powertrain, and chassis, peaking between 57dB SPL and 74dB SPL. Since there is (often) nothing we can easily do about this boundary condition, we must accept it and respect it in the DSP parameter creation and work our way around it. The mitigation and shaping of this noise are done by the NVH engineers in cooperation with the engineers for the used component parts.

Since the noise is static, only slightly modulated and often not of tonal nature, it is not perceived as disturbing. With electric drivetrains the situation changes not fundamentally. Electric cars have a quieter drive train, but their relative high weight squashes the tires more and creates other noises in the same band where the powertrain noises were. Wind and chassis noise are similar with electric and internal combustion engine cars.

Buzz, Squeak, Rattle, and Hum

Noise sources can also be found in the interior. Since these acoustic events are often temporary and within high-sensitivity regions of our hearing, our

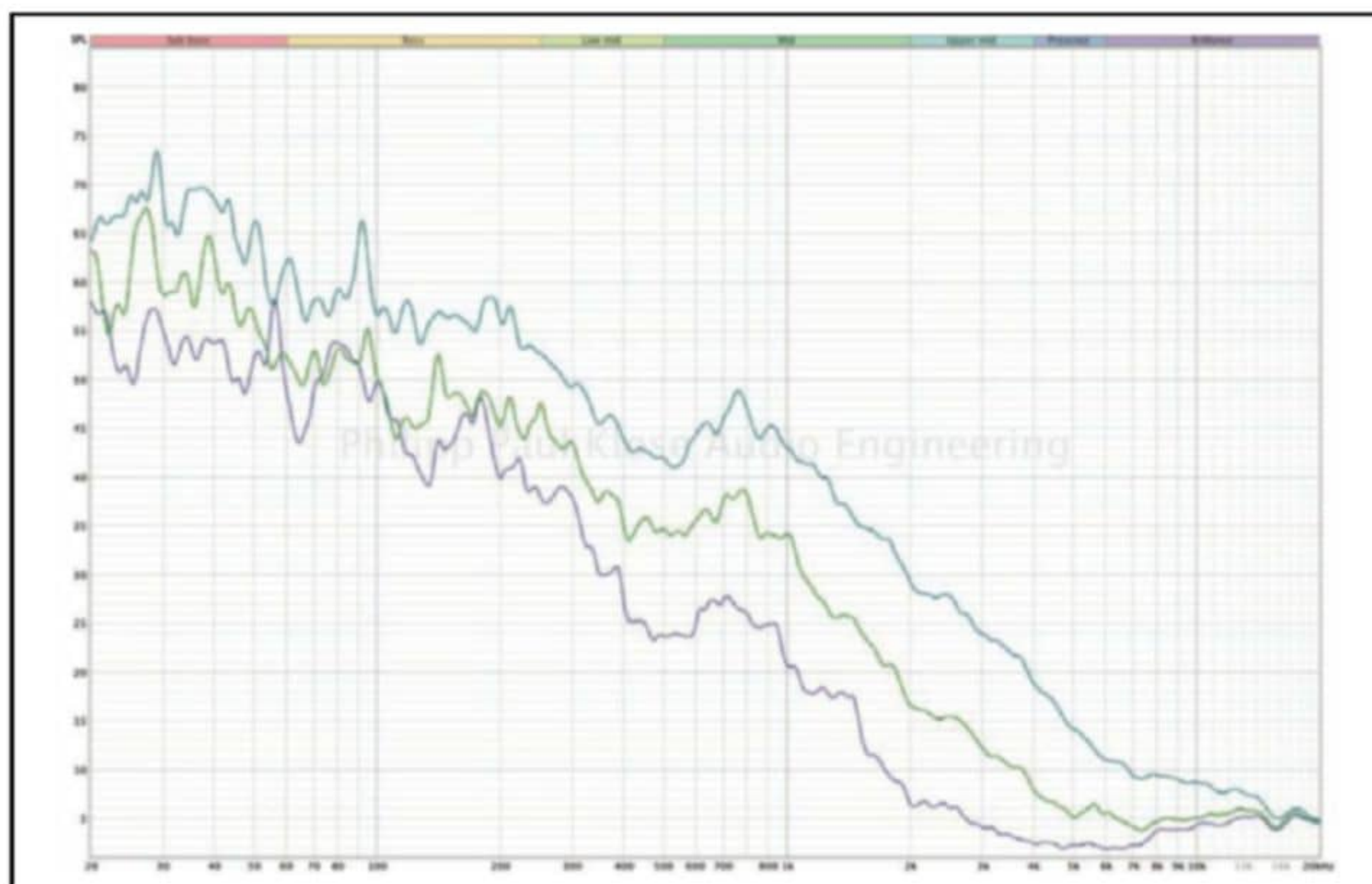


Figure 7: Driving noise measurements at 30km/h (violet), 80km/h (blue) and 130km/h (green) of a luxury midsize SUV with a six-cylinder inline diesel engine. Graphs are not weighted. The peaks below 30Hz are the engine speed. The faster the car, the louder the noise in the interior.

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ear can easily detect them. Often trim pieces that have become loose vibrate in the audible range. This can be a problem especially when you are pumping a lot of acoustic energy into your cabin. To communicate about these issues, engineers have created plenty of rather funny sounding terms, such as oilcanning, whining, and jangling. Rub and Buzz are terms that are also probably familiar to loudspeaker engineers. They refer to the sound of broken or unaligned parts (rub) and loose particles (buzz) in a speaker housing. If the vibrating parts are bigger, then it is called rattle. Often fixation clips become loose and create these acoustic events. A low vibration is called humming. Very powerful sound systems let for example the bumper and the mechanical overpressure vent hum. When two materials with different degrees of freedom move against each other, a stick-slip is created which is interpreted as a squeak by our ears.

To get the chassis and interiors quiet from acoustic effects, the application of acoustic measures such as alubutyl, butyl, foams and mass loaded vinyl can be used to stop resonances from interior and chassis parts [4].

A good way to check and locate vibrations is to use a slowly stepping signal generator with sine waves on the single amplifier outputs.

Summary

Automotive Audio is a complex and wide-faceted area, demanding many skills from engineers. This primer has shown

that a lot of thought in designing and a boat load of work in executing a good sounding car is required.

With the system design implications, pressure chamber effect, null points, and the NVH issues overlaying and influencing each other, the effort is high to make a car sound great. In the development of a mass-market oriented series car the number of challenges will become even bigger. This includes issues like software integration, bus systems (A²B, CAN, time-sensitive Ethernet), wiring, cable harnesses, value engineering, connectors, and parasitic loads/capacities. The list of challenges could be extended almost endlessly—and if there is interest in these topics, I might shed light on them in the future. In the end it is about connecting the human urge for mobility and the emotions of automobiles and music. But the gratification from playing music in a finished car and happy customers is worth the effort. 🎧

About the Author

Philipp Paul Klose is a senior engineer for automotive audio concepts and functions at CARIAD SE and previously served as an audio system development engineer and a project lead and infotainment test engineer. He holds a Bachelor of Arts (Hons.) degree in Audio Production. Within his 10 years of automotive experience, he has worked for customers like Audi, Volkswagen, Porsche, Lamborghini, and Ferrari.





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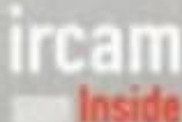
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Bass Reflex Performance Envelope

Part 1 — The Foundation

By
Gary Gesellchen (Vanatoo)



This article takes a novel approach to estimating the performance envelope available from bass reflex loudspeakers. It shows what determines how loud and how low a system will play. Part 1 discusses how bass-reflex works, the analysis approach, and the assumptions. Part 2 expands on the system analysis, and Part 3 details an example of a design option. A series of Excel spreadsheets will accompany Part 3 to allow the reader to experiment with different designs.

This is a semi-technical article intentionally written in a conversational style to tell its story. You might need to be an engineer to dig into the underlying math and challenge the assumptions, but the intent is that you do NOT need to be an engineer to understand the conclusions and the point of it all. If you know what “frequency response” means,

and you know that all devices have physical limits, you should be able to follow the argument and understand the conclusions.

When confronted with a clean sheet, loudspeaker designers must ask themselves “what’s possible?” While overall sound quality is what defines a system, the bass response is typically what dictates the size, power requirements, and a good deal of the cost of a loudspeaker system. This paper will look at a subset of designs typically possible from a “bass reflex” loudspeaker.

The results can be used by the designer to quickly home in on what’s possible, and to define the requirements for a proposed system. The results can also be used to “reverse engineer” an existing design to know what to expect. This article also shows which bass reflex designs can be realized with a ported cabinet versus the family of very attractive design options that are only realizable with a passive radiator design.

The Basics

“Bass Reflex” is sort of an old-fashioned term that was applied to what today we more typically call ported or vented speakers (**Photo 1**). I’ve chosen to use the term here since I think it evokes the sort of working action shared by both ported and



Photo 1: An example of a front ported speaker. A bass reflex system, also known as a ported, vented box or reflex port, explores a port (hole) or vent cut into the cabinet and a section of tubing or pipe affixed to the port. This port enables the sound from the rear side of the diaphragm to increase the output of the system at low frequencies as compared to a typical sealed- or closed-box loudspeaker or an infinite baffle mounting.

passive radiator speaker systems and can be used as a single term to cover both. A bass reflex system is one in which another moving element is added to the speaker enclosure in addition to the driver. This element, be it the mass of air in a port or the diaphragm of a passive radiator, is driven by the air pressure in the box (**Photo 2**). When properly constructed, the additional moving element will resonate at a lower frequency than the driver and effectively help extend the system's bass response. While "resonance" is typically considered a dirty word in audio, a well-designed bass reflex system can have tight, articulate bass that is very satisfying to even the most critical listener.

There seems to be a lot of misinformation on how a bass reflex system works. I often read about how it utilizes the sound coming off the backside of the woofer cone to increase the bass output. While this is sort of true, it really blurs the physics of what is happening. A brief description that technically holds water seems in order.

A ported system is one that typically has a tube inserted in the cabinet such that air can flow in and out. Sometimes a slot is used instead of a tube, and for the most part the exact shape of the opening is of secondary importance. The important thing is that there is a slug of air that is defined by the dimensions of the port, and this slug of air has a known mass. When the air in the port moves into the box it compresses the air that is already inside, increasing the pressure, which wants to force the slug of air back out of the box. The compressed air in the box therefore represents a "stiffness," or spring.

In classic engineering mechanics, the two requirements for a resonance are a mass and a stiffness. Our system has both, and therefore the mass of air in the port will resonate against the stiffness of the air in the box, resulting in what is known as a "Helmholtz resonance." This resonance is something you probably experienced at a young age when you blew across the top of a bottle and were able to generate a steady tone.

The Helmholtz resonance of the port and box is what a bass reflex speaker uses to extend the bass a speaker might otherwise possess. The in-and-out motion of the driver produces pressure in the box, which drives the in-and-out motion of the air in the port. Each of these vibrating "surfaces" produce sound. The visualization of how these add together in our system is shown later, but the design trick is to get these two sound-producing mechanisms to blend together seamlessly.

Figure 1 shows the cross-section of a ported bass reflex. At low frequencies the outward motion of the driver creates a vacuum in the box, drawing



Photo 2: A bass reflex system with an opening called a port or vent cut into the cabinet, generally consisting of a pipe or duct (typically circular or rectangular cross section).

air in through the port. As the frequency increases to the Helmholtz resonance and above, the moving air in the port can no longer keep up and starts to lag the cone motion. At higher frequencies the air in port has enough inertia that it moves very little, and the lag is great enough that it is effectively in phase with the motion of the driver.

Thuras was issued the original patent for this type of system in 1932. It wasn't until the 1960s (Albert Neville Thiele) and 1970s (Richard Small) until the behavior of the system was thoroughly defined mathematically. The Thiele-Small (T-S) model of bass reflex loudspeakers shows how to properly configure the driver, box, and port such that the overall sound generated by the port and the driver blends together in a predictable and desirable manner. The widespread adoption of the T-S model and the success of systems designed using it is responsible for the explosion of high quality (if often somewhat similar) ported systems on the market today.

At this point, it is worth noting that a passive radiator, which often just looks like a driver with no magnet or motor, works almost identically the same as a port. This also describes its behavior, and

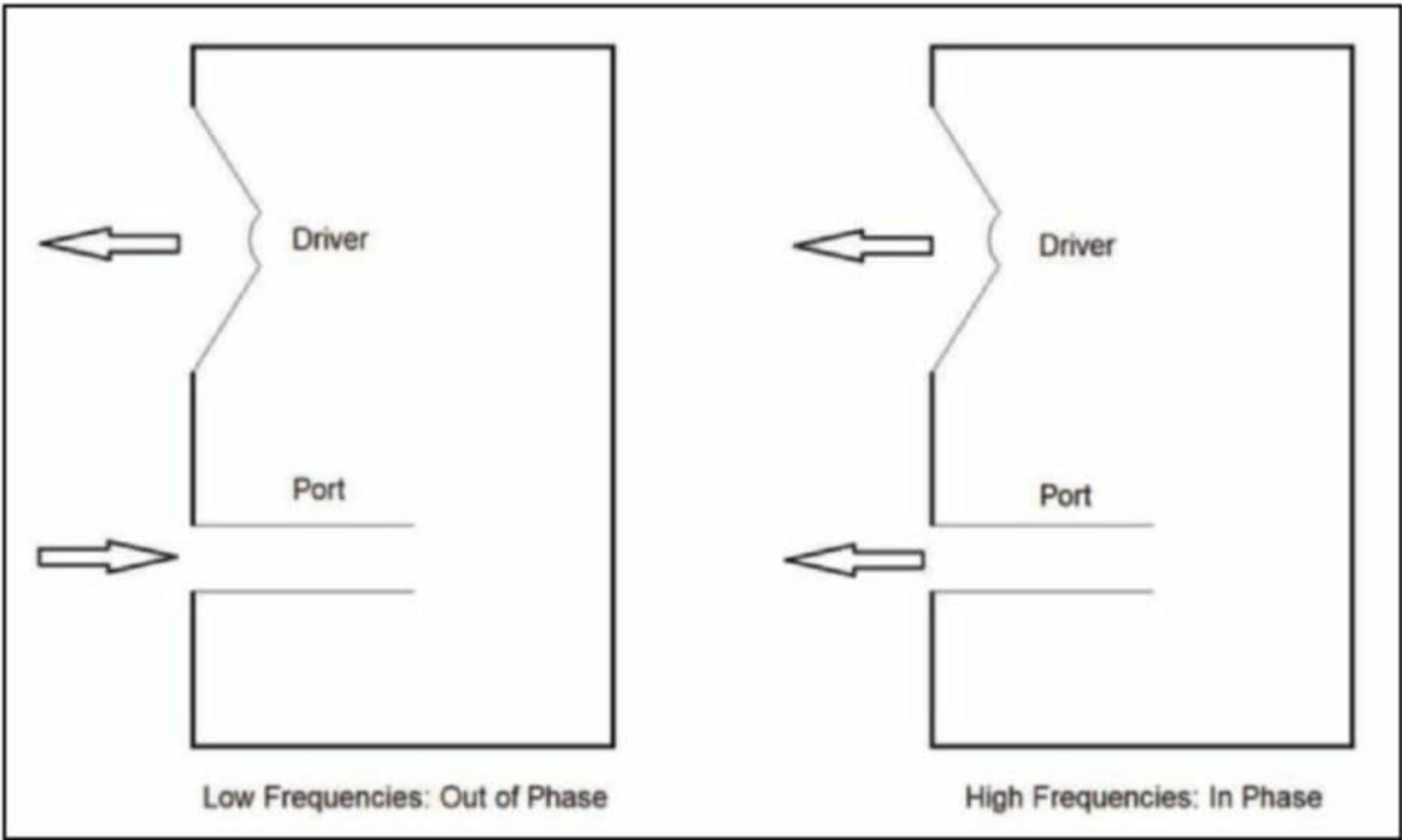


Figure 1: Ported bass reflex speaker, motion of driver, and air in port air at low and high frequencies

Photo 3: The Vanatoo Transparent Zero active speakers can reproduce lower bass with a small cabinet by combining a custom aluminum cone woofer with a passive radiator, optimized with DSP to provide a frequency response from 56Hz.



both can be considered “bass reflex.” The primary advantage of a passive radiator is that it allows a lower Helmholtz frequency in a small box than can be achieved by a port. This means deeper bass from small boxes, which is why Vanatoo incorporates passive radiators in its Transparent line of powered speakers (**Photo 3**).

Response Types

Thiele, Small, and most of the pioneers of loudspeaker design were electrical engineers. Different ways of filtering electrical signals were a hot topic in electrical engineering in the early 1900s, and a lot of different filters with different response types were developed and documented. Thiele recognized that a bass reflex speaker is analogous to an electrical high pass filter, and that if he looked at its behavior from the right mathematical perspective, the already well-developed theory of electrical filters could be used to describe a speaker’s behavior. Thiele defined different ported speaker “alignments” that would cause a speaker’s response to match one of the already well-known electrical filters and generated a set of design tables that showed how to build such systems. A decade later Small extended Thiele’s work to include boxes with losses, more alignments, and other useful insights. While a speaker is not required to match any known filter, doing so meant you understood its behavior at a time when no engineer had a computer on their desk!

This article will concentrate on just one of the infinite possible responses: The Fourth-Order Butterworth high-pass response (or “alignment”). The Butterworth response is chosen for several good reasons. It is arguably “the best” of all possible responses because it typically has the lowest -3dB frequency of any system that has a “flat” frequency response without humps or ripples. The Butterworth

design also requires that multiple design parameters be precisely related mathematically, which allows the analysis to be reduced to some simple yet powerful relationships. While it might be possible to have higher output or lower f_3 than this model predicts with a different “alignment,” the results of the Butterworth analysis should be indicative of bass reflex in general.

The Analysis Approach

The approach taken in this analysis pulls together a lot of different pieces from loudspeaker design and acoustics to tell its story. It starts with limiting the scope to systems with a Butterworth response. It shows that if you have a ported system with a Butterworth response, you don’t need a lot of other constraints to really understand the system. If you know the system’s -3dB frequency (f_3), you already know the frequencies that limit the maximum sound pressure level (SPL) the system can produce.

If you know the size of the driver based on the currently popular sizes on the market (4”, 8”, etc.), you can make a reasonable estimate of the system’s maximum SPL. The port is defined as being big enough to keep up with what the driver can deliver in terms of SPL. The relationship between box size and power is developed, where it is shown that the designer is theoretically free to choose any box size. The box size and port that will work best for the chosen design are defined. Lastly the requirements that the design places on the driver are defined.

Assumptions

This analysis is going to apply the T-S model to a Fourth-Order Butterworth system (B4 from now on) and assume that the model can predict when the system reaches its physical limits. Experienced speaker designers will know that T-S is a “small signal” model and is not generally considered valid at

large amplitudes. There are newer models than T-S that include the effects of the speaker parameters varying with displacement and current in a nonlinear manner, but they are messy and complicated. We know that the linear T-S model isn't really valid at the larger amplitudes that we are going to encounter in this analysis, but it is better than throwing our hands up and saying we don't know anything. In the author's own experience, the driver stiffness (and PR stiffness if one is employed) is the parameter that changes most with large excursions.

Drivers tend to look about 50% as stiff when driven to their Xmax than they do when measured in the small-signal domain. Reducing the driver stiffness by 50% in the model shows that in most cases the overall results aren't changed dramatically. The maximum SPL is typically reduced by about 3dB when the stiffness drops by 50%, with all the trends remaining the same. The sum of the nonlinear effects means the system won't really behave like our model at its limits, but the overall behavior will be close enough to make the analysis worthwhile.

An assumption of the T-S model is that the driver is mounted on an infinite wall, radiating into half-space. This becomes important when we get to the

part of the analysis where we consider how much power is needed to drive the system.

We are going to assume specific relationships between a driver's advertised size and its cone area (Sd) and maximum usable displacement (Xmax). Neither one of these relationships will match many real-world drivers exactly, but in the author's experience they represent good averages of what is currently available in high quality hi-fi drivers. We will also assume a few other details about our driver and port.

Next Month

In Part 2 of this three-part article, we will answer the question of what determines how loud and how low a system will play, how that relates to generally available drivers, and make the observation that box size is entirely up to you. 

About the Author

Gary Gesellchen is a Mechanical Engineer with a background in design and manufacturing. He built his first pair of speakers in high school and has been hooked ever since. Other interests include being in the mountains with nature, travel, and time with family and friends.



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Power Transformer Parameters, Selection, and Testing

Part 9 — Testing and the Results

In Part 9 of this audio electronics series, we focus on testing and the results that are obtained. In the previous articles, we have already detailed the history of transformer cores, their various construction methods, their materials, various testing methods, and their losses, commercial power quality, the design of current transformers, and most recently, additional transformer core types and their advantages and disadvantages.

By
Chuck Hansen

Test 1

For Test 1, I focused on no-load excitation with various sheet metal chassis materials, designed to explore concerns whether the metal chassis on which a transformer is mounted will adversely affect its core losses.

I measured the baseline primary no-load exciting current test for a shell-type double C-core step-down power transformer (**Photo 25**) with a pair of Kapton-insulated flat-wire secondaries with different numbers of turns. The smaller orange wire connects to the 115V AC 400Hz primary winding (the other orange-white primary wire is hidden

to the left behind the core). The white removable Teflon spacers are used to prevent epoxy from entering the forced-air cooling passages during epoxy impregnation.

The total secondary is rated for 3.92kVA, and it is one of three transformers used in a transformer-rectifier unit (TRU), whose full-load output current is 400A at 28V DC after interconnecting the wye and delta windings to full-wave rectification using 12 Schottky power diodes. The three transformers in a three-phase TRU assembly use three different color wires to delineate the phases: brown/brown-white, red/red-white and the orange/orange-white shown here. The core material is the aerospace alloy supermendur with a Bmax 122% higher than 3% silicon-steel.

The test transformer was placed on the Formica-covered MDF shelf of the test stand for its initial production test. The transformer primary was connected to an AC power supply with variable frequency and voltage capability.

The no-load magnetizing current was 73.3mA at 115V AC 400Hz. This compares favorably with the 74.5mA value as recorded in the production traveler where the C cores are clamped together with the windings installed, before the bands and the mounting plate are installed. We clamped each of the pieces of 7" x 7" sheet metal against the top air gap using a large wooden hand-screw clamp. We clamped the transformer with the top air gap facing down onto the large piece of cast iron for Test 5.

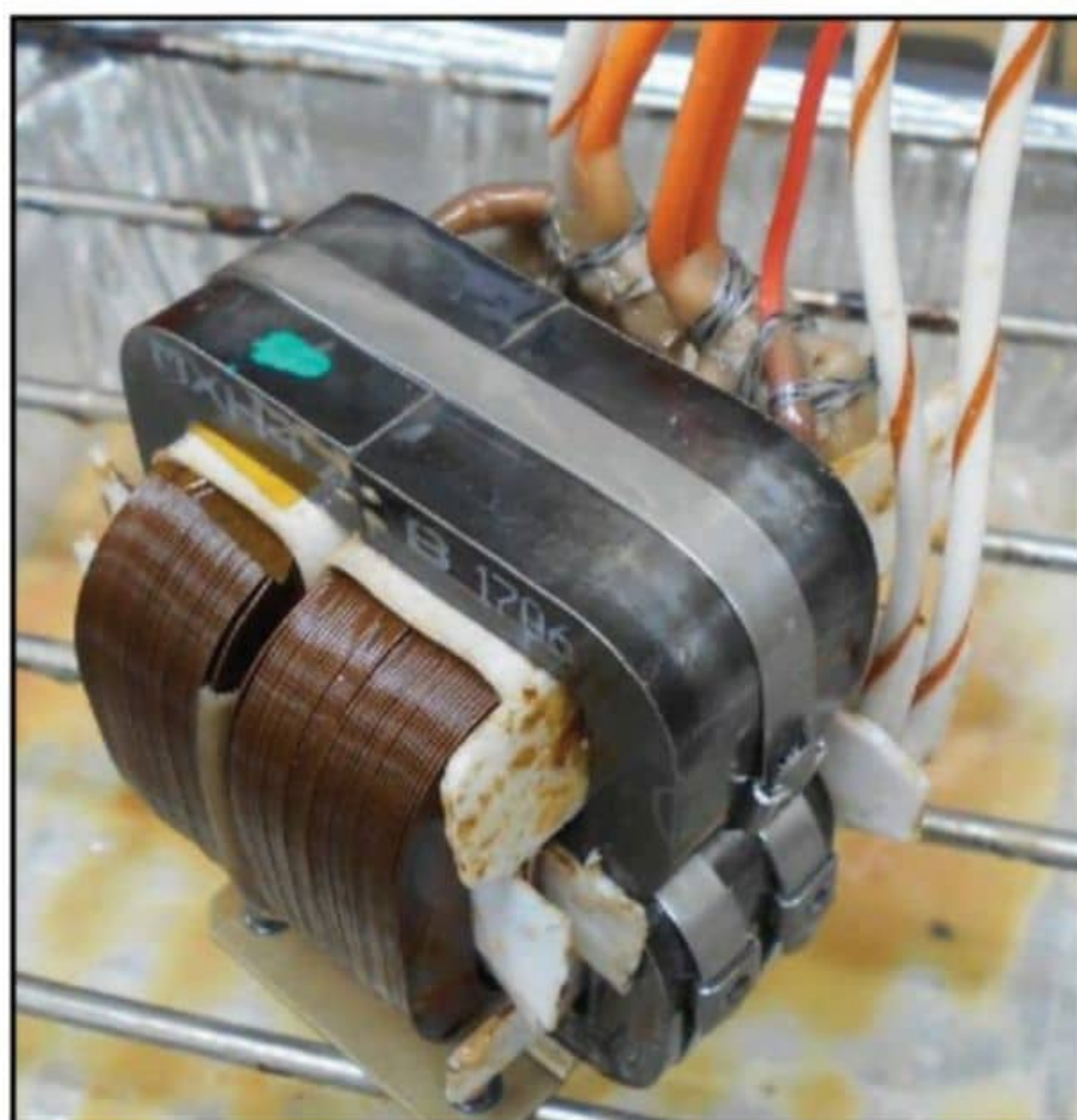


Photo 25: Shell-type double C-core step-down power transformer. Transformer production engineering after epoxy vacuum encapsulation, with the removable Teflon spaces still installed.

Test	Material	Thickness	Initial μ Permeability	Pri Excitation Current (mA)	Remarks
1	Mild Low Carbon Steel	18 gauge (0.05")	100	73.2mA	
2	316 Stainless Steel	18 gauge (0.05")	1.005	73.2mA	Non-magnetic
3	420 Stainless Steel	18 gauge (0.05")	950	73.2mA	Ferromagnetic
4	Galvanized Steel	20 gauge (0.04")	n/a	73.2mA	
5	5052 Aluminum	16 gauge (0.05")	1.000022	73.2mA	
6	Cast Iron	1/2"	5500	73.3mA	
7	MuMetal	0.36mm (0.014")	4000	73.3mA	at H=0.010e
8	Copper	22 gauge (0.028")	0.999994	73.2mA	
9	Brass	20 gauge (0.031")	1.06	73.2mA	
10	Carbon Steel	20 gauge (0.037")	100	73.2mA	see Note 1*
11	5052 Aluminum and Carbon Steel	20 gauge (0.037")	100	73.2mA	see Note 2*
*Note 1. This grade of steel is used for metal enclosures and decorative products. *Note 2. The 16-gauge aluminum was clamped between the top core gap and the 20-gauge carbon steel plate. I was told this method was suggested some time ago on one of the Audio forums to reduce the adverse magnetic effects of a steel chassis. Since the relative permeability of aluminum is essentially the same as air, it has no magnetic attenuation effect and would simply function as a spacer.					

Table 2: We obtained 11 test results with nine different types of metal.

I had not planned on using so many different metal samples, but once the machine shop found out about our experiment, more sheet metal alloys came out from under the storage benches. I believe, based on these tests, that the material on which a C core transformer is mounted (and by extension, a toroidal core, O core or R core) does not substantially affect the exciting current. The data can be found in **Table 2**.

It may be more noticeable with EI lamination transformers, but I did not have a suitably large 400Hz transformer at the time of our tests. More about this is in Test 3 where I measure the leakage flux for different types of 60Hz transformers.

Magnetically coupled H-field noise is much more difficult to shield against than electric E-fields because it can penetrate conductors. Materials that provide adequate shielding at high frequencies are much less effective at low frequencies because shielding decreases exponentially with wavelength. It varies roughly between 1/√2 and 1/√3, depending on distance, shielding materials, and other factors.

To be effective, shield material thickness must be at least three skin-depths at the H-field frequency. These thicknesses in millimeters at three skin-depths for three common metals are shown in **Table 3**.

High-permeability magnetic materials such as mu-metal offer even better low frequency H-field performance than the three listed in Table 3. However, while mu-metal is more effective than any other shielding material below 1kHz, as the frequency increases, the permeability of mu-metal (and other high permeability materials such as permalloy) and hence shielding efficiency, decreases

while non-magnetic materials such as copper or aluminum become more effective.

The permeability of all high-mu materials falls off at frequencies above a few tens of kilohertz. Around 100kHz shielding efficiencies compared to copper and aluminum are about equal. If mu-metal is saturated by an extremely large H-field, it loses its advantage at low frequencies as well. It is also very expensive and sensitive to handling.

The best methods to protect against low-frequency H-fields involves proper physical layouts inside the chassis and selection of shielding materials which are appropriate for the frequency and magnetic field strength. The most successful approach at low frequencies is to maximize the distance between H-field sources such as power transformers, and the sensitive circuits and interconnect wiring.

H-field pickup depends on the area inside the receiving wire pairs (the loop antenna area). This is the product of the wire length times the distance between wires. Using a twisted pair with an even number of turns will minimize the loop antenna effect.

H-Field Frequency	Copper	Aluminum	Carbon Steel
60Hz	25.65	37.59	2.540
400Hz	9.936	14.56	0.984
1kHz	6.248	8.001	15.24
100kHz	0.610	0.838	0.076
1MHz	0.229	0.229	0.025

Table 3: Three skin-depth thicknesses (mm) are shown for three common metals at five H-field frequencies.

There is a book on Electromagnetic Shielding by Donald R. J. White, MSEE/PE called *Electromagnetic Shielding Materials and Performance*, published by Don White Consultants, Inc. It is now out of print, but the second edition is available online. White was well-known in the aerospace community, where control of H-field and E-field EMI is extremely important.

White always stressed that anticipating and minimizing EMI problems early in the design cycle is vastly more cost-effective than trying to solve them after the fact. EMI mitigation is often an iterative process, where testing and solving EMI problems early in the program development cycle helps reduce overall costs and rework.

Test 2

My second set of tests was performed to compare a commercial 60Hz and a MIL-PRF-27 400Hz transformer with 115V AC VA ratings as closely matched as I could find in available transformers, using just 400Hz power. We use commercial 60Hz 115V transformers in some of our 400Hz 115V AC test setups to save cost, so I was curious to measure the losses of each type at 400Hz. The two transformers both have a 115V AC primary and a 36 VCT secondary. The commercial 60Hz transformer is a Hammond 166G36 (18 VA) with E-I laminations, and the MIL 400Hz transformer is a Pico Electronics 88660 (15 VA) with a toroidal core.

First, I did a baseline test with the 60Hz Hammond transformer to measure the VA with its specified 115V AC at 60Hz frequency.

I used an Instek APS-7100E 1000 VA linear AC power source with 0.12% THD, to avoid inevitable changes in the 60Hz power line AC amplitude and crest factor due to fifth and seventh harmonics, as well as the third harmonic and higher frequencies imposed on the 60Hz line when our 250hp variable frequency drives are operating. The secondary load points were no-load, 20% load, 50% load, and 100% rated load.

There is one lab setup where we drive the Pico MIL-PRF-27 transformer connected in reverse, with a 36V AC 400Hz Class-D amplifier source into the secondary winding and our 115V AC test loads connected to the primary winding.

I ran the tests on both transformers connected in forward and reverse. I plotted four graphs for the Hammond primary voltage and secondary voltage configurations, at 60Hz and 400Hz. I plotted two graphs for the Pico transformer at 400Hz. They do make MIL transformers with 26V AC 400Hz primaries for older avionics systems, but that is incidental to what I wanted to observe.

The maximum flux density for 60Hz transformers with 3% silicon-steel E-I lamination is about 1.5T. A 60Hz transformer driven at 400Hz could theoretically handle up to 6.67 times the primary voltage rating, based on the reduced B_{MAX} at 400Hz. However, the eddy current loss (P_{EDDY}) could be over 40 times larger due to the squared core tape thickness and frequency terms in the eddy current formula:

$$P_{EDDY} = (\pi * t * B_p * f)^2 / 6 * \rho$$

Even if we were willing to accept the higher eddy current losses, the primary winding insulation might be overstressed at the increased voltage. At a minimum the transformer is being used well above its optimum operating point. Conventional wisdom says a 115V AC 60Hz transformer used at 400Hz should still be treated as a 115V AC transformer. If the frequency is increased while the primary voltage is held constant, then hysteresis loss is reduced while eddy current loss is unaffected since B_p decreases proportionally when f increases.

For the same VA output, 60Hz transformer windings tend to have more turns of smaller gauge wire, while 400Hz designs generally use fewer turns of larger gauge wire. Winding resistances for a 60Hz transformer operated at 400Hz will be much higher than a transformer designed specifically for 400Hz. As a result, the voltage regulation for a 60Hz transformer operating at 400Hz will be worse than that of a transformer designed for 400Hz.

Figure 27 shows the Hammond tests at 60Hz with the primary energized at no load and with the three different values of fixed secondary resistive loads based on the center-tapped 36V AC secondary voltage (20% load, 50% load, and rated full-load). I applied primary voltages in 10V AC steps from 10V AC to beyond the 115V AC rating until the secondary voltage curves showed evidence of approaching saturation by skewing downwards, or 140V AC, whichever came first.

This 115V AC transformer can be operated at 120V AC, but the full load losses are beginning to increase, causing the curve to droop. This results in reduced efficiency and a higher operating temperature.

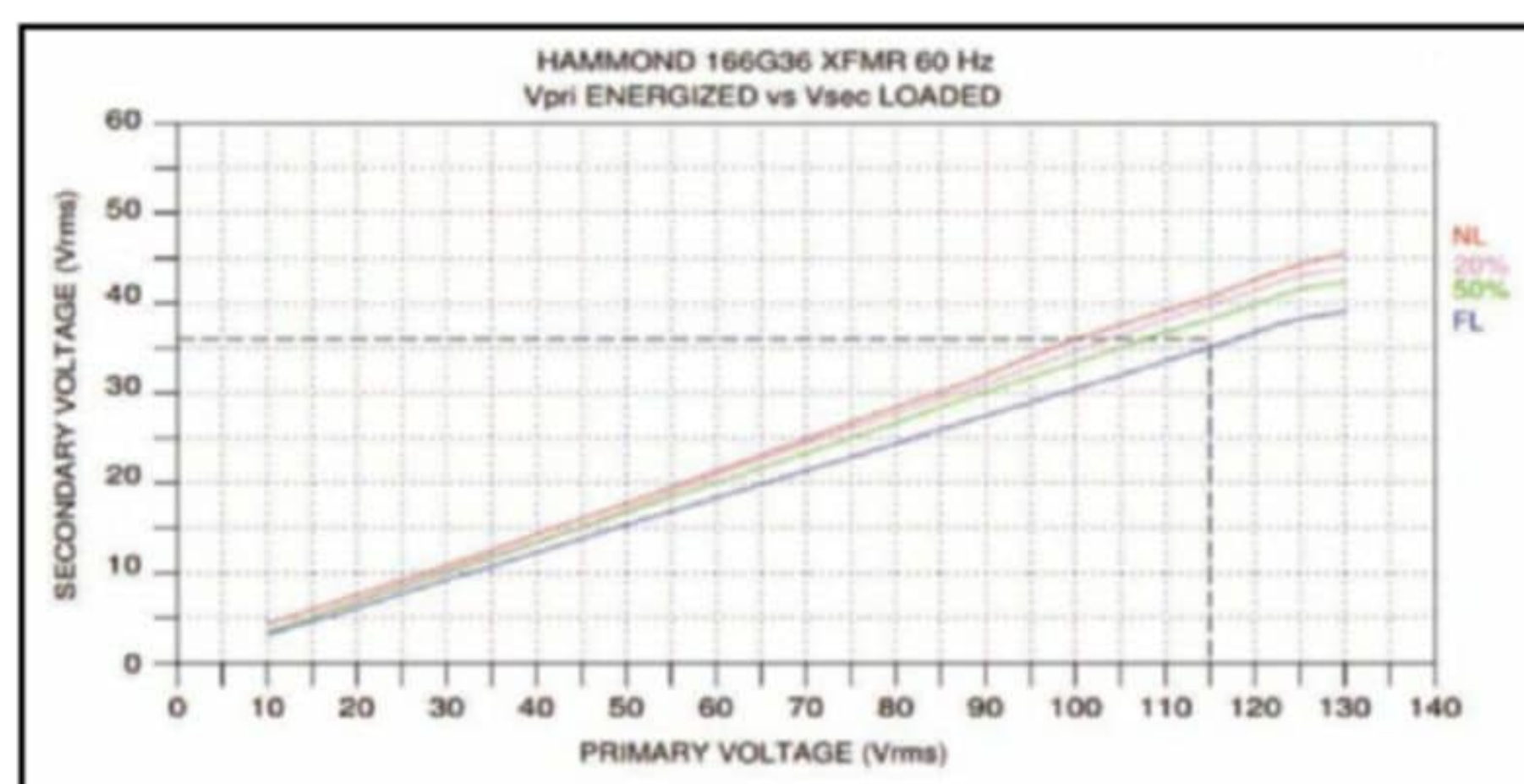


Figure 27: Hammond 166G36 tests at 60Hz with the primary energized vs. secondary loaded.

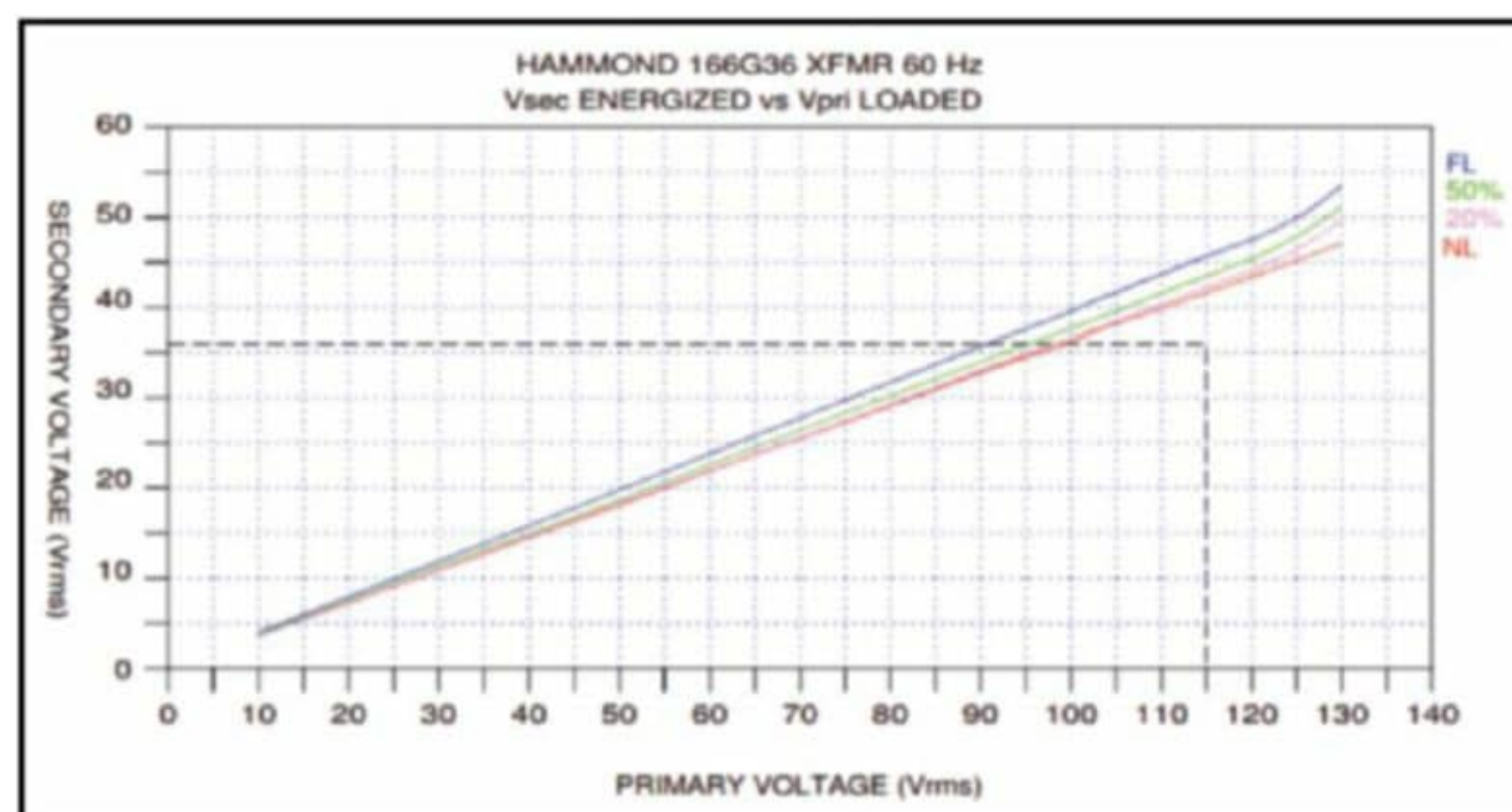


Figure 28: Hammond 166G36 tests at 60Hz with the secondary energized vs. primary loaded.

The opposite occurs when a transformer rated for 50Hz/60Hz operation is used at 60Hz. The flux density drops by 16.7%, which requires less primary current for the same primary voltage.

I recorded the primary voltage and current, as well as the secondary voltage, and the secondary current for the three loaded conditions, at each discrete input voltage step. Then I calculated the resulting VA, the effective transformer ratio, and the regulation factors (ϵ):

$$\epsilon = (V_{NL} - V_{FL}) / V_{FL}$$

The change in the center-tapped secondary voltage from no-load to full-load with the primary voltage held at 115V AC was from 40.76V AC to 35.09V AC. The regulation factor was 0.162, or 16.2%. This is within the usual range for small commercial 60Hz E-I lamination transformers. A toroidal transformer of the same rating would have tighter regulation since the core losses and winding resistances are lower.

Then I reversed the transformer connections and energized the secondary at 60Hz from 6V AC to beyond the secondary 36V AC full-load rating, at no load and with the three different values of fixed resistive loads (20% load, 50% load, and rated full-load), connected to the primary winding, based on the rated 115V AC primary voltage (**Figure 28**).

I recorded data to the point where the voltage curves showed evidence of approaching saturation by skewing upward, or 140V AC at each load. I limited the time above rated voltage if the current rating of either winding was exceeded. I recorded the same data as before, and calculated the equivalent secondary-referred VA, ratios, and regulation factors (ϵ).

The change in primary voltage from no-load to full load with the secondary voltage held at 36V AC was from 99.50V AC to 90.80V AC. The regulation factor was 0.087, or 8.7%.

I found it interesting that the reversed connected transformer had better regulation than the forward regulation. The winding bobbin is divided into two equal sections for the two windings, but the secondary winding fills a bit more area than the primary (**Photo 26**).

The Hammond 166 series in this C8H size transformer has secondary voltages available from 2.5 VCT to 50 VCT, and VA ratings from 12.5VA to 18VA. The 166G36 has the highest 18VA rating. Perhaps each secondary wire gauge is optimized to fully fill the bobbin area. The black PVC primary leads and the green PVC secondary leads are both the same wire gauge. I guess this is a matter of economy when the same transformer frame has several different available secondary voltages.

Figure 29 and **Figure 30** show the same two tests on the Hammond transformer, but at 400Hz. The change in center-tapped secondary voltage from no-load to full-load with the primary voltage held at 115V AC was from 40.30V AC to 33.23V AC. The regulation factor (ϵ) was 0.175, or 17.5%. This is 1.3% higher than the same test at 60Hz, despite the core flux density being much lower. In theory, the core losses should stay the same if the primary voltage is held constant while the frequency increases, but the wider spread could be due to an increase in other losses, such as skin effect.

Figure 30 shows the change in primary voltage from no-load to full load with the center-tapped secondary voltage held at 36V AC is from 100.3V AC to 85.70V AC. The regulation factor was 0.146, or 14.6%. This is 5.9% higher than the 60Hz test in Figure 28.

Finally, I ran the same 400Hz tests as Figure 29 and Figure 30, on the Pico transformer. These are shown in **Figure 31** and **Figure 32**, respectively. The change in secondary voltage from no-load to full load with the primary voltage held at 115V AC was from 39.30V AC to 36.6V AC. The regulation factor was 0.069, or 6.9%. Figure 32 shows the change in primary voltage from no-load to full load with the secondary voltage held at 36V AC as from 109.0V AC to 99.2V AC. The regulation factor is 0.090, or 9.0%.

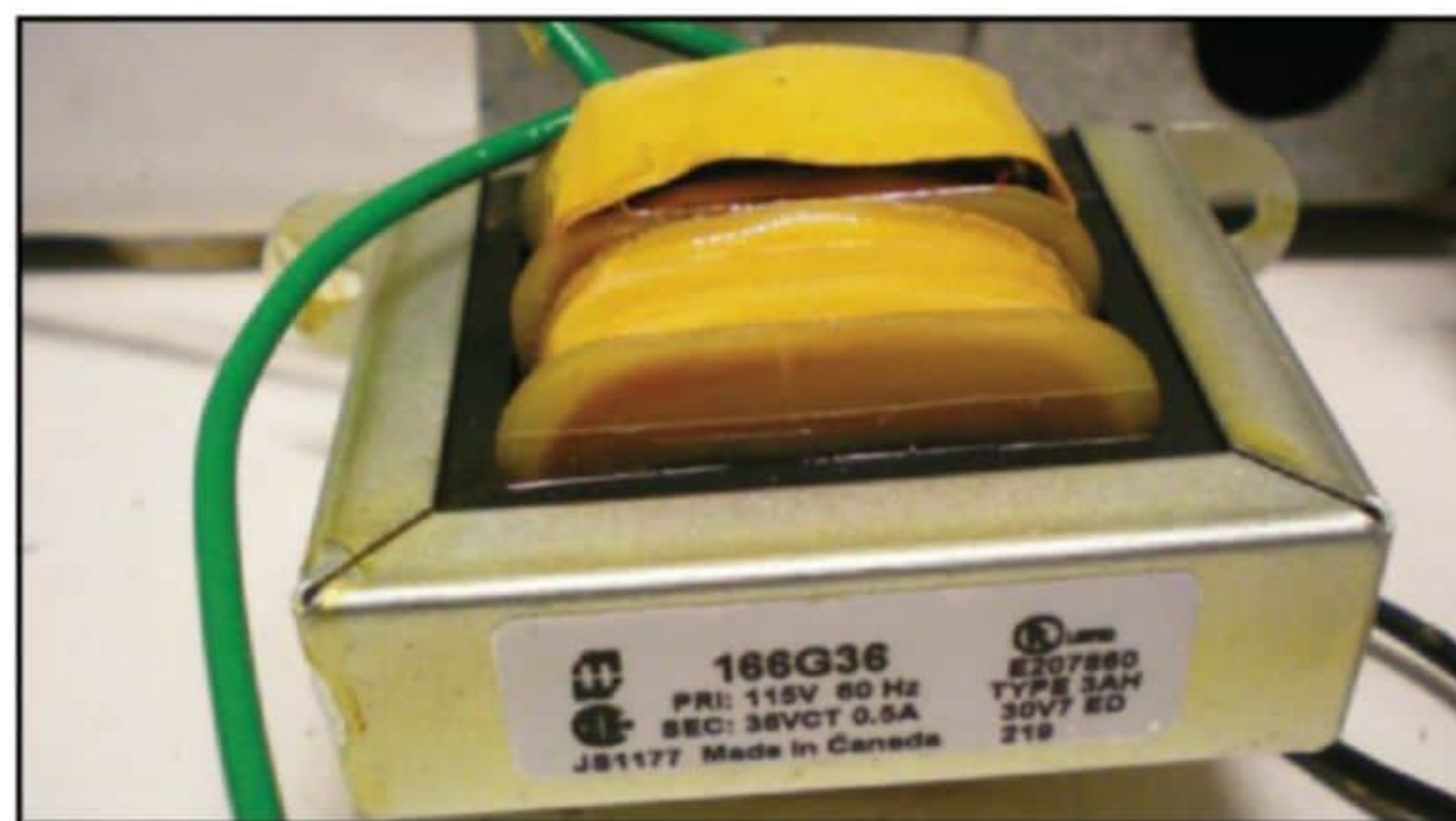


Photo 26: Hammond 166G36 transformer secondary (rear, green) and primary (front) windings

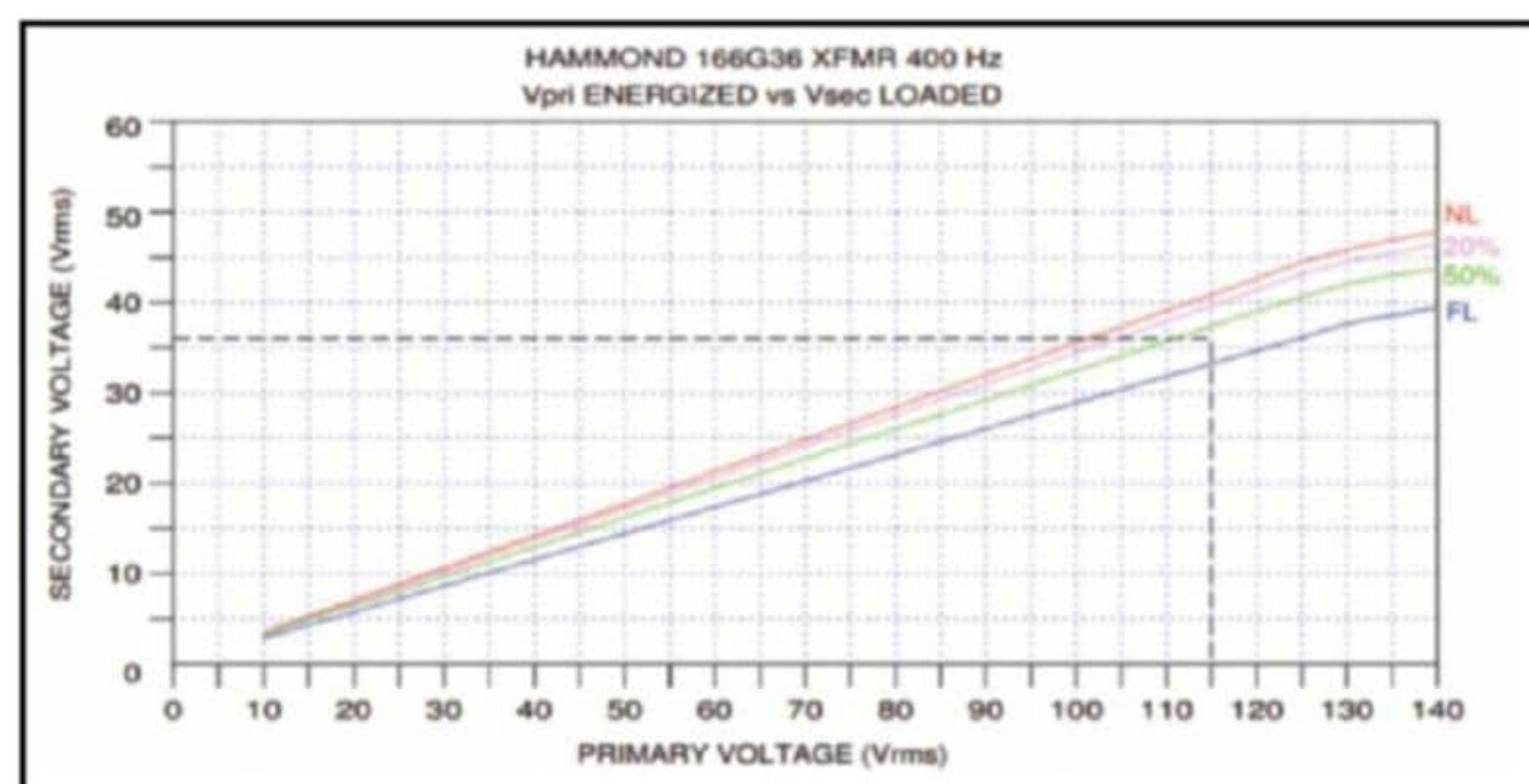


Figure 29: Hammond 166G36 tests at 400Hz with the primary energized vs. secondary loaded.

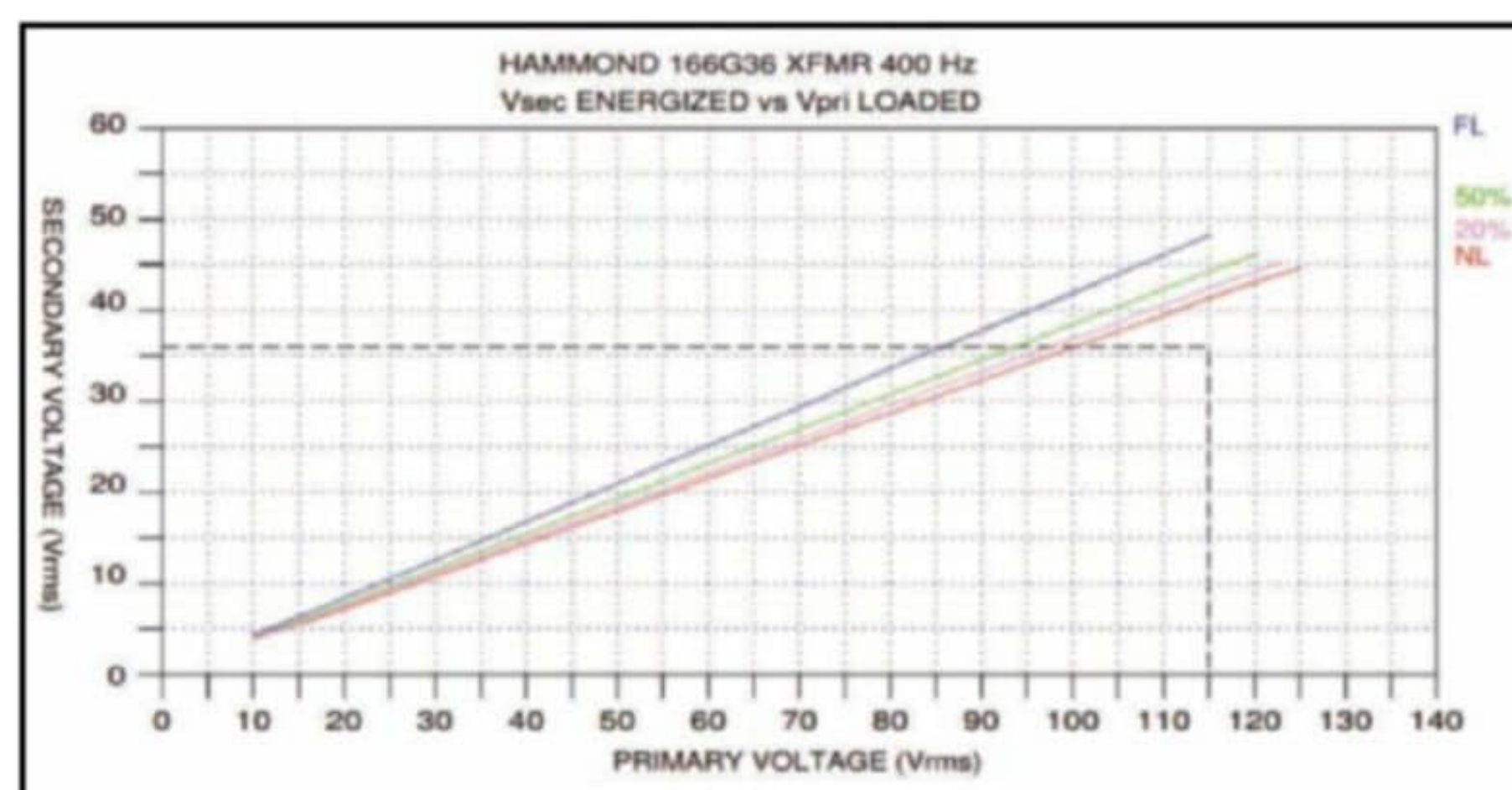


Figure 30: Hammond 166G36 tests at 400Hz with the secondary energized vs. primary loaded.

The Pico 88660 was designed with a toroidal core specifically for this 400Hz application. You can see how tightly grouped the four load lines are in Figure 31 and Figure 32, especially Figure 31, which is the normal usage with the load connected to the secondary winding. The 0.069 regulation factor (6.9%) is impressive for such a small transformer.

Test 3 – Magnetic Field Leakage Flux

Electromagnetic field (EMF) exposures are complex and exist in the home and workplace because of all types of electrical equipment and building wiring, as well as nearby power lines [6].

In the next series of tests, I mounted different 60Hz transformers on pieces of sheet metal, one at a time. I used three E-I cores, two toroidal cores and two R core transformers. I didn't have any small 60Hz C core transformers available. I used an Axial AC Magnetic Field Evaluator (MFE) sensor (**Photo 27**) to measure the relative leakage flux. It is accurate from 10Hz to 3kHz.

The chassis and housings for aerospace electronic equipment are made of aluminum and, in some cases (helicopter equipment for instance), lighter cast magnesium. The removable cover gaps are sealed with EMI gaskets. Steel is a good low-cost material for consumer use, but too heavy for aerospace electronic equipment.

Given this information I decided to do the magnetic leakage flux tests with the transformers alternately mounted on a 16-gauge (0.05") 5052 aluminum and a 20gauge (0.037") carbon steel plate. The 20-gauss relative permeability of carbon steel (100) is 1/10th of that of the 3% silicon iron used in the transformer laminations

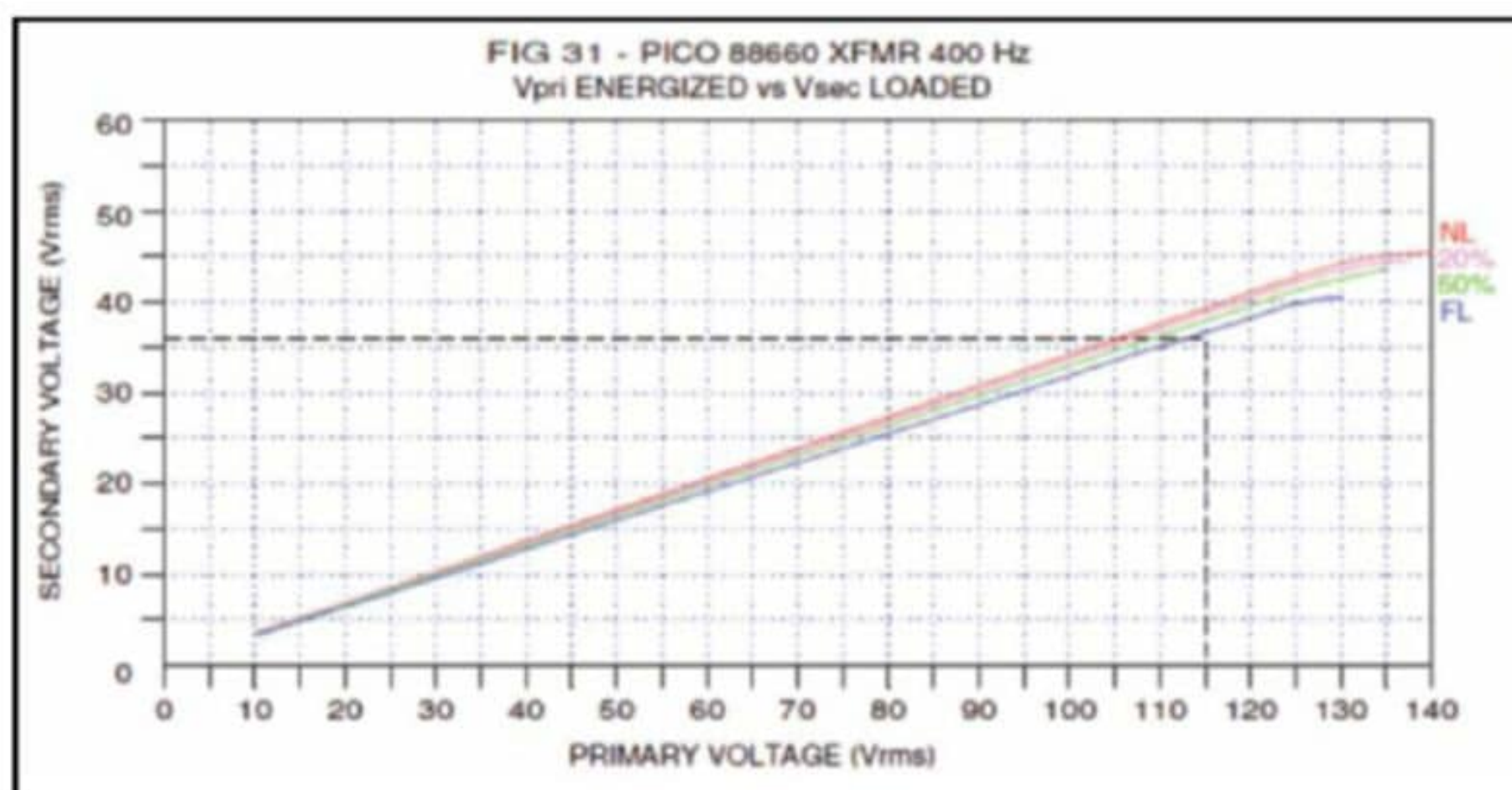


Figure 31: Pico 88660 tests at 400 Hz with the primary energized vs. secondary loaded.

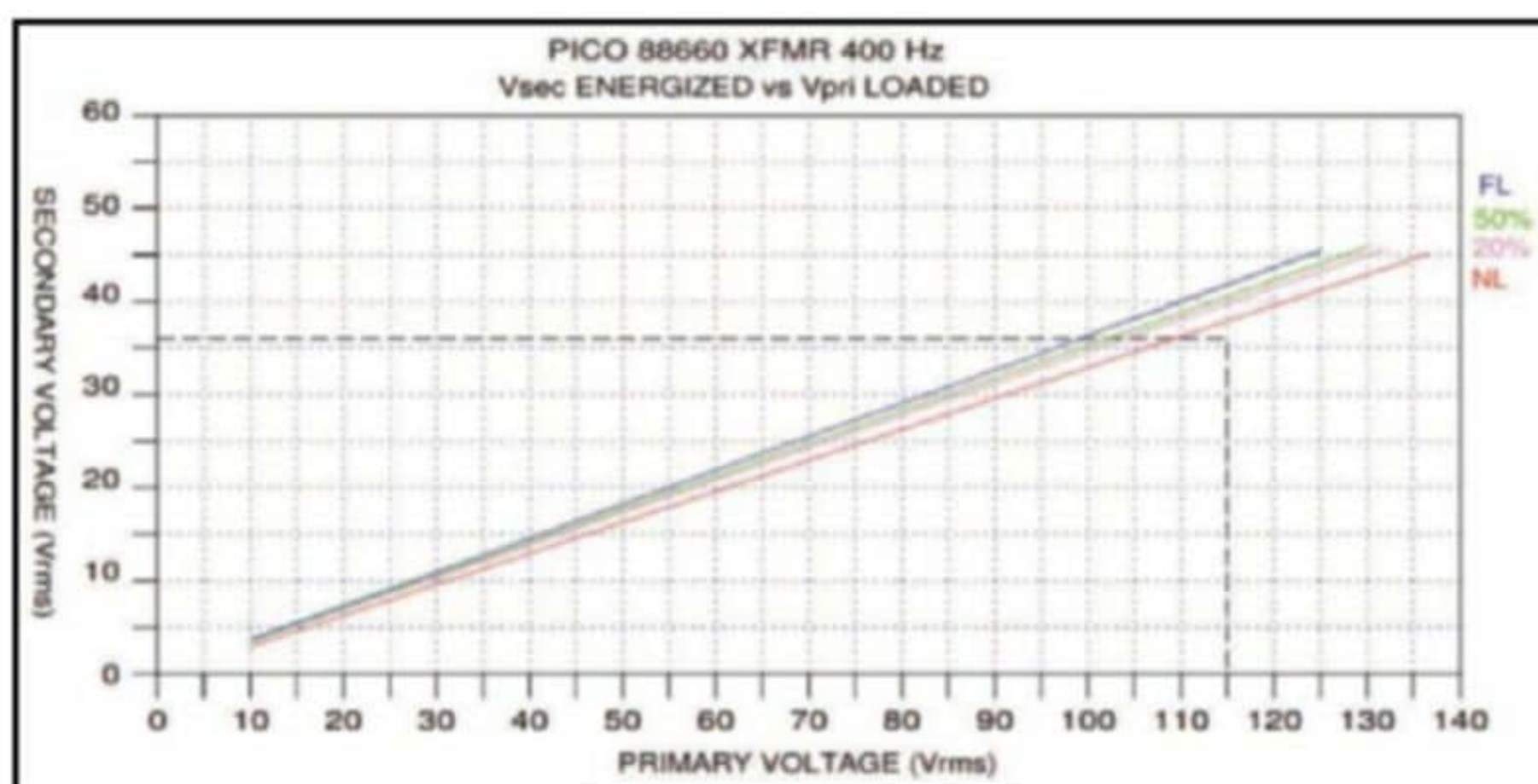


Figure 32: Pico 88660 tests at 400 Hz with the secondary energized vs. primary loaded.

(1,000) and is not even close to mu-metal (20,000). The relative permeability of aluminum (1.000022) is about the same as air (1.0000). See Table 2 for the permeability of other metals.

The MFE generates a millivolt AC (mV AC) signal proportional to the flux density, B, that it senses, which is inversely proportional to distance from the source. The millivolt AC signal has to be multiplied by a correction factor given by the MFE manufacturer to obtain flux density.

My Keysight U1241C True-RMS DMM AC voltage range has a minimum resolution of 0.1mV AC. To get better resolution for low MFE sensor millivolt levels I added my precision $\times 10$ gain voltage amplifier between the MFE sensor and the DMM. After recording the MFE readings I used a spreadsheet to apply the MFE correction factor multiplied by 1000 to get milligauss (mG) and divided by 10 to remove the gain added during MFE mV readings (in one step, multiplying by 100).

These are all near field measurements. Far field conditions begin when the distance from the field source is more than three wavelengths. At 60Hz, that is a very long distance.

The transformer primary leads for my test fixture connect to one end of a 12-position barrier terminal block across the width of the mounting plate. From there they connect to the variable AC supply with a longer 22" (~56 cm) twisted-pair wires. The secondary leads are connected to the other end of the barrier terminal block, and then to my power resistor load bank with another set of 15" wires (~38 cm) twisted-pair wires. I drilled holes on the right-hand side to accept the mounting hardware for the various transformers I tested.

Stainless steel type 316 is a better choice for the mounting hardware since it is non-magnetic and has a higher resistivity ($74.9 \times 10^{-8} \Omega \cdot \text{cm}$) than carbon steel ($14.3 \times 10^{-8} \Omega \cdot \text{cm}$) or aluminum ($4.30 \times 10^{-8} \Omega \cdot \text{cm}$), but stainless steel is also more expensive. We use type 316 exclusively for aerospace electronics units. I also used 316 stainless-steel hardware for this testing to avoid any extraneous magnetic paths that may influence the data.

Based on several MFE probe measurements at various locations on the Hammond 166G36 E-I transformer, I selected four locations that gave unique readings. The readings were essentially identical at the front and back of the windings, and identical for both the left and right sides of the transformer. I made sure that I had the

About the Author

Chuck Hansen is an Electrical Engineer and holds five patents in his field of electric power engineering. He has worked in both the electric utility and aerospace electric power industries. He was assigned as the Instrument and Controls Startup Group Lead Engineer for two nuclear power generating units. He was the Electrical Systems and Controls Supervisor for a number of military programs as well as commercial and business jet electric power programs. He has written two books for Audio Amateur publications, and more than 260 magazine articles. He began building vacuum-tube audio equipment in college, and enjoys restoring and modifying audio equipment and test equipment. In his most recent roles as Senior Design Engineer and Lab Calibration Manager, he designed specialized test equipment particular to the aerospace electric power sector.



MFE sensor perpendicular to and at the center of the surfaces I measured. This is necessary for consistent results that minimize flux spill-over from other surfaces. The location at the side of the frame is especially sensitive because the E-I core gaps are located along the sides of the outer legs of the core, near the top and the bottom. I also rotated the probe around its axis to find the peak leakage flux density at each specific location.

I initially only used one mounting screw for the E-I transformer so I can rotate it 90° to measure the flux at the windings and the frame side plates without removing it. To easily measure the flux at the bottom of the transformer, I mounted it upside down on a pair of threaded aluminum spacers with 316 stainless hardware so I could contact the exposed laminations. The top of the transformer is against the sheet metal mounting plate as shown in the drawing at the right side of **Figure 33**.

There are several factors that influence the leakage flux density for a given size and shape of the object generating the magnetic (H) field. I measured the leakage flux density (B) at different distances from the magnetic field evaluator probe (**Photo 28**). I started at the closest point to the transformer surfaces and moved the probe away until it reached a minimum millivolt reading, or 20cm, whichever came first.

I measured leakage flux density for each transformer at no load (NL) and at rated full load (FL) using its nameplate 60Hz voltage rating. I used a B-K Precision 1655 AC Power Supply, which contains an input isolation transformer, and a Variac with an output adjustment range of zero to 130V AC.

I can remember when the nominal 60Hz AC voltage was 110V AC in the early 1950s. Then the utilities increased it to 115V AC, then 117V AC, and finally 120V AC when the power grid expanded sufficiently to provide good local regulation. Some of the articles in the old *Popular Electronics* and *Radio-Electronics* magazines from the 1980s described the AC voltage as 117V AC. The EU 50Hz voltage has been 230V AC to 240V AC since the 1920s.

The V-shaped support allows me to adjust the height of the MFE sensor, center it in front of the transformer, and rotate it to find the peak leakage flux density. The centimeter scale measures the distance from the sensor to the transformer surface under test. I made measurements in contact with each of the locations in Figure 33, and at distances of 1cm, 2cm, 4cm, 6cm, 8cm, 10cm, 12cm, 14cm, 16cm, 18cm, and 20cm.

It is important to have the MFE sensor perpendicular to the surface under test. Any angular difference, either horizontal or vertical, reduces the measured flux. Once the MFE sensor was properly rotated about its axis to maximize the reading I kept it in that orientation as I moved the probe away from direct contact.

I put a piece of masking tape over the AC primary winding connections to the terminal block (to the left of the photo) to avoid an inadvertent shock from working around that area of the test fixture.

Next Month

Next month in Part 10, we will make more transformer tests with different types of transformers, for leakage flux and external magnetic shielding with both steel and aluminum, core loss

sensitivity to the presence of these metals, and measurements of external magnetic leakage fields with distance, with and without shielding. ax



Photo 27: Axial AC Magnetic Field Evaluator (Image Source: Magnetic Shield Division, Perfection Mica Co.)

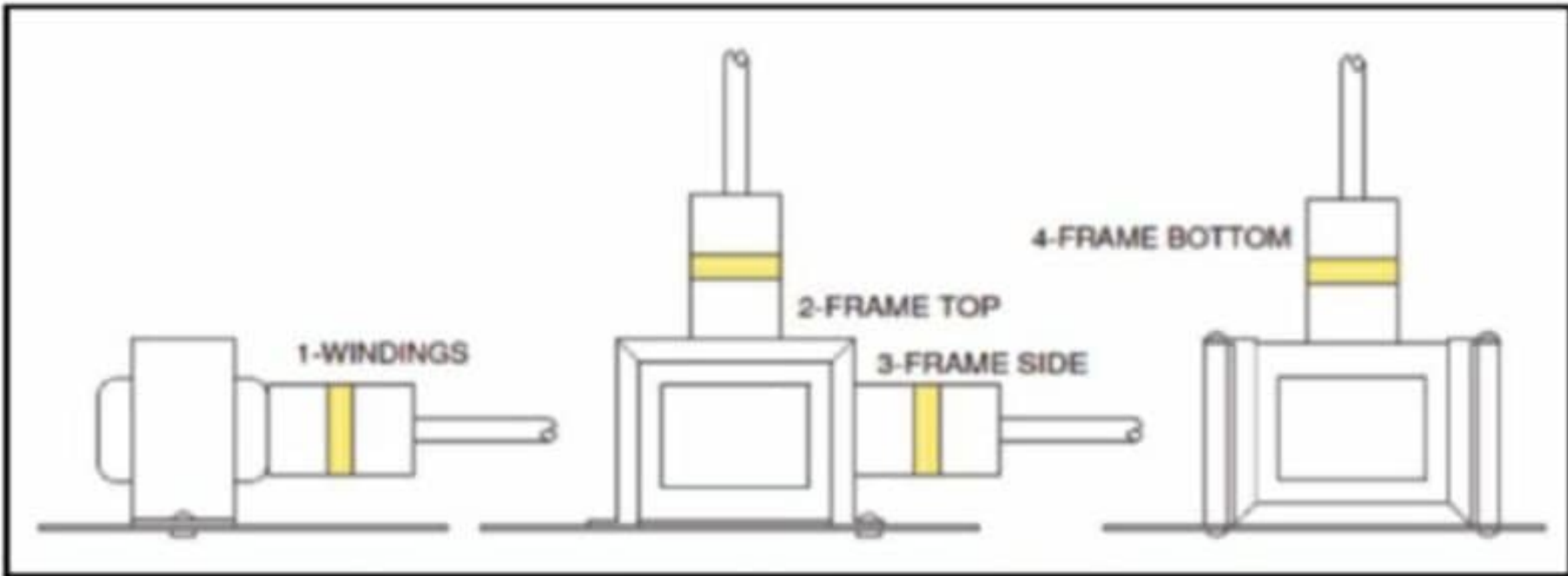


Figure 33: Here are the Magnetic Field Evaluator (MFE) sensor locations for E-I core transformers.



Photo 28: Leakage flux density measurement setup with Hammond 166G36 E-I core transformer mounted on the aluminum plate.

References

[6] *EMI/EMC Test Requirements for Military and Commercial Applications*, National Testing Services (www.nts.com); EMF Questions and Answers, June 2002, www.niehs.nih.gov/health/materials/electric_and_magnetic_fields_associated_with_the_use_of_electric_power_questions_and_answers_english_508.pdf

Ten Decades of Home Music Systems

A Family History

The history of home music systems is directly intertwined with the use of tube electronics and how these devices were crucial to the development of radio, television, sound recording, and reproduction, and analog and early digital computers, among many other things. This article revisits part of that history from a personal perspective.

By
Richard Honeycutt



Photo 1: My grandfather's first home audio system comprised this 1942 Atwater Kent Model 42 AM radio.

My Grandpa Briggs, raised in a farming and sawmilling community, was a music lover. He played guitar in his 20s and inherited a crank organ and a foot-pumped harmonium (see Resources) that he didn't play, but that his oldest daughter taught herself to play quite well. The first home audio system he had was a 1924 Atwater Kent Model 42 (**Photo 1** and **Photo 2**), on which he could listen to WBT (Charlotte, NC), WPTF (Raleigh, NC) after 1928, and WSJS (Winston-Salem, NC) after 1930. Although Grandpa played traditional and Gospel music on his guitar, he was delighted when the weekly Metropolitan Opera broadcasts began in 1931, and he was a regular listener thereafter.

At nighttime, skip (skywave) propagation allowed the family to listen to WNBC (originally an experimental station W2XBS) and WCBS from New York, WSM from Nashville, WLS from Chicago, and KDKA from Philadelphia.



Photo 2: This view of the innards of the Atwater Kent Model 42 shows the huge power transformer, three tuning capacitors, Bakelite-form air-core coils, and six tubes.



Photo 3: Grandpa's improved home audio system was this 1936 Philco Model 620 AM broadcast/short wave receiver.

The Model 42 was designed mainly for use with a headset, but an optional “magnetic reproducer” (speaker in a cabinet) was available as well. The 171A (or in some units, 371A) audio output tube could provide 700mW into a 5350Ω load. The optional speaker was essentially a headphone with a paper cone glued to the steel diaphragm. Neither the frequency response nor the maximum audio level even compared to what we’d have expected from a cheap table radio in the 1960s.

A Giant Step Up

After the Depression began to ease, Grandpa bought a 1936 Philco Model 620 radio (**Photo 3** and **Photo 4**) and moved the Atwater Kent to the attic, where it stayed until I discovered it in my teen years. In addition to receiving the standard AM broadcast band, it also picked up three short-wave bands: the band selector switch allowed the listener to choose among the 540kHz to 1720kHz, 1.75MHz to 5.8MHz, and 5.7MHz to 18.0MHz bands. The type 42 pentode output tube and the electrodynamic speaker produced very good audio for the time, at levels up to about 84dB SPL, with tolerable distortion. The 8” speaker in an open-back cabinet produced a reasonably flat frequency response from about 100Hz to 5kHz, except for a 4dB hump centered around 180Hz, due to the open-back cabinet.

The Next Generation

When my dad came back from the Army after World War II, he bought a combination radio-record player (**Photo 5**). Of course, it only played 78-RPM records. The combination of a 5” speaker and Class-A single-ended pentode output stage limited the maximum SPL it could produce, although the closed-back cabinet with substantial cubic volume did permit a reasonable facsimile of bass, for the small speaker.

Our next “home audio system” was an Arvin AM table radio. The radio design with its 5” speaker was quite similar to those in the radio/record player combination. Thus, the only advantage was the smaller size. Sound quality was even more limited because of the smaller, open-back cabinet.

Like most families I knew, we weren’t particularly concerned about sound quality in those days. The “hi-fi” era had not yet begun, and we really didn’t have anything better than a cheap table radio for comparison. Unsurprisingly, when replacement time came, size considerations won out again, and we bought a small plastic-cased GE table radio (**Photo 6**). This was our first AC-DC or “hot-chassis” “All-American Five” radio. This design included no RF amplifier; instead, the signal was fed from the antennas directly to a “converter” stage; thence, to an If amplifier, a detector/audio amplifier, and a 50C5 power tube. The fifth tube was the 35C5 rectifier. Like all AC-DC sets, the filaments were connected as a series string. This radio could put out about 2.3W into a 4” speaker, and had an even smaller cabinet than the Arvin, and a perforated Masonite back panel. Bass? What bass?

My Sister and Me

During all this time, the only records we played were from Dad’s collection of old 78s. By the time I was 10, pop music singles



Photo 4: The Philco’s innards are more tightly packed, and show the bottom of the electrodynamic speaker, several electrolytic can capacitors, one tube with a “coke-bottle” envelope, and shields around the inductors and some of the other tubes.



Photo 5: This post-War radio/record player was pretty much the limit of audio quality for many families in the late 1940s and early 1950s.



Photo 6: With our 1950s table radio, we reached the nadir of audio quality.

Resources

“The Gem Roller Organ,” John Wolff’s Web Museum, March 2017, <http://johnwolff.id.au/mechmusic/GemRollerOrgan/Gem.htm>

“Harmonium,” *Wikipedia*, April 2023, <https://simple.wikipedia.org/wiki/harmonium>

were on 7" 45-RPM discs, and albums were on 12" 33 1/3-RPM discs. My sister was given a Symphonic portable four-speed record player (**Photo 7**) that provided only slightly better sound than the plastic table radio, but at least we could listen to the same music as our friends. Since it used a crystal phono cartridge, the designers could get by using only a

Photo 7: Our first four-speed record player was a Symphonic portable.



Photo 8: This Olson mono phono preamp was similar to the GE unit I used.



About the Author

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



50C5 tube and a 35C5 rectifier. Needless to say, it was not hi-fi and couldn't play very loudly.

About this time, I began reading electronics magazines, and learned two life-changing things:

- The crystal cartridge in my sister's record player provided little bass due to the input impedance of the built-in tube amplifier, and that cartridge was also wearing out our records.
- Better technology was in fact available.

Because of the growing market for amplifiers and speakers to use with stand-alone turntables, some TV manufacturers had started including phono inputs on the backs of their sets, as early as 1954. Even though the 5W single-ended pentode output stages of the TVs limited maximum sound level, even the cheap 5" speaker sounded better in a 4ft³ case. Not hi-fi, but still a step-up. If a listener was able and willing to pay for a magnetic cartridge for the tonearm, about \$20 would buy you a small tube-based phono preamp whose output could be fed to the TV's phono input. Although I owned such a TV set, I never used it for phono reproduction.

Evolution of My Personal Audio System

During the 1960s, having reached the point of having some independent income, I bought parts and built myself a transistorized integrated amplifier. I also began to visit a nearby hi-fi/stereo store that handled some used equipment. My first purchase was a Garrard turntable with a Shure magnetic cartridge. I built a decent speaker cabinet that housed an equally decent Utah 8" speaker, and although stereo was still in the future for me, at least I had achieved a real improvement in sound quality. Since my system lived in our dining room, the whole family benefitted from the upgrade.

While I was in college, I pressed my Silvertone Twin Twelve guitar amplifier into service as a dorm-room audio system. Since my Garrard turntable had a magnetic cartridge, I added a GE one-tube phono preamp (**Photo 8**). The Silvertone's tone controls could be set to produce a reasonably flat frequency response from 50Hz to over 10kHz, although the speakers were voiced for guitar and so the system response was only tolerable. The 50W maximum output from a push-pull pair of 6L6GCs could put out more audio than my fellow dorm-dwellers would tolerate.

After graduation from college, I upgraded to my first truly hi-fi components by purchasing a used Altec 440A phono preamp (**Photo 9**) and an A340A

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Photo 9: My first really hi-fi phono preamp was a used Altec Model 440A.



Photo 10: The system using the Altec 440A preamp was rounded out by an Altec A340A power amplifier.



Photo 11: My first stereo power amp was two Heathkit W5M's.



Photo 12: My H. H. Scott FM stereo tuner is still in service.

power amp (**Photo 10**). The preamp was designed before the Recording Industries Association of America (RIAA) equalization was standardized, so it had multiple equalization options. There was a function switch that included settings for PHONO, PHONO 250~, PHONO NARTB-AES, PHONO 500~, and LOW FREQ. There was also a record compensation switch that had settings for HIGH FREQ, 0dB, 4dB, 8dB, 12dB AES, and 18dB NARTB. With all these options, one would expect to be able to find settings with which RIAA-equalized records would sound right. Not so! So soon I was in the market for something better. And I decided that it was time to finally make the move to stereo; after all, stereo records had been available for more than 20 years, and most of my albums—and all my reel-to-reel tapes—were stereo.

The Altec power amp was superb. Its push-pull pair of 6550s lazily put out a very clean 40W with distortion under 0.5% and a flat response from 20Hz to 20kHz. The Peerless power and output transformers were impeccable. But it was mono. A colleague at Western Electric, where I worked, sold me a pair of Heathkit W5Ms (**Photo 11**). These put out 50W each from a pair of KT66s in a Williamson circuit, with a frequency response that was only 3dB down at 3Hz, and low distortion when outputting 20W at 20Hz. Since my speakers were pretty efficient, I could get all the sound level my duplex neighbors could stand.

Having drifted away from watching TV, I made no effort to have a good TV set until DVDs became available for rental. Then I became interested in surround sound or video when the technology had matured sufficiently. This forced me to investigate buying a surround processor or a surround receiver. The processors were (still are) not easy to find, and quite expensive. But good solid-state surround receivers are available at an affordable price from several manufacturers. A few even have enough inputs that I can still use my H. H. Scott LT-111 vacuum tube FM stereo tuner (**Photo 12**).

Built from a kit, this tuner with its discriminator to detect FM sounds much better than the phase-locked-loop tuners included in today's surround receivers. I also managed to locate a surround receiver with enough inputs to handle my reel-to-reel tape recorder, a cassette player, and a DVD-VHS video recorder/player, while driving an LED video panel and a surround system including an outboard subwoofer amplifier. After all these years, do I finally have my ultimate home entertainment system? Probably not, but the system does quite well for now.

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- Well balanced SPL to Xmax ratio equates to a loud and long-throw sub
- Robust 18" ribbed paper cone for strength and durability
- Under spider and rear periphery venting allow for a cooler running voice coil
- 1,200W RMS/ 2,400W Peak Power Handling

Odeum Apollo 18F

- Best 4" voice coil ferrite subwoofer value on the planet
- Aluminum demodulation ring for low distortion and low power compression

Odeum Apollo 18N

- Powerful neodymium motor for superior output
- Copper inductance cap and aluminum demodulation ring for low distortion and low power compression



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With a Dayton Audio 5 year warranty, it's more than a guarantee, it's a promise. We stand behind our products and our warranty is a demonstration of our commitment to quality and reliability.