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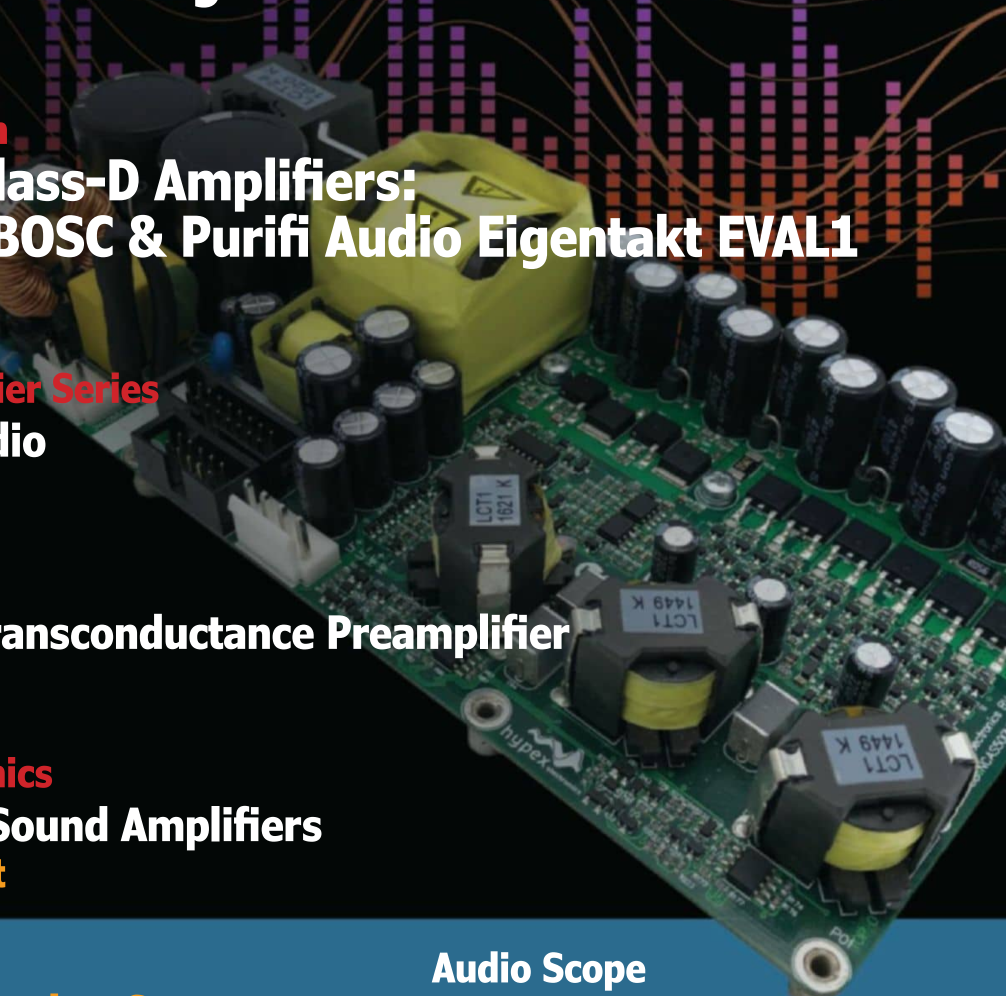

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JULY 2020

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Infineon

www.infineon.com/microphones

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Time to Embrace Evolution

Audio amplifiers—from the latest OEM amp modules and topologies to the most advanced and efficient devices and integrated circuits—target different applications, while developers focus on different requirements.

Meanwhile, there's a quiet revolution happening in audio amplification. Essentially, it is the direct result of strong innovation in Class-D technology, with incremental improvements introduced on multiple fronts, together with the demanding requirements of new high-volume applications in mobile and personal audio, wireless speakers, and automotive audio. The reality is, in the consumer space at least, there was never so much activity in the segment.

And multiple target applications is the reason why so many advanced products launched over the last decade, while showing technical merit and measurable benefits, very seldom have we found widespread recognition by the audio industry outside of their intended target market.

Many times we have seen new products that benefited from advanced performance in specific criteria (e.g., energy efficiency, dynamics, or distortion) being criticized as disappointing over subjective criteria, such as perceived sound quality. And that is the reason why we can find products—in totally extreme segments—where inventors and/or manufacturers have not fully explored the product's unique propositions and advantages.

That is also the reason why so many great Class-D amplifiers have reached the market and were poorly received by demanding users in the high-end audio segment, where the criteria for evaluation is frequently subjective and hard to confront.

As Stuart Yaniger writes in this edition, for far too long, Class-D amplifiers were quickly disregarded with statements such as: "It's improving, but they're not there yet." In *audioXpress*, we like to evaluate things objectively and that is why we review some of the latest products that serve to show how things are not only improving but reaching a level of perfection that is even difficult to measure with standard lab-quality audio analyzers.

As Yaniger also writes this month: "...it is no secret that I believe an amplifier's job is to make a small signal larger—things like "conveying the emotion of music" are, in my view, best left to the musicians and producers. So the best amplifier is one that has no sound of its own."

I understand that some of our readers will disagree with this view, and that is precisely why the audio industry still has room for manufacturers who sell $50,000 USD analog monster-monoblocks that remained basically the same designs for at least 30 years—and whose clients continue to praise them as the best thing there is, because they "deliver exceptional musicality," while other companies are now focusing on the latest "smart" audio systems, which of course include an advanced stereo amp with integrated DSP, the highest power efficiency and all the connectivity needed to enable applications for today's market and lifestyles—while costing less than $40 because they target product applications manufactured in the millions, not hundreds.

And that is why we also embarked on a visit to Roskilde in Denmark, to show a bit more about audio development house Purifi Audio, which is a great example of a new company working on the cutting edge to break the barriers of audio fidelity and elevate the threshold of what high-end audio can be—objectively.

As our readers can also attest this month, even a DIY project from an expert such as Dimitri Danyuk—who has been the principal hardware engineer at Harman Luxury Audio since 2013—serves to show how amplifier performance can be significantly improved when there's an understanding of what was previously done, and what we are trying to achieve.

In this month's market update, we also discuss a wide gamut of breakthrough solutions in audio amplification, from new-generation devices designed to boost the audio quality and efficiency of phones, wireless speakers, and wearables, to truly smart amplifiers that are meant to solve the challenges of speaker management.

And to add a valuable perspective, Richard Honeycutt travels into the past to reflect on the early inventions that formed the foundation of the live sound equipment we have today—and serves to show how much audio amplification has evolved.

J. Martins
Editor-in-Chief

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

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

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

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

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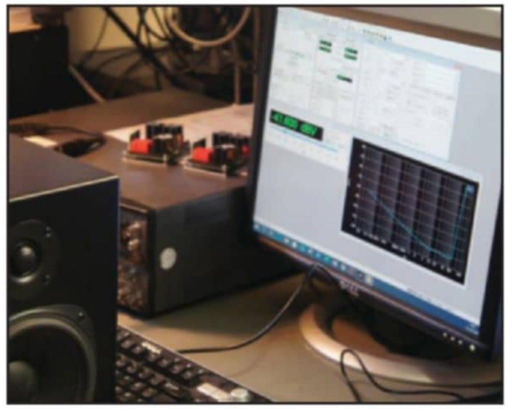

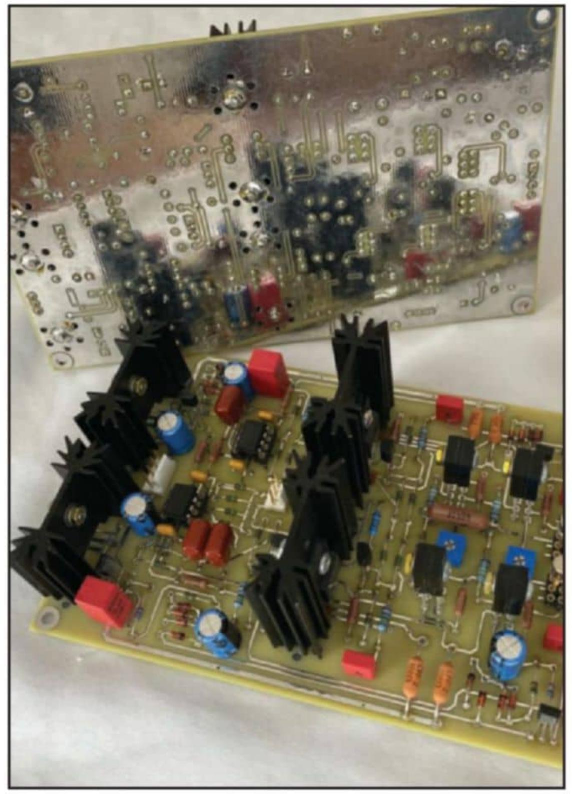

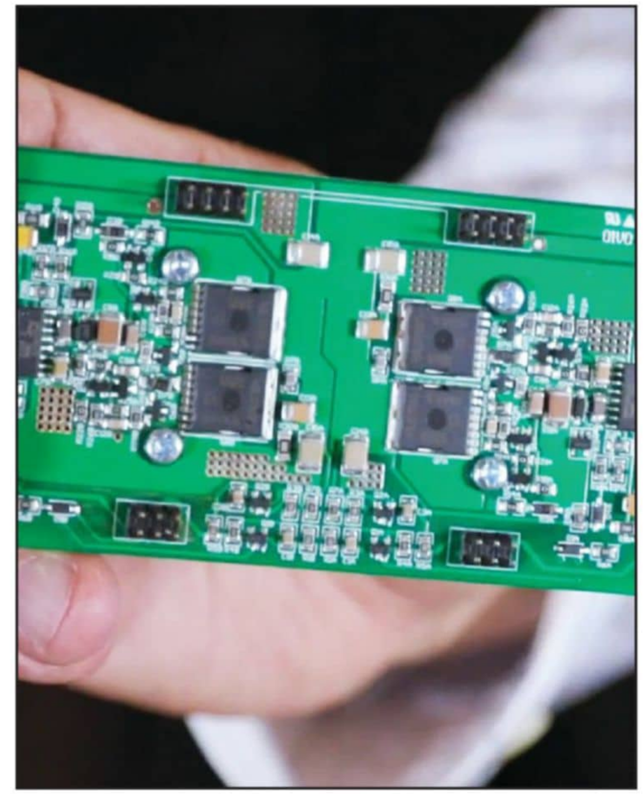

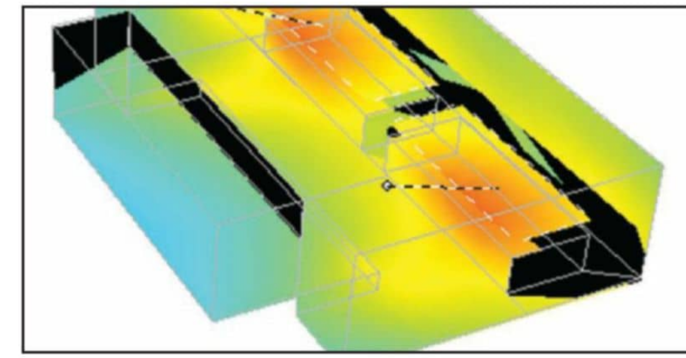

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Class-D Products Getting Smarter

By
J. Martins
(Editor-in-Chief)

Audio amplifiers are no longer an isolated product category. They are currently at the core of every audio product and system design and they need to be smarter, more integrated, and more efficient than ever. This Market Update covers the latest trends in new Class-D solutions.

There's a quiet revolution happening in audio amplification. Essentially the direct result of strong innovation in Class-D technology, with both incremental and exponential improvements introduced on multiple fronts, it matches the demanding requirements of new high-volume applications in mobile and personal audio, wireless speakers, and automotive audio. The reality is, in the consumer space at least, there was never so much activity in the segment.

From another perspective, over the last five years, the audio industry has seen new-generation Class-D products completely take over pro audio applications. They now have much more power efficient designs and the ability to manage variable power distribution, even in the most extreme configurations—with more output power, an increasing number of channels, and the output voltage required by the latest generation of high-performance loudspeakers. Thanks to these new Class-D products, professional audio was also able to transition to fully networkable, network-controllable systems with fully integrated low-latency signal processing of input sources, channel mixing, and routing to full management and monitoring of the connected speakers.

And even though the output power, number of channels, and number of speaker boxes have been steadily increasing in touring and large entertainment installations, even the largest, most complex system can be controlled from a simple smartphone or tablet app.

On the other extreme, there are increasing requirements for extremely compact and power efficient solutions for mobile devices and wearables, where the increasing role of new integrated platforms benefits immensely from completely new Class-D chipsets. High-volume applications in areas such as automotive, smart home systems, and personal audio (headphones and earphones) leave no other option than to fully manage DSP, output (speaker/driver), and interfaces. But the dominant discipline is clearly in smartphone audio, where system-on-chip (SOC) or chipset solutions are combined with audio codecs, voice coprocessors, and power amplifiers. It is an area in which the big players such as ESS, AKM, Cirrus Logic, NXP Semiconductors, Maxim, Qualcomm, and Infineon are working to deliver the goods on audio circuit performance.

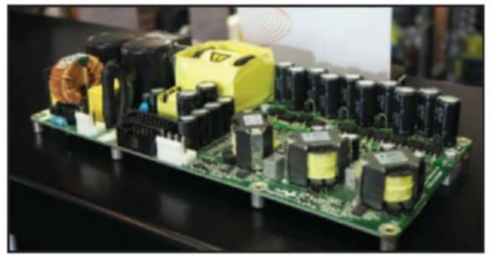
From the examples selected for this market update, we can see a clear trend for a new generation of "smart amplifiers" that are able to optimize an audio system on all the required fronts, including the most sophisticated integrated DSP being able to run the last generation enhancement and protection algorithms—with the lowest power consumption and in the smallest package.

Hypex Electronics

New NCore NCAS500MP Amplifier for Two-Way Active Speakers

After years of development, Hypex has launched the NCAS500MP amplifier module, the first member of an all new NCORE family of amplifiers specifically designed for two-way active speakers. From professional audio solutions to high-end hi-fi speakers to studio monitors, the NCAS500MP can be used for all active speaker designs. With a power rating of 400 + 100 W in 8 Ω, it is a very powerful solution for its size.

As Hypex explains, the new NCAS series has many similarities to the company's NCxxxMP series, using the same standby power supply (meets 2013 ERP Lot 6 0.5 W requirements), universal mains and is completely pin compatible by using the H-Box and H-Bus connectors. But while the NCxxxMP series is designed for multipurpose solutions, the NCAS family was specially designed for active speakers so the footprint and the price can be kept to a minimum. And manufacturers can combine the NCAS500MP with the Hypex MP-DSP solution to get a complete active speaker setup.



NCAS500MP is the first in a new family of NCORE modules optimized for two-way active speakers.

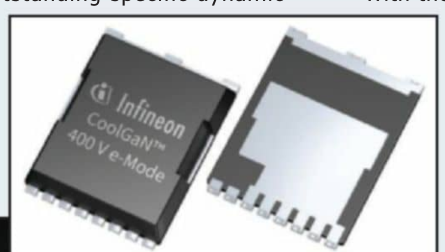
This means that the Hypex NCore's audio quality, renowned in high-end circles, is now available to all audio markets regardless of budget. And the NCAS family will further expand that goal. Future family members to be available soon include the NCAS250MP (200+50 W) and NCAS1000MP (800+200 W).

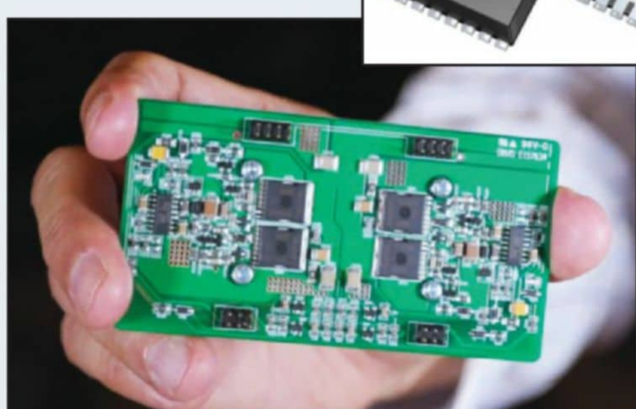
www.hypex.nl

Infineon Technologies

CoolGaN Portfolio Expands with New Devices Tailored for Premium Audio Systems

Infineon broadened its CoolGaN series with the CoolGaN 400 V enhancement mode high-electron-mobility power transistor (e-mode HEMT). Gallium nitride (GaN) offers fundamental advantages over silicon, with outstanding specific dynamic on-state resistance and smaller capacitances compared to silicon MOSFETs. The new CoolGaN 400 V device will appeal to audio product designers looking for a premium solution for home or professional Class-D amplifiers, and car audio systems that are able to translate high-resolution audio material, applications addressed typically by bulky linear or tube amplifiers.





To simplify product design, Infineon pairs the CoolGaN 400 V device in an HSOF-8-3 (TO-leadless) package with a popular Class-D controller (IRS20957STRPBF) in an evaluation board.

With the CoolGaN 400 V switch as a Class-D output stage, audio designers are able to deliver an excellent listening experience that will rival top quality designs.

The CoolGaN 400V e-mode GaN HEMT is a derivative of the industry benchmark CoolGaN 600V technology. This enhancement-mode power transistor (IGT40R070D1 E8220) overcomes the technology barriers (e.g., linearity and power loss) by introducing zero reverse recovery charge in the body diode and very small, linear input and output capacitances.

The CoolGaN 400 V switch enables smoother switching and more linear Class-D output stage by offering low/linear C_{oss}, zero Q_{rr}, and normally-off switch. Ideal Class-D audio amplifiers offers zero distortion and 100% efficiency. What impairs the linearity and power loss is highly dependent on switching characteristics of the switching device. The CoolGaN 400 V breaks through the technology barrier by introducing zero reverse recovery charge in the body diode and very small, linear input and output capacitances. The resulting benefit to the end user is a more natural and wider soundstage audio experience.

The CoolGaN 400 V (IGT40R070D1 E8220) device and EVAL_AUDAMP24 evaluation board can be ordered now.

www.infineon.com/gan

Cirrus Logic

Low-Power Boosted Smart Audio Amplifiers

To support the growing trend toward stereo audio in smartphones and portable devices, Cirrus Logic offers the Cirrus Logic CS35L41 boosted smart power amplifier, based on a 55 nm process design. This Class-D amp features predictive algorithms, where the power amp increases the audio volume when in speakerphone mode, while reducing noise and power consumption.

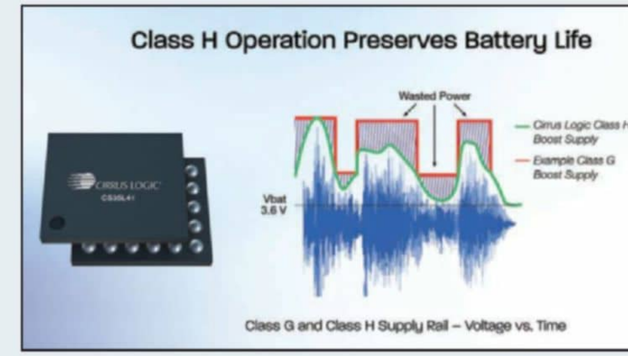
The Cirrus Logic CS35L41 11 V boosted Class D audio amplifier with DSP is complemented with Cirrus Logic's SoundClear Playback software, which improves audio quality and increases the loudness of the speaker output while protecting the speaker using both hardware and software techniques. Its advanced battery management system and predictive algorithms adapt to changing audio, speaker, and battery conditions to minimize power consumption and battery current, without sacrificing audio performance.

The core of the Cirrus Logic CS35L41 features a 5.3 W digital input, mono Class-D amplifier with the lowest noise and idle power consumption in its class, combined with an integrated 11 V Class H DC-DC converter that boosts the supply voltage and maintains higher efficiency than other audio power amplifiers relying on Class-G boost regulators. By adaptively tracking the audio level, the Class-H boost converter helps improve system efficiency, minimizes power dissipation and preserves battery life.

In addition to its Class-H boost and super low 6.7 mW quiescent current, the CS35L41 preserves battery life using an advanced battery management system. By predictively managing the demand on the battery, the amplifier is able to maintain maximum loudness and audio quality, while protecting the battery by minimizing battery current draw and eliminating unexpected "brown out" conditions that shut down the phone.

The CS35L41 is available in production volumes in a WLCSP package. The company also offers tuning tools for OEM support as well as mid- and lower-cost designs.

www.cirrus.com/products/cs35l41



The Cirrus Logic CS35L41 smart power amplifier integrates an 11 V Class-H DC-DC converter that boosts the supply voltage and maintains higher efficiency than other audio power amplifiers that rely on Class-G boost regulators. The Class-H boost converter adaptively tracks the audio level to improve system efficiency, minimize power dissipation and preserve battery life.

Qualcomm Technologies

Single-Chip Qualcomm CSRA6640 DDFA Amplifier Solution

With the CSRA6640, Qualcomm Technologies offers an highly integrated single-chip amplifier solution with Qualcomm DDFA Digital Amplifier Technology. This architecture brings a new level of integration to make Class-D amplification more commercially viable on smaller form factors and lower-tier products, helping manufacturers create portable and power-efficient speakers that offer differentiated audio quality.

The CSRA6640 was essentially designed to enable audio OEMs to optimize their materials costs by reducing the need for additional external components.

Built on more than 10 years of R&D, the solution was designed to almost perfectly balance the power efficiency of Class-D devices alongside the high-performance audio output of traditional, linear amplifiers.

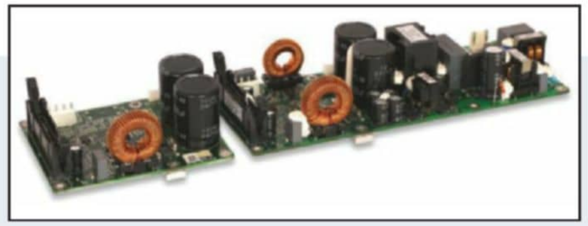
To further simplify product scoping and integration, the CSRA6640 can be supplied with an evaluation kit, including the necessary hardware, software, development tools, and information needed. Designed to support developers, including those who may not have specific amplifier expertise and help them to integrate DDFA capabilities into their products more easily, the kit is directly compatible with several of Qualcomm's other speaker, smart speaker and soundbar platforms. The evaluation kit offers a unique modular architecture to support streamlined scoping of both the CSRA6640 and the CSRA6620 amplifier chips with DDFA, as well as supporting the ability to simply scale up to 12 channels being evaluated simultaneously.

www.qualcomm.com

The CSRA6640 chip expanded Qualcomm's DDFA technology to lower power applications

Pascal

U-A1 Extension Modules for Studio Monitor Designs



U-A1/U-A1D and U-PRO amplifier combinations offer ultra-compact solutions for three-way loudspeaker designs.

The new Pascal U-A1 and U-A1D amplifier extension modules complement the company's solutions offerings to include three-way studio monitor designs, augmenting its U-PRO series modules to provide an additional high-frequency channel power option. The U-A1 and U-A1D are single-channel amplifier modules without Switch Mode Power Supplies, directly interfacing with U-PRO series modules (from which they draw mains power) to offer three-channel amplification configurations.

The new U-A1 and U-A1D "extensions" offer 280 W and 140 W RMS ratings, respectively, with ±70 V high voltage rails providing ultra-high dynamic range and extensive headroom, for ultra-clear, low distortion and improved sonic performance.

With performance characteristics precisely matching the U-PRO series modules, specifications extend to 119 dB dynamic range and the lowest distortion performance for comparable pro audio amplifiers.

With the U-A1 and U-A1D modules measuring half the length of U-PRO series modules, the U-PRO + U-A1 or U-A1D combinations offer space saving solutions for compact loudspeaker designs. Three-way monitor design projects can also benefit from the company's UREC Power Factor Correction (PFC) power supply technology—enabling universal AC mains operation with consistent regulated power worldwide, minimizing the requirement for heatsinks and cooling, and contributing to the amplifier's long-term reliability.

Supporting modern connectivity requirements, U-PRO modules also provide an auxiliary power feed for integrated DSP and network or analog I/O boards. Readouts of protect/mute, temperature, and clip signals are accessible for DSP and network or wireless control implementations, and I/O stages. The U-PRO series features ultra-low standby power consumption for EuP2013 and green Energy Star compliance, with an Auto Standby/Wake-up feature with selectable time settings, making the module ready to handle future regulatory requirements.

www.pascal-audio.com

Maxim Integrated Products

MAX98390 Smart Amplifier with Dynamic Speaker Management

The miniaturization of consumer devices requires speakers to fit within smaller form factors, which has led to more applications moving toward the use of microspeakers. As speakers shrink, loudness or sound pressure level (SPL) decreases while the resonant frequency increases, leading to less bass. Driving the speakers harder to increase loudness and bass response can easily damage the microspeakers through overheating and over excursion.

The new MAX98390 Smart Amplifier solves this challenge by utilizing integrated IV (current and voltage) sense and Maxim's DSM algorithm to drive speakers to their maximum specified limits, while protecting against over excursion and over-temperature events. DSM's thermal protection empowers designers to safely push their speakers well beyond their specified power rating, enabling the speakers to produce their maximum loudness. DSM's excursion protection enables designers to drive speakers to their specified excursion limits, producing sound up to 2 octaves below the resonant frequency limit. Effectively, the MAX98390 delivers up to 2.5× loudness (sound pressure level) and up to 2 octaves deeper bass vs. conventional 5 V amplifiers in a small form factor.

To properly protect the speaker, amplifier algorithms must know the characteristics of the speaker (e.g., resonant frequency within its enclosure, excursion limit, and voice coil thermal limit). Traditionally, designers had to go through the time-consuming and complex characterization process or rely on direct supplier support to characterize their speakers and enclosures. This challenge is further exacerbated as most projects start prototyping with multiple speakers per project, requiring several weeks of supplier support or requiring special equipment and expertise.

The MAX98390 significantly reduces the design time through the easy-to-use DSM Sound Studio GUI that enables users to quickly and easily characterize many speakers. Combined with DSM's thermal protection, the result is maximum loudness across the significantly extended frequency ranges in minutes, with no complex programming required. To cater to the miniaturization of devices and shrinking batteries, the MAX98390 also offers industry-leading peak efficiency at 86%, which is even further improved with DSM's Perceptual Power Reduction (PPR) feature that can yield up to an additional 25% efficiency, and the lowest quiescent power consumption of ~24 mW, making it the ideal choice for low-power devices that need longer battery life.

The MAX98390 is a high-efficiency mono Class-D DSM smart amplifier that features an integrated boost converter, integrated Dynamic Speaker Management, and field-effect transistor (FET) scaling for higher-efficiency at low output power. The maximum boost converter output voltage is programmable from 6.5 V to 10 V in 0.125 V increments from a battery voltage as low as 2.65 V. The boost converter supports bypass mode for lower quiescent current and improved mid-power efficiency as well as envelope tracking, which automatically adjusts the output voltage for maximum efficiency. The boosted supply efficiently delivers up to 6.2 W at 10% THD+N into a 4 Ω load.

Maxim recently introduced the MAX98390—a boosted, digital Class-D DSM smart amplifier that is able to safely drive higher power levels (up to 5.1 W) into tiny speakers typically rated for much lower power between 1 W to 3 W.

The PCM interface supports I²S, left-justified, and 16-channel TDM formats as a slave or master device with I²C control. Either BCLK or MCLK can be used as the internal clock source providing system level flexibility. Patented active emissions limiting edge rate and overshoot control circuitry minimizes EMI and eliminates the need for output filtering found in traditional Class-D devices. A flexible brownout-detection engine (BDE) can be programmed to initiate various current limiting, signal limiting, and clip functions to prevent dips in battery voltage. Threshold, hysteresis, and attack-and-release rates are programmable.

The MAX98390EVSYS# evaluation kit is available for $200.

www.maximintegrated.com

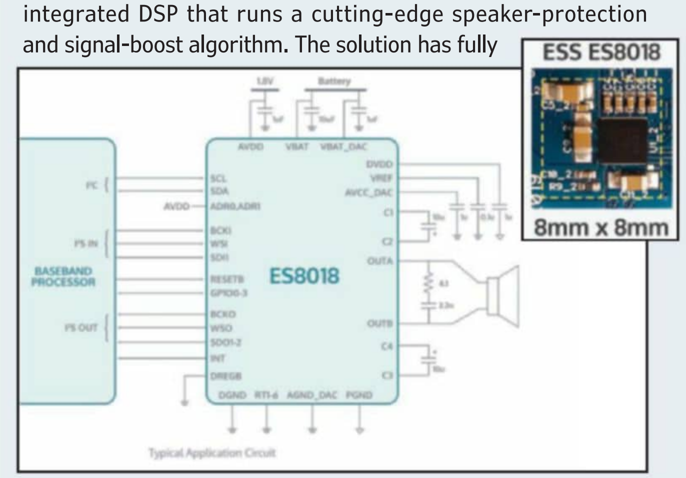
ESS Technology

New Smart Power-Amplifier Technology

ESS Technology launched the ES8018 Sabre Smart Power-Amplifier (SPA), a new-generation device designed to boost the audio quality and efficiency of phones, tablets, wireless speakers, and even the smallest connected wearable devices. According to the California company, the ES8018 combines major advances in amplifier design with advanced speaker-protection algorithms, creating a new integrated amplifier design in the smallest solution available currently to the industry.

The ES8018 is based on a new patented seven-point multi-level Class-D amplifier topology offering improved efficiency and lower EMI than standard amplifiers. These advances are coupled with the renowned sound quality of ESS Sabre DACs, and advanced audio processing algorithms. As a result of this combination, audio product designers can obtain amazing audio quality from microspeakers while having a small and cost-effective solution.

The ES8018 is a self-contained Smart-Amplifier with integrated DSP that runs a cutting-edge speaker-protection and signal-boost algorithm. The solution has fully proven temperature and excursion protection that adapts to the changes in real time, and is capable of safely delivering 4.5 W into an 8 Ω speaker, and can be used to drive both microspeakers and receivers in a dedicated low-noise mode.



Typical ES8018 application circuit. The combination of the Voltage-Tripling boost and MLCD amplifier in the ES8018 requires 40% less power consumption in real-world applications.

The ES8018's unique voltage-tripling boost is based on using the combination of a voltage-doubler and voltage-mirror circuit. This voltage-tripling results in an effective boost-voltage of 13 V from a typical phone battery, and can reach 15 V from a 5 V supply. Both the high and low-side boost are accomplished using captive techniques, eliminating the need for a large and expensive inductor, replacing it with two low-cost and small capacitors, resulting in a solution that requires 40% less board space and has a significantly lower profile. The advanced analog control loop in the amplifier automatically corrects for any loss of power due to battery discharge preventing impacts to audio quality, while simultaneously preventing brown-out due to large currents.

The patented Multi-Level Class-D (MLCD) amplifier in the ES8018 tracks the amplitude of the audio signal and seamlessly switches between seven different switching patterns on the output. The output effectively floats along with the signal. The result is an amplifier that can deliver peak-output swings of 15 V while only ever switching 5 V. This gives improvements in efficiency and lowers EMI radiation. Unlike competing amplifiers that rely on a fixed boost circuit, the ES8018 never switches the full 15 V, avoiding significant EMI and efficiency-loss. This is critical for real-world audio signals with large crest factors. The digital control loop, used in the competitors inductive-boosts, requires a look-ahead approach to charge the converter and wastes power when idle or turned-off, the ES8018's analog control loop has none of these penalties, instantaneously providing peak-power.

The ES8018 amplifier is available now with a full application suite to enable effective speaker modeling and sound-quality tuning. Samples and evaluation boards are available.

www.esstech.com

ICEpower

200AS2 Compact Amplifier Module

Ideal for active speakers, studio monitors, and stereo amplifiers, the ICEpower 200AS2 amplifier module, part of the AS amplifier family, features a capable 300 W fly back power supply, combined with a 2× 215 W Class-D amplifier.

In a time when active speakers are increasingly becoming the norm the ICEpower 200AS2 amplifier module is able to support applications as diverse as bass or guitar amplifiers to a high-end amplifier component. But active speakers come with their own set of challenges. Adding power electronics inside a compact, often wooden enclosure, demands a lot in terms of heat management and dissipation. Frequently, space is also at a premium. The ICEpower 200AS2 was specifically designed for enabling maximum output power—2× 215 W Class-D (4 Ω)—in small compact form factors with excellent sound quality. This has been achieved by utilizing a power supply capable of sustaining high continuous power levels at elevated ambient temperatures with minimal cooling needed.

In many designs, the 200AS2 avoids the need for fans, enabling quiet operation. This is an advantage in recording studios or in a home stereo environment where the user wants to focus on the details of the music without a noisy fan blurring the picture. The 200AS2 module also features a capable 300 W fly back power supply, designed with universal mains and includes ErP and Energy Star-compliant standby functionality, regulated auxiliary power supplies, wake on signal sense and a DC-bus output, short circuit, and overcurrent protection. The DC-bus output is intended to power a separate tweeter amplifier, for three-way applications.

The ICEpower AS Series of amplifier modules, from 100 W to 1200 W per channel.

The 200AS2 has been designed to speed up time-to-market and reduce the product development cost for manufacturers.

The length and width are just 158 mm and 77 mm, enabling it to fit on the backside of compact active speakers. The height is also limited to 36.5 mm, enabling the module to fit in simple 1U rack amplifiers. And the 200AS2 module is, as are all ICEpower modules, fully EMC and safety pre-approved.

Similar to the other modules in the ICEpower AS series, the 200AS2 has been designed for maximizing power, performance, and reliability, while keeping the cost at a level that makes the technology available to a broad scope of applications.

Utilizing the same topology as the existing 100AS1, the 100AS2, and 200AS1 modules, the 200AS2 has the same input voltage sensitivity, simple mounting-hole pattern, electrical interface, and connectors. These features make it easy for manufacturers to create a series of products using the same power platform. The use of a regulated power supply topology ensures consistent power performance globally across both low and high mains.

www.icepower.dk

SpeakerPower

The Go-To Amplifier Company in California

Since 2002, SpeakerPower has been helping pro audio loudspeaker manufacturers become powered speaker manufacturers, and custom-installation companies gain flexibility and confidence for their projects. From rack mount to plate amplifiers to custom-made modules, SpeakerPower designs, assembles, and tests its audio amplification products at its home base of Santa Ana, CA. Offering built-to-order Class-D designs from 200 W to 12,000 W and constructed with components sourced entirely from local vendors, SpeakerPower products qualify for the "Made in USA" label. For the home theater market, the company offers an extensive range of subwoofer amplifier plates, optimized for cinema and music. Custom faceplates, silkscreens, and other features are easily accommodated.

Some models have UL/CSA approval as a standalone product—while UL approval is not easy or cheap, SpeakerPower made the investment for the benefit of its customers. This means users don't have to pay for an expensive UL investigation, quarterly inspections of manufacturing, calibration of test equipment, and so forth. The combination cUL mark is already screened on the faceplate of SpeakerPower products.

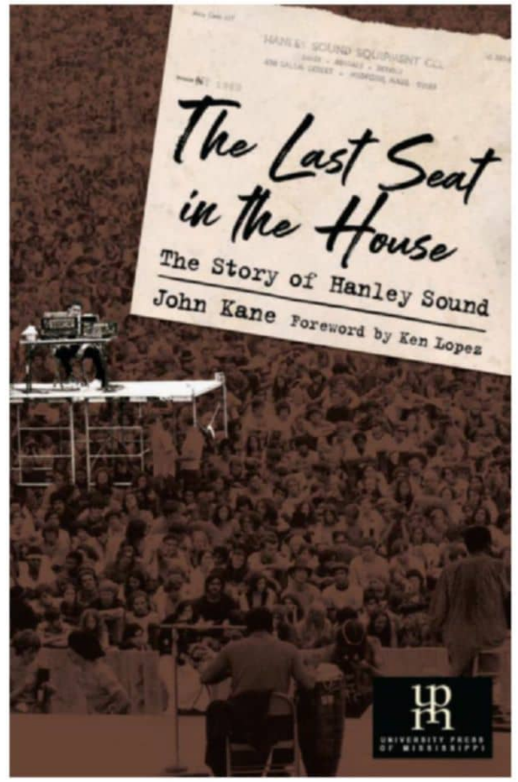
Products use the highest power amplifier modules available and can offer full-featured remote control and monitoring, as well as digital audio networking.

In 2018, Justin Ryan, CEO of SpeakerPower, acquired the company from its founder, Brian Oppegaard, upon his retirement. As an experienced executive in a variety of industries, and an MIT-trained engineer, Ryan is determined to invest in product development and further expand SpeakerPower's commitment to deliver high quality, efficient, and intelligent amplifiers.

www.speakerpower.net

A Look Inside
The Last Seat in the House—The Story of Hanley Sound

In this article, Richard Honeycutt reviews John Kane's book, which examines how sound engineering pioneer Bill Hanley's contributions in outdoor live sound helped create the entire music production industry. Through an extensive nine-year interview and archival process, Kane uncovered how Hanley and his company, Hanley Sound, impacted a developing festival and concert marketplace.



Review by
Richard Honeycutt
(United States)

In today's world, we take live sound and public address systems for granted. It has not always been so. Although the first electrical public address system was demonstrated in 1915 (Jensen/Pridham), and such systems have been used occasionally since then, in 1946 when third-grader Bill Hanley of Medford, MA, fell in love with electronics and audio, good sound reinforcement at concerts or even at public speeches was the exception rather than the rule. Sound Guru Don Davis's book title, *If Bad Sound Were Fatal, Audio Would Be the Leading Cause of Death*, indicates that the same could have been said in 2004.

Yet undeniably there have been major improvements in the quality of reinforced sound in the years since 1950, and much of the credit for this improvement rightfully belongs to Hanley. Hanley and his crew provided sound systems at the Newport Jazz, Folk, and Opera Festivals, and Fillmore East. They covered concerts by most of the best-known 1960s rock bands including the Beatles and Jefferson Airplane, and the Lyndon Johnson inauguration. Hanley Sound was the go-to sound company for the huge rock music festivals of the 1960s and early 1970s, culminating in Woodstock.

The new book *The Last Seat in the House* by John Kane tells the story of this immensely successful self-trained engineer with an unmatched ear and instincts for good sound in a very human manner. The author is a faculty member in the Design and Media Department of the New Hampshire Institute of Art. He recounts the history of Bill Hanley and his company that literally created the modern concert-sound industry during the decades of the 1950s and 1960s. The story begins with Hanley's early development of his passion for good sound. Then it is organized topically, rather than chronologically, although the overall thread generally moves with Hanley's history. So readers will occasionally find themselves having jumped back a few years in mid-chapter. This stylistic feature is easy to get accustomed to, once the reader figures out what is happening.

Kane's purpose was not to write a technical journal of the evolution of live sound systems, although he does list some of the equipment models Hanley used for various venues through the years: the Altec A7 and the Klipsch La Scala speakers, the McIntosh 275, and the Crown DC-300 amps, for example.

An interesting perspective is presented by the author's listing the emergence of other major sound companies during Hanley's career: Stanal Sound in 1962, Clair Brothers in 1966, and Showco in 1970, to name a few. Without exception, the founders of these companies credit Hanley for his pioneering ideas in providing good sound for large venues.

Although Hanley used primarily horn speakers for bass, midrange, and high frequencies, Kane quotes him as saying that the speakers had to be stacked high, because such stacking "compresses the sound" [into the audience area]. In today's jargon, we would say that the tall stacks provide directional control down to bass frequencies, putting the sound where the people are, rather than wasting it by shooting over the heads of the audience and thereby creating complaints from neighboring residents. This is the exact principle behind line arrays—pretty much the standard speaker configuration used today for large indoor and outdoor venues. Hanley was aware of the importance of correct aiming to provide even coverage without creating interference that would mar the sound.

Like many other sound companies, Hanley Sound built many of its own speakers and even its own mixers. In fact, Kane recounts the story of the custom sound console that Hanley and his employee Harold Cohen collaborated on for Woodstock. Despite heroic efforts including transporting the console to Yasgur's farm by motorcycle, the console was not operational in time for the show.

Anyone interested in the personalities, vibes, and technical advancement of the jazz, folk, rock, and even classical music industry from the 1950s through early 1970s will find *The Last Seat in the House* fascinating. Kane has done an admirable job of transporting the reader back in time to a very complex and swiftly changing era, rife with technical, societal, political, and musical revolutions. Anyone who is active or interested in live sound will learn from this study of its roots. Those who lived through those times will find that the book evokes many memories and emotions. Those who have grown up since then will find it enlightening.

The Last Seat in the House
By John Kane, 2011
University Press of Mississippi, 2020.
www.thelastseatinthehouse.com

A Tale of Two Class-D Amplifiers

Orchard Audio BOSC and Purifi Audio Eigentakt EVAL1

Stuart Yaniger examines new amplifier solutions from two new disrupting audio companies. Purifi Audio from Denmark, founded by Bruno Putzeys, Lars Risbo, and Peter Lyngdorf, together with a remarkable engineering team; and Orchard Audio, with a product developed from the creativity and entrepreneurship of Leonid (Leo) Ayzenshtat, in this case exploring the latest gallium nitride (GaN) semiconductor technology.

By
Stuart Yaniger
(United States)

In perusing various audio websites and forums, I have seen much discussion about the refinement of conventional linear amplifier technologies. In reality, amplifier design became fully mature more than 40 years ago, with the methods to achieve audible transparency and the ability to drive various loads being well-understood. But that hasn't stopped a continuous stream of analysis, refinement, incorporation of new active devices, novel topologies, and endless arguments about things like Class A vs. Class AB, current vs. voltage feedback, feedforward, two-pole compensation, bipolar transistors vs. MOSFETS and so on.

Outside of the enthusiast niche (of which I freely admit to belonging), these are all dead issues. Not that they are fully understood and agreed upon by those chasing ultimate measured performance well beyond any questions of audibility, but that they are analogous to arguments over the best way to print a scale on a slide rule. Class-D amplification has taken the non-hobbyist market by storm. Though it's become a cliché to say, "The D in Class D does not mean 'digital,' it was just the next letter after C," it is not far from the truth to note that the D could just as easily stand for "disruptive."

Class D (a method of pulse width modulation, PWM) is not a new concept but it laid fallow for many years with only a scattering of commercial attempts to implement PWM circuitry in audio amplifiers, most notably by Infinity and Sony. But starting about a decade ago, the commercialization of Class-D amplifiers has gone from slow and sporadic to warp speed. The advantages are compelling, notably much higher efficiency (90% or higher), meaning smaller, lighter, and cooler packages, and freedom from artifacts such as crossover distortion and g_m doubling inherent to conventional Class-AB amplification.

Enthusiasts have grudgingly admitted that Class-D amplification has improved over the years, but the mantra was always, "It's improving, but they're not there yet." Of course, the performance bar to compete, not just sonically but also in the performance numbers wars, has been raised following years of analytical work and improvement by makers of Class-AB amplifiers. One example is the superb Cambridge Audio Edge W that I reviewed a few months back (see Resources), which had astonishingly low distortion and noise, and seemed to drive any load I threw at it with ease and imperturbability.

Two Class-D amplifiers—the Orchard Audio BOSC and the Purifi Audio Eigentakt EVAL1—spent some time in my listening room and on my lab bench and represent how close to perfection this technology has come. Each has innovative design features and performance well past any question of audible perfection and both packaging and power that deliver on Class D's promise. They take very different approaches to achieving high performance—one unit using novel devices, the other using novel topologies. I will consider each of these separately.

Orchard Audio BOSC Monoblocks

As we were going to press, we were informed that due to a trademark dispute with Bosch, the BOSC amplifier will be renamed STARKRIMSON later this year. But for our purposes here, we will still refer to it as BOSC.

For those on the cutting edge of high-efficiency power conversion, the semiconductor gallium nitride offers huge potential advantages compared to conventional silicon (see Resources). Although GaN has found wide application in optoelectronics, its use in power conversion and audio applications is much more recent. Compared to silicon, the electron mobility in GaN is higher, the temperature coefficient is lower, and the higher bandgap compared to silicon means a lower degree of thermally generated charge carriers when the devices heat up.

Translating these characteristics into application advantages, the higher charge mobility means that the substrates can be smaller, which means lower capacitances. Lower capacitances reduce switching time, allowing a higher switching frequency. Additionally, GaN's conduction does not involve minority carriers, so the reverse recovery charge (QRR) is essentially zero, a significant advantage over conventional MOSFETs in reduction of dead-time and increase of switching frequency. Dead-time is the portion of the cycle where both output devices in a totem pole arrangement are switched off to prevent "shoot through," where the two power rails get shorted by the RDS(on) of the output devices.

GaN transistor producers claim that the reduction in dead-time correlates with a reduction in harmonic distortion due to the output devices controlling the load through a greater portion of the signal cycle. The lower dead-time and higher carrier mobility should also increase efficiency, all other things being equal. Lower threshold voltages reduce drive requirements. And a faster switching frequency should theoretically be better suppressed by the output low-pass filter.

How does this work in practice? Enter Orchard Audio, the audio company run and staffed by Leo Ayzenshtat, an aerospace engineer who has specialized in PCB design. Orchard's initial products were a series of USB-based DACs available in different configurations (PCB-only for OEM, in a case with a headphone amplifier, and with and without streaming options). Recently, Orchard introduced the BOSC power amp (see **Photo 1**), a 150 W monoblock module with a separate power brick.

The BOSC is supplied in a rather plain black box with extruded side panels. The company name and logo are silkscreened in white on the top. Each monoblock has its own external power brick, which is actually somewhat larger and heavier than the amplifier module. The inputs are balanced XLR, and Orchard supplies XLR-to-RCA adapters for those whose systems are run strictly single ended. The speaker binding posts are sturdy and gold-plated. They accommodate bare wire, spade, and banana plugs, and are, I'm delighted to say, spaced at a standard 0.75" apart.

It should be noted that the outputs are differential. For the vast majority of systems, this is no issue, but beware if you have a headphone adapter or speaker switchbox that uses a common ground between channels.

Photo 1: The Orchard BOSC amplifier packs a lot of power into a very small box. The power supply is external.

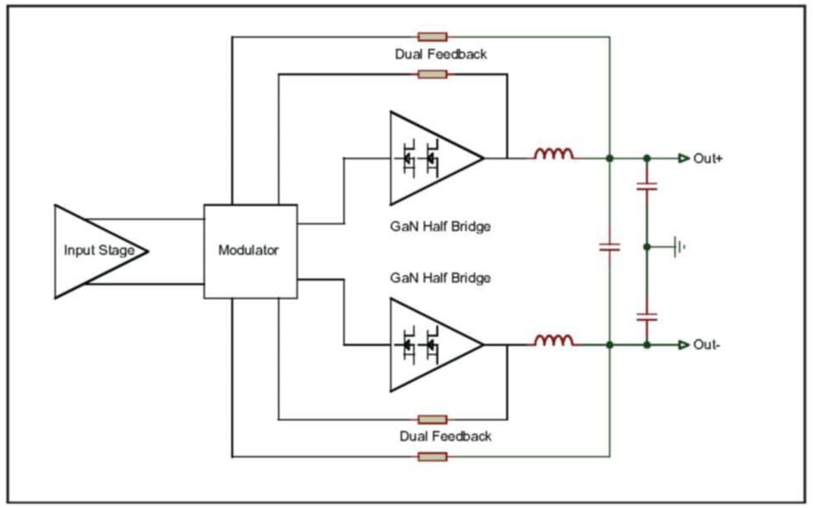


Figure 1: A block diagram of the Orchard BOSC shows its differential nature and the dual feedback loop system.

The nominal gain can be selected at the time of purchase at 16.8 dB (A = 6.92) or at 13.8 dB (A = 4.89), meant for 8 Ω and 4 Ω nominal impedance speakers, respectively. This is much lower than a typical home audio power amp gain of 26 dB or so, so don't be surprised if you have to crank up your preamp volume knob. If your preamp or source is rated for a 2 V output, you might have to either add a gain block or replace them with units having a higher output.

Also, the BOSC's input impedance is rather low, at 5 kΩ and 7 kΩ, respectively, for the two gain options. Most modern sources will have no trouble driving it, but older gear or tube sources could huff and puff a bit. On the output side, the damping factor is claimed to be 550, which translates to a commendably low-source impedance of about 15 mΩ. The value will be swamped by any practical speaker cable and certainly by speaker voice coil resistances.

The BOSC amps' circuit is based on a proprietary dual feedback self-oscillating design (see **Figure 1**). The amplifier is balanced and differential from input to output with fully bridged GaN output stages. According to Orchard, the output LC filter, which is used to covert the PCM output of the bridge to an analog signal, is a critical element in determining the sound quality of a Class-D amplifier. To this end, the BOSC amplifier uses what they term as "oversized ultra-high-quality Japanese inductors wound with oxygen-free copper, specifically made for Class-D amplifiers." The capacitor used in the output filter is a film capacitor, which Orchard says is sized larger than needed to help keep the amplifier's distortion very low through the whole powerband, and also help keep temperatures lower as efficiency increases. Presumably, the capacitors in this circuit position were also selected for low equivalent series resistance (ESR) to minimize heating and increase reliability.

The amplifier's higher-than-usual switching frequency combined with the output filter design is intended to result in minimal phase shift below 30 kHz. The amplifier is DC coupled, which minimizes phase shift at very low frequencies as well. Orchard claims that this results in a better-controlled bass response from loudspeakers.

The diagram also shows what Orchard calls a "dual feedback" system. The feedback to the pulse width modulator is taken differentially from both the half-bridge output and the speaker output, that is, from both sides of the output filter. The intention here is to minimize the effect of the output filter on the amplifier's response and distortion performance within the audio band.

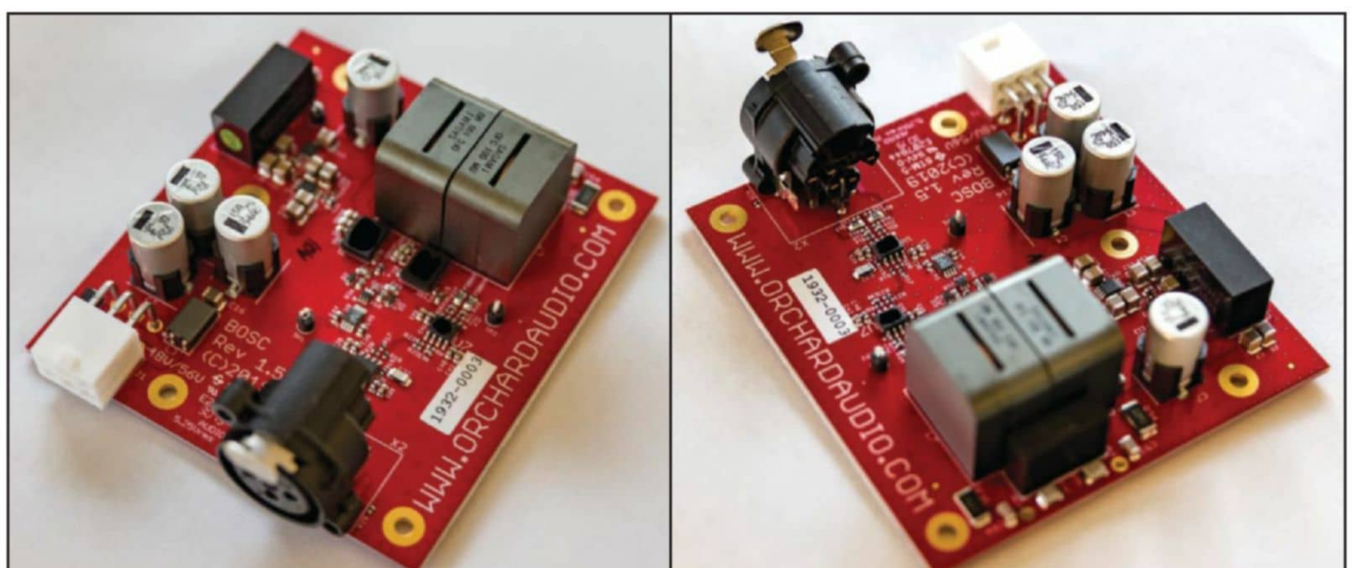


Photo 2: The PCB layout of the BOSC is simple and neat, with very high-quality boards and parts.

Since Ayzenshtat's day job in aerospace involves critical PCB design, it is unsurprising that he has gone to great lengths to optimize this aspect of the amplifier design and layout. The BOSC's four-layer PCB is state of the art, using custom stack up with high-end dielectric material and ENIG (gold) finish. **Photo 2** shows the populated board, which indeed appears compact and neat.

Unusually, the circuit is direct coupled from input to output. The modulator works with the feedback network to essentially eliminate the DC offset on the output. This DC offset is dependent on the input offset voltage of the op-amps used in the input stage, typically 50 µV, which given the amplifier's gain translates to an output DC offset below 5 mV. Since the amplifier is DC coupled, it will also amplify any DC on its input, so be careful when pairing these amps with sources that have direct-coupled outputs.

Purifi Eigentakt 400 Amplifier Demonstration System

As previously discussed in *audioXpress* (see Resources), Purifi Audio is a Danish company founded by some of the best minds in audio. One of the notable partners is Bruno Putzeys, who has done more than just about anyone else to refine Class-D technology to performance levels that would be nearly impossible to achieve with conventional linear amplification. The high audio performance is coupled with the high efficiency and the small size for which Class-D amps are noted. Following his successes with UCD and nCore amplifiers, he and his collaborator, Lars Risbo, accepted the challenge to take amplification in general to new performance levels, and to do so in a way that didn't destroy the superb economics of the previous generations.

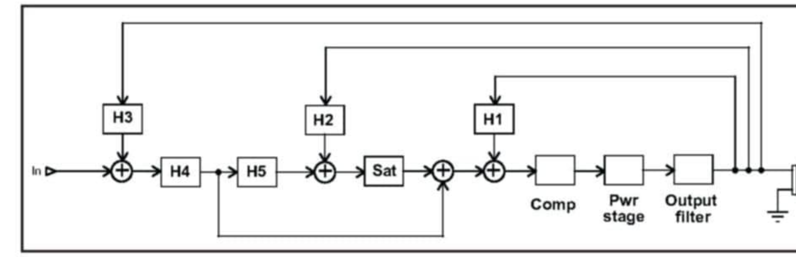


Figure 2: The Purifi amplifier system uses a combination of feedback, feedforward, and loop gain tailoring to achieve very high levels of feedback while maintaining stability.

A key to this is using extremely high amounts of feedback, which Putzeys explained nicely in an article a few years back (see Resources). Negative feedback lowers distortion and reduces output impedance, but applying large amounts is tricky to pull off while maintaining amplifier stability.

In a series of patent applications (see Resources), Putzeys and Risbo demonstrated methods to use multiple feedback loops with tailored responses to

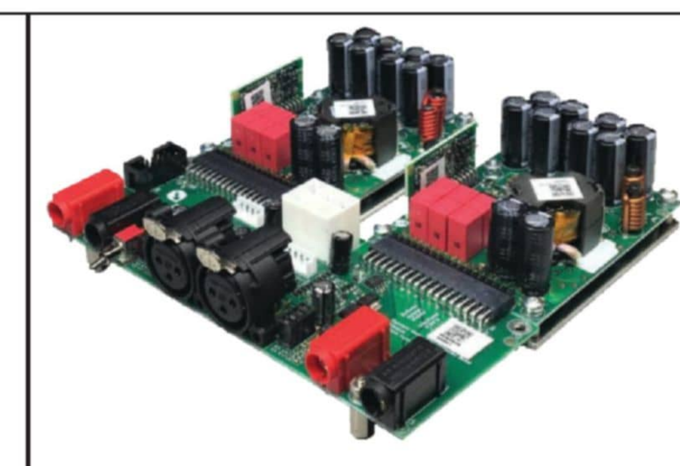


Photo 3: Purifi's EVAL1 system combines two 1ET400A power amplifier modules with a front end and control system, suitable for use as a stereo power amplifier.

Photo 4: The Purifi Audio EVAL1 chassis provides a convenient two-channel evaluation platform for the 1ET400A, combining balanced analog inputs (XLR) and speaker connectors that is also suitable for DIY projects.

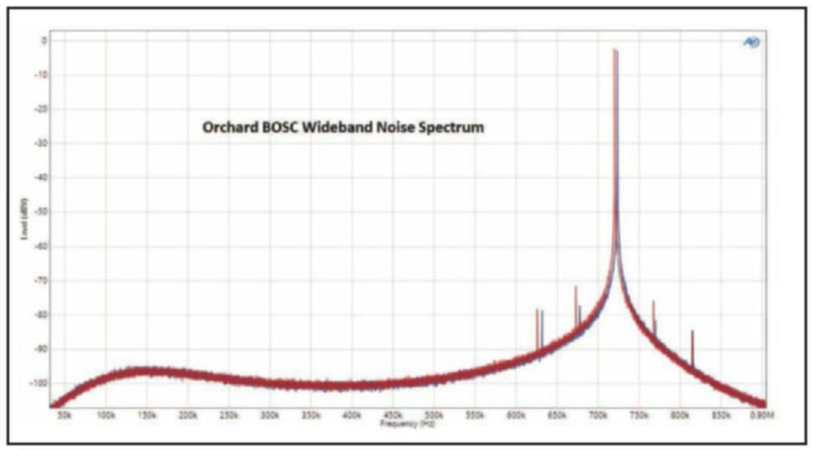


Figure 3: The no-signal ultrasonic noise spectrum of the BOSC is dominated by a low level of the switching frequency.

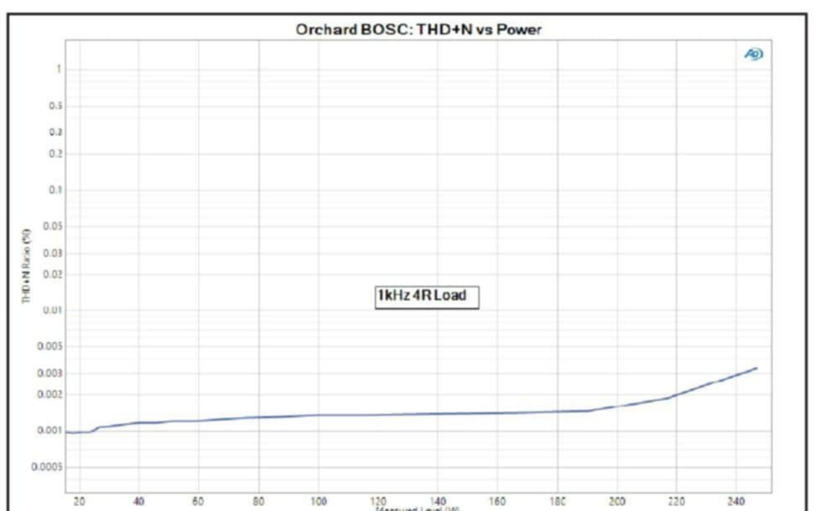


Figure 4: THD vs. power for the BOSC shows no signs of clipping up to 250 W.

extend the amplifier's gain-bandwidth product, which allows a large increase in loop gain. Thus, feedback can be massively increased to levels that are not possible with conventional amplification via manipulation of the response of each of the amplification blocks and the feedback and feedforward loops. **Figure 2** shows a block diagram of their amplification scheme. The various "H" blocks represent active or passive circuits with tailored transfer functions. For example, if a "classical" Class-D amplifier is represented by the combination of the comparator, the switching output stage, the output filter, and the feedback system H1, the amplifier can be nested within a secondary set of loops, which would include H2, H3, and their respective summing amplifiers, along with feedforward from filters H4 and H5, plus the compensator.

It's clear that there are a LOT of knobs to turn (metaphorically), and optimization of the overall system for flat response, low distortion, wide bandwidth, and unconditional stability is a rather daunting proposition. To this end, Purifi developed a sophisticated proprietary computer modeling system that can juggle all of the variables to reach a design target. I would highly recommend reading the patent applications referenced in Resources to get a good idea of the sophistication of Purifi's approach.

Purifi has released its 1ET400A OEM 400 W (4 Ω, 1% distortion) mono amplifier modules for manufacturers and DIYers to integrate into complete products. To demonstrate their performance, Purifi sent me its EVAL1 evaluation system, which consists of two 1ET400A modules and an FE02A front end system with all the analog input and the control circuitry needed to form a working stereo amplifier (see **Photo 3**). For review convenience, this particular demo unit was packaged with a power supply and in an aluminum case with I/O connectors, though these are not included in the normal EVAL1 system (see **Photo 4**).

Integrators have the option of using the FE02A or developing their own proprietary input circuitry. The bare 1ET400A module has a gain of 12.8 dB (A = 4.4), which combined with the FE02A input module's selectable 0 dB (actually bypassing the input amplification) or 13 dB gain results in an overall power amplifier gain of 12.8 dB (low) or 26 dB (normal). The input module, when not bypassed, also has a buffering function, increasing the input impedance from 4.4 kΩ of the bare 1ET400A to just under 100 kΩ, which in conjunction with the 26 dB overall gain can be driven by nearly every conceivable source.

So, how do these two different approaches end up working for the user?

The First Test: Listening Room

Before measuring the Orchard BOSC (I only had it for a few weeks, so I put it on the lab bench first), I hooked both the BOSC and the Purifi amplifiers up in my system. Currently, I am using a biamped setup in my lab, with speakers based on the classic NHT 3.3 four-way dynamics, but with modified crossovers and cabinet volumes. These speakers represent a relatively non-brutal but typical load (nominally 6 Ω), but are not terribly efficient at about 87 dB/watt/1 m. Crossovers are electrically a fourth-order Linkwitz-Riley (LR) low pass for the woofers and a second-order Butterworth high pass for the rest of the drivers, to achieve an overall fourth-order acoustic LR crossover at 110 Hz.

The electronic crossovers were, at different times, a Didden-modified Behringer DCX2496 (with the active option) and a miniDSP 2x4HD. Both of these units allowed me to adjust relative gains between the woofers and the rest of the drivers (mid-woofer, midrange, and tweeter). When using the Didden/Behringer, the amps were connected in balanced mode. When using the miniDSP 2x4HD, I used RCA-to-XLR adapters to drive the amps in unbalanced mode. Neither crossover has any difficulty with loads in the impedance range of the BOSC amps' inputs.

I switched the amplifiers' positions between driving the woofer and driving the rest of the drivers several times over the course of two weeks of listening and use tests. Now, it is no secret that I believe an amplifier's job is to make a small signal larger—things like "conveying the emotion of music" are, in my view, best left to the musicians and producers. So the best amplifier is one that has no sound of its own.

For both of these amplifiers, regardless of where they were inserted in my system, I could discern no difference in the sound, nor was there any difference compared to some pretty fine conventional linear amplifiers that I've used recently. No matter what sort of music I played—from acoustic jazz to Americana-folk to electronic progressive rock, whether my own minimally miked uncompressed recordings or highly produced loud studio rock—the amplifiers did exactly what they were supposed to, noiselessly, without stress or strain, and in a perfectly transparent way. Some might complain that this reduces the role of amplification to mere appliances—that's fine with me, I want to listen to music, not to my amplifier.

I should also note that neither amplifier got anything close to hot, even after extended listening sessions at brisk volumes. This is of particular

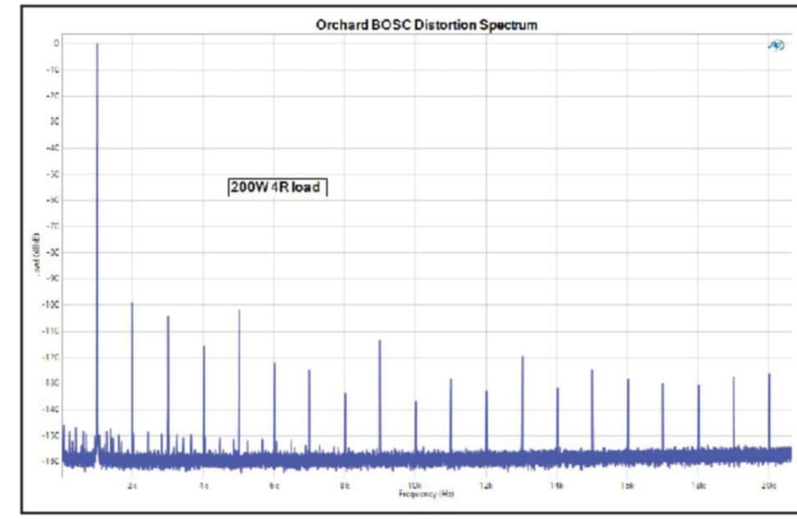


Figure 5: The distortion spectrum of the BOSC at 200 W into 4 Ω is extremely clean, with the highest harmonic at 0.001%.

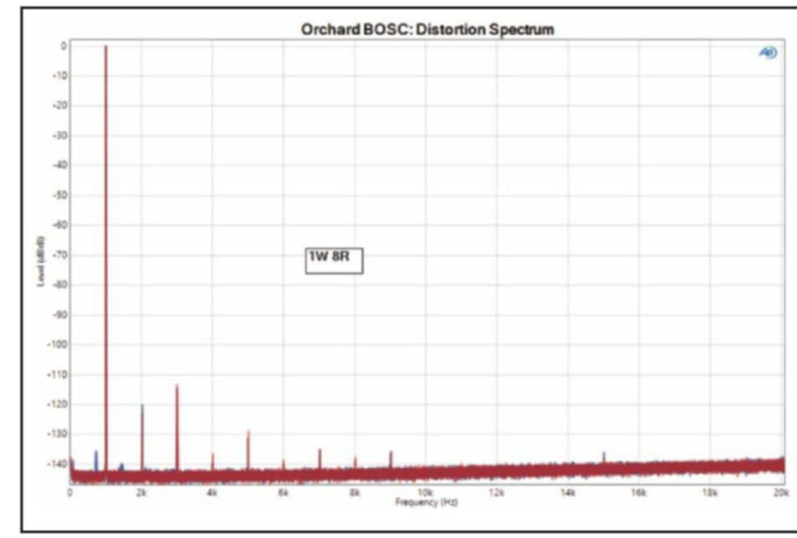


Figure 6: At 1 W into an 8 Ω load, the BOSC's distortion is below the Audio Precision analyzer's measurement capabilities.

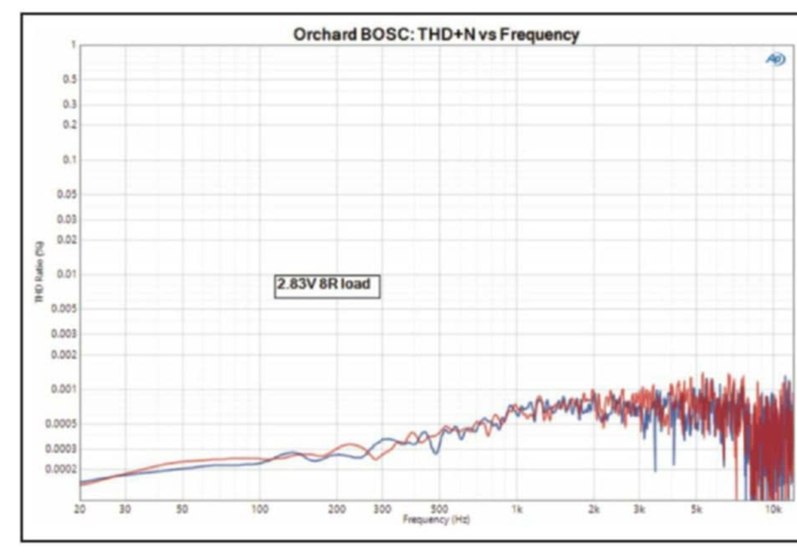


Figure 7: The BOSC's THD vs. frequency at 1 W shows absolutely negligible distortion through the audio band.

importance to hot climate area dwellers like me, but also suggests higher reliability.

The BOSCs on the Bench

After using these amps for a few weeks, I predicted that there would be nothing unusual or alarming in the measurements. Unfortunately, because of the very short length of time I was able to keep the BOSCs, my measurements aren't comprehensive, but I think they are sufficient to back up my listening impressions.

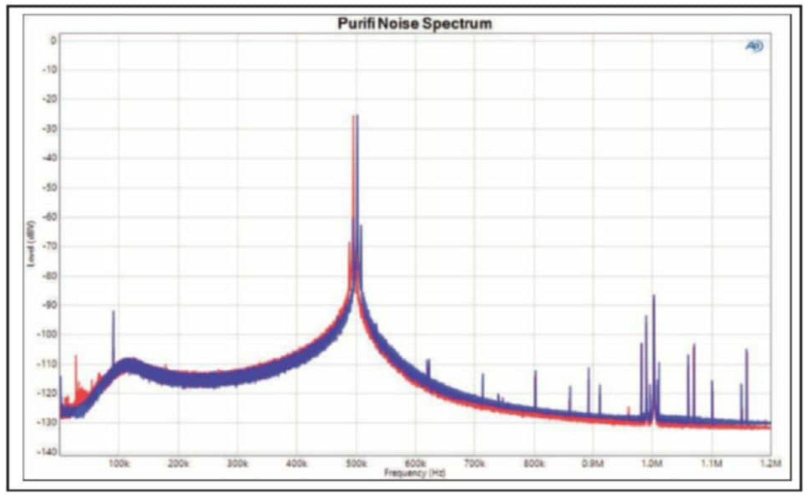


Figure 8: The no-signal ultrasonic noise of the Purifi amplifiers is exceptionally low and, like most Class-D amps, is dominated by its switching frequency.

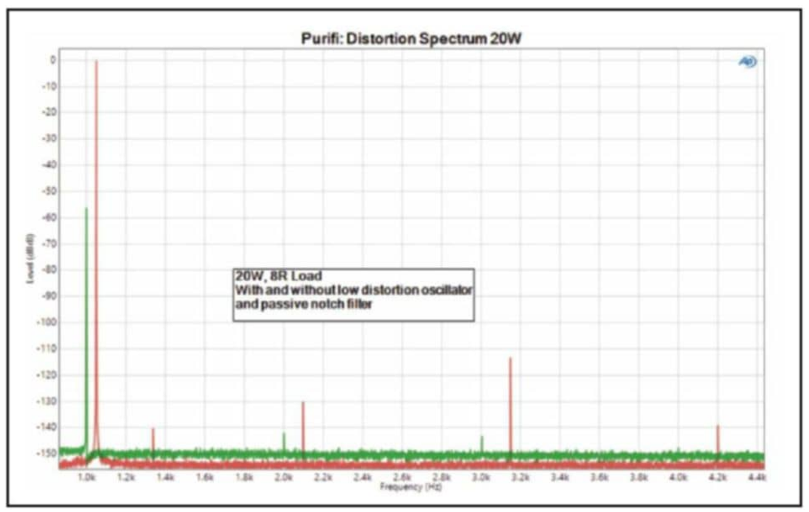


Figure 9: Using a dedicated low-distortion oscillator and filter, the distortion spectrum of the Purifi can be seen to be significantly lower than the analyzer residual.

About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as an R&D director for a construction products company in Mesa, AZ.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.

As usual, my test setup comprises an Audio Precision APx525, though I briefly pressed an APx555 B Series into service. Later on (unfortunately after the BOSCs had to be returned), I had access to a very low distortion 1 kHz sine wave source (Vicnic) and a tuned twin-T notch filter, which extended my 1 kHz distortion measurement capability. Using the "bare" APx-525 as a signal source and analyzer, my distortion measurement capability extends down to about -115 dB below the signal level, dominated by the third harmonic. Dummy loads were variously 8 Ω, 4 Ω, and a simulated loudspeaker load (this load simulates a closed-box system with a 50 Hz fundamental resonance and a crossover at about 3 kHz), all mounted on large heatsinks to minimize thermal modulation.

The gain of the two BOSC amps was 16.88 dB, almost exactly the rated value, with the difference in gain between the two amps less than 0.005 dB. This suggests excellent quality control and parts matching. Input impedance was likewise right at spec, with measurement indicating barely under 5 kΩ at 1 kHz.

I've seen much angst and dark implications about the ultrasonic noise output of Class-D power amps, so I routinely measure them for this characteristic. **Figure 3** shows the no-signal output spectrum of the BOSC amps with an 8 Ω load connected. The ultrasonic noise floor is low, never exceeding -95 dBV (18 µV) except near the switching frequency of about 720 kHz, where it rises to about -3 dBV (710 mV). These noise results seem absolutely inconsequential, and you can see the noise floor drop further as the frequency goes down to the audio range. Measuring the audio frequency noise for a 24 kHz bandwidth, the total noise was 46 µV for the worse channel, slightly higher than specification but still well below any reasonable worry. To put this in perspective, if you have a very efficient 100 dB/2.83 V/meter horn, this translates to an SPL of 4 dB at 1 meter, which might be barely audible to a toddler in an anechoic chamber. Maybe. As a practical matter, I couldn't tell if the amps were on or off when my ear was inches away from my speakers.

Figure 4 shows THD vs. power at midband (1 kHz) into a 4 Ω load. My dummy load made unhappy noises as I hit 250 W, and the distortion was still under 0.004%. The spectrum of the distortion at 200 W is shown in **Figure 5**, and has second harmonic as the highest level distortion product at -100 dB (0.001%). Likewise, with a two-tone intermodulation signal at 19+20 kHz at 35 W, the sidebands are all below -100 dB, and the actual intermodulation product (1 kHz) is below -115 dB. These are impressive measurements.

Besides full power measurements, amplifiers are often measured at low power, traditionally to bring out the effects of non-ideal behavior (e.g., crossover distortion), and in recognition that for most musical signals at most reasonable listening levels, the amplifier will not be putting out anything close to its rated power. Even though this isn't likely relevant to Class-D amps, I bow to tradition.

Figure 6 shows the distortion spectrum for a 1 kHz signal at 2.83 V into an 8 Ω load (1 W). It is dominated by the residual distortion of the AP analyzer, which is extremely impressive! Sweeping the frequency at the same fixed 2.83 V output level, the total harmonic distortion plus noise (THD+N) results are shown in **Figure 7**, and are likely dominated by the noise. In any event, any distortion is well below 0.001%, so it's safe to declare it inaudible.

Enter the Purifi EVAL1

Next on my test bench was the Purifi Audio EVAL1 system. Purifi's 1ET400A datasheet (see Resources) has a very comprehensive set of measurements, so rather than repeat all of them, I checked the main ones (which all checked out) and added a few other measurements.

First, the basics: gain of the system was 27.45 dB, slightly higher than specification. As with the BOSC, the gains were very tightly matched between channels, within 0.05 dB. Input impedance measured about 95 kΩ differentially. The no-signal noise spectrum is shown in **Figure 8**, and indicates a 500 kHz switching frequency at about 0.2 V, which is inconsequential. The ultrasonic noise floor is likewise quite low with the largest component being a spur at -93 dBV at 90 kHz. This is low enough not to disorient bats passing by.

I proceeded to perform some measurements at moderate and high power. The 100 W 19+20 kHz intermodulation distortion (IMD) spectrum is shown in Figure 8 for an 8 Ω load (in red), and the sidebands are below -115 dB, with the main IMD product of 1 kHz below -125 dB. Substituting the simulated speaker load (in blue), the distortion actually drops slightly, though at these minuscule levels, the differences are academic.

The distortion spectrum of a 1 kHz sine wave at a moderate level of 20 W into the 8 Ω load is shown in **Figure 9**; the light red curve is generated using the AP analyzer alone and is below the analyzer's residual. Using the low distortion Vicnic oscillator and a twin-T notch filter with 55 dB of fundamental suppression, it can be seen (green curve) that the amplifier's distortion is below -140 dB (0.00001%), which is... astounding, and proof that the limitation here is indeed the AP analyzer, not the amplifier.

The source impedance was below my ability to measure, but looking at the frequency response differences when substituting the simulated speaker for the resistor load, the frequency response change was less than 0.01 dB. This amp's sound will not be affected by the loudspeaker's impedance. THD vs. frequency at 2.83 V is shown in **Figure 10** for the 8 Ω and simulated speaker loads—there's no significant difference. Repeating the measurement at 30 V (112 W into 8 Ω), I obtained the results shown in **Figure 11**, where we can finally see a small rise in distortion in the top octave, but still below 0.02% and relatively unaffected by load.

Figure 12 shows a 2.83 V sine wave at 1 kHz, along with the residual magnified by 1000. It's

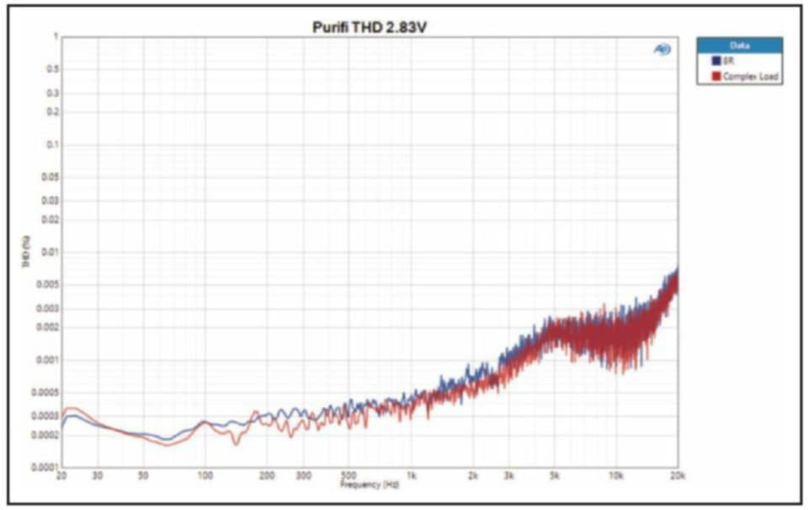


Figure 10: At 2.83V output, the Purifi's distortion vs. frequency is low and independent of load.

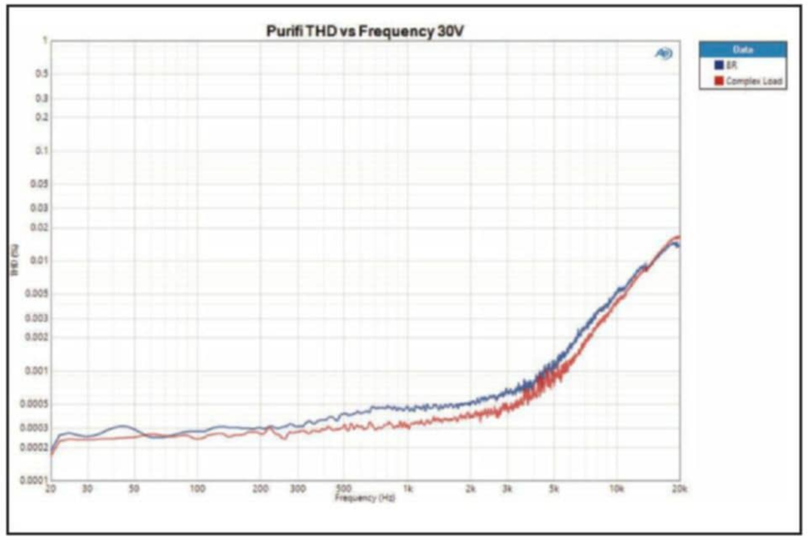


Figure 11: At 30 V output (112 W into 8 Ω), the Purifi's distortion remains extremely low, even at high frequencies and with complex loads.

clearly dominated by noise, with no visible glitches or harmonics. These are essentially perfect measurements.

Wrap-Up

Earlier in the article I used the term "appliance," and really it's the best way I can think of amplification here in the year 2020. Both of these amplifiers ran cool, mechanically quiet, noise-free, and basically... unexciting. And that's the highest compliment I can give to electronics that are intended to be transparent.

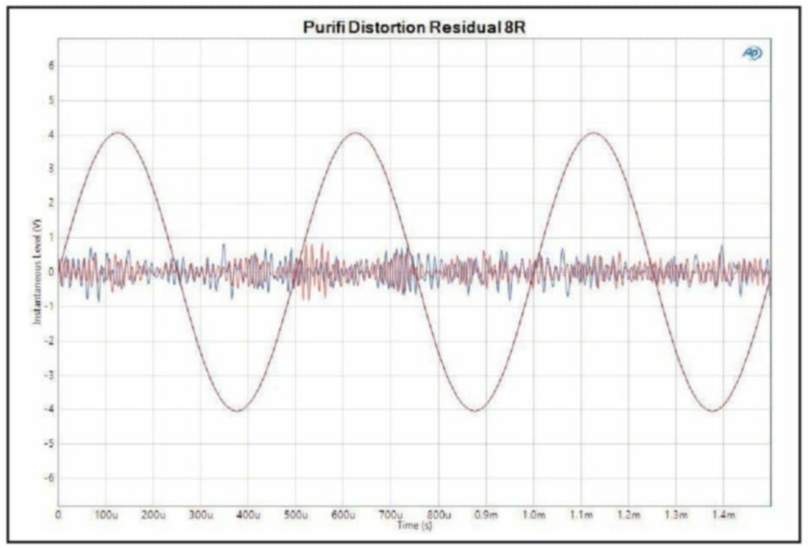


Figure 12: The Purifi's distortion residual at low power (here magnified by 1000 times) is dominated by noise and shows no signs of typical amplifier artifacts.

For integrators or those who want to experiment with front-end design, the Purifi is the clear choice. For someone who just wants an amplifier to run the stereo, one of the many integrated versions of the Purifi or (if going for separates) the Orchard BOSC will do pretty much equally well—and by "well" here, I mean "superbly." Special system requirements (e.g., common grounding between channels) or just a non-sonic esthetic choice (e.g., look, feel, reputation) might sway a prospective purchaser in one direction or another, but soundwise, either is essentially perfect. Any measurement differences seen here fall orders of magnitude below human hearing thresholds.

This does raise a philosophical question in my mind—where do we go from here? Although audibly transparent amplification capable of driving most loudspeaker loads has been available for decades, we now can do it at a relatively low price, high power, compact, cool, with vanishingly low distortion and source impedance, and with efficiencies in the 90+% range, well past diminishing returns. Could it be time to declare victory and go home? Or just admit that the only thing left to do is to scale upward and downward in power? Is this analogous to Francis Fukuyama's "End of History"? Personally, I'm certainly leaning toward making amps such as the Purifi or BOSC the last ones I'll ever own.

Author's Acknowledgements: I'd like to thank the kind people at Orchard Audio and Purifi for providing product photography and technical support, and Jan Didden for laboriously drawing the block diagrams.

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A Visit to Purifi Audio

Where the Magic Happens

The Purifi Audio EVAL1 stereo demo amplifier is shown side by side with the first 1ET400A production modules.

The Danish city of Roskilde is probably best known for its annual rock festival. During that time, the number of visitors exceeds the population by more than a factor of two. Less known is that Roskilde, half an hour away from the Danish capital of Copenhagen, is also home to the audio development house Purifi Audio. *audioXpress* had the opportunity to visit Purifi's headquarters, where the magic happens.

By
Ward Maas
(The Netherlands)

The world learned about Purifi Audio during the 2019 Munich High End show, where the company introduced a new Class-D amplifier and a new 6.5" driver. While the products themselves were mainly the first visible results of an intense research and development effort, it was the names involved with the company and its development team that made it clear that there was something special happening. We trust that most *audioXpress* readers will be familiar by now with the names behind Purifi Audio and the team's extensive work, so we will not repeat it here. (We recommend reading any of the articles mentioned in Resources).

Faced with the opportunity to visit the lab in Roskilde—and knowing *audioXpress* would finally be able to publish an actual review of the 1ET400A Class D amplifier—we seized the chance! The Purifi building is located in a small industrial estate just outside the center of Roskilde. This is a purely functional building, optimized for the company's research efforts that lead to the creation and assembly of working prototypes and measurement facilities. No superfluous things that require time or attention. The available space is completely filled with electronics, loudspeakers, loudspeaker parts, computers, and measuring equipment. The need to understand the designs is felt by everyone. During our visit, we were able to meet Lars Risbo, together with Kim Madsen, Applications Manager, who joined the company from Texas Instruments, and Claus Neesgaard, co-owner and director.

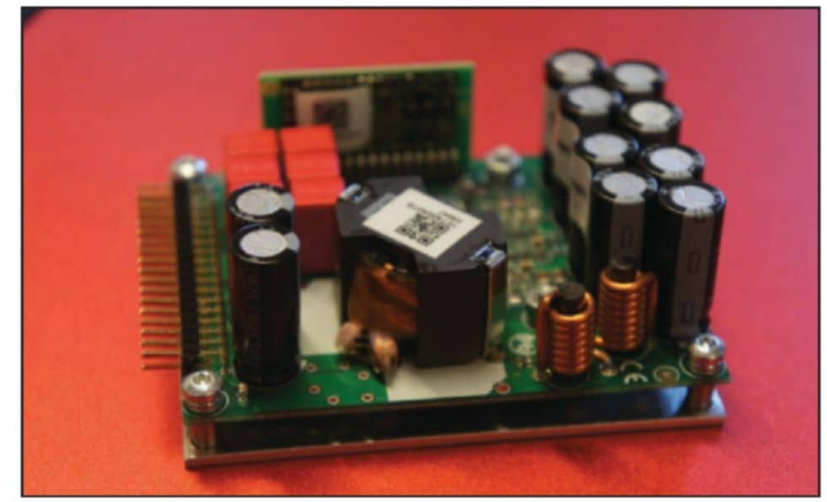
The early Purifi Audio demonstrations confirmed the expectations about the disruptive potential of the company's first projects. We would like to write that the rest was history. But of course it's not. It takes time and effort before such products get to market—and get the recognition they deserve—and since Purifi is pursuing mainly licensing and OEM opportunities, the company's challenge is mainly to keep the buzz going until actual products reach actual consumers. Until then, the Purifi team is using every opportunity to explain their work and research, while promoting the fundamental approach and demonstration platforms.

The Class-D Amplifier

And their work, for now, has been mainly to promote the potential of the company's new "self-oscillating" Class-D amplifier platform. The platform has been the main focus of years of research by Bruno Putzeys, whose work began at Philips, with the early development of the new topology, later followed by his work at Hypex, where he had a major role in the development of the famous Ncore series—a contribution that firmly placed Class-D definitively in the "High End" category. In collaboration with Lars Risbo—who pioneered direct switching PCM-PWM audio amplifiers as the founder of Toccata Technology, later acquired by Texas Instruments—they both became convinced that it could be even be better. This led to the development of the Purifi 1ET400A Class-D module. The specifications of this module were such that it caused a lot of amazement. It is not every day that you see an amplifier leaving people speechless in listening tests.

A similar story applies to the introduction of the PTT6.5W04-01A Long Stroke driver, Purifi's first attempt to demonstrate its new transducer ideas. Theoretically, air displacement can be achieved by making a speaker with a large surface area move little or a speaker with a small surface area make large movements. The latter did not prove practical until now. Until recently, bass speakers had to be dutifully large. The Purifi driver demonstrates that you can indeed generate lower and midrange frequencies with a small speaker, with low distortion over the full excursion.

Purifi Audio's headquarters. The building is well-known in Denmark, being the place where PointSource Acoustics served the R&D needs of the speaker industry since 2009. PointSource was founded by Carsten Tinggaard, who is now also one of Purifi's co-owners.



Purifi's 1ET400A amplifier module

Purifi's Underlying Values

This also illustrates one of the company's foundations. The Purifi team tirelessly tries to understand the underlying mechanisms and build up knowledge before developing products. In this case, the knowledge accumulated through the research even lead to recognizing the need to develop their own measurement methods. Another aspect that raises curiosity is the fact that Purifi is very open about the results achieved—being scientific and goal driven, the datasheets are full of measurements that are seldom seen in the industry but reflect the depth of their research.

The Amplifiers

The Purifi 1ET400A Class-D module came about after audio enthusiasts thought they heard some granularity ("a recognizable grainy texture in the sound," in Putzeys' words) in the existing Class-D modules. Apparently, there was something that escaped everyone's attention. After a long search

The PTT6.5W04 6.5" long-stroke drive with its unique Neutral Surround profile

Inside Purifi Audio's acoustics lab and loudspeaker facilities

it was finally found that the hysteresis curve of the used filter coils at micro level behaved slightly different than generally assumed. Jan Didden wrote about this for *audioXpress* in his report of an Audio Engineering Society (AES) meeting in The Hague, the Netherlands (see Resources). Meanwhile, Bruno Putzeys made a detailed report of his investigation on hysteresis distortion available on the company's website. In itself, the method to strongly reduce these nonlinearities was not so difficult. Only the required high open-loop gain was something at which Class-D amplifiers do not excel. By further developing the understanding of Class-D amplification, Purifi managed to significantly increase open-loop gain, and thus, reduce the nonlinearities to extremely low values. The result is, as indicated earlier, an amplifier module of which measurement values are almost ideal. The noise floor of the 1ET400A is a serious task for all but the best audio analyzers.

Some members of the Purifi team have a strong background in the semiconductor industry—mostly at Texas Instruments. This is also reflected in the specifications that are guaranteed. Every amplifier module has been tested in every possible and impossible way to ensure that, used within the specified limits, it will always meet the specifications.

The amplifier modules from the first production batch that had just arrived during our visit, were all tested individually. Some were submitted to the so-called "horror test." This is not only meant to check if a product meets the specifications, but especially to see how a product behaves under several error conditions at the same time. It is interesting to realize that temperature effects behave differently under multiple error conditions. Indestructible may be a bit too much to ask, but it is clear that the Purifi team has a deep understanding of production quality and its implications in result of their semiconductor industry background.

The 1ET400A is currently limited for supply voltages up to ±65 V and an output current up to 25 A. Of course, this concept is expandable. However, this doesn't mean a simple increase of the supply voltage and use of heavier power field-effect transistors (FETs). The concept of how this amplifier works means that for each new amplifier the extended parameter set has to be recalculated. With this, Purifi gains significant competitive advantages, but it still needs to overcome production hurdles.

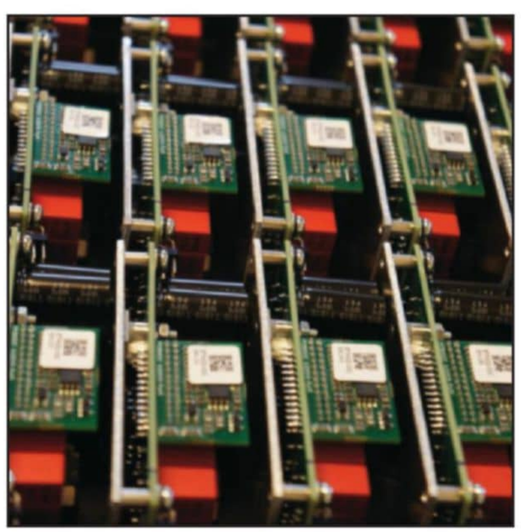
The development program also includes a "high voltage" version with an output current limit of 40 A. With the introduction of the "bridged" version and the associated switched power supplies, the aim is to cover the entire spectrum from 400 W to 1 kW to 2 kW.

For now the technology will reach the market in the high-end audio segment. The first company that agreed to use the Purifi amplifier technology was NAD Electronics, followed by Peter Lyngdorf's own company, Lyngdorf Audio. NAD promoted demonstrations of its Masters M33 BluOS Streaming DAC Amplifier at CES 2020, in January.

Judging by the demos promoted, it seems that the Purifi Class-D based amplifiers will be able to compete with all "reference" amplifiers (for an actual evaluation, see Stuart Yaniger's review in this edition). No doubt, Purifi is bound to have a special impact in the industry.

The Speakers

As already indicated, Purifi wants to disrupt the market with two products in completely different categories. Again, the knowledge drive was a basis for product development. If you look at the cone surface of a loudspeaker when it moves forward



The first batch of 1ET400A amplifier modules for individual testing

Inside the facilities, we had a chance to look at Purifi's own internal demo room, including some of the company's original demo speakers.

One of the many Purifi workstations, with Kim Madsen shown here performing audio measurements.

About the Author

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

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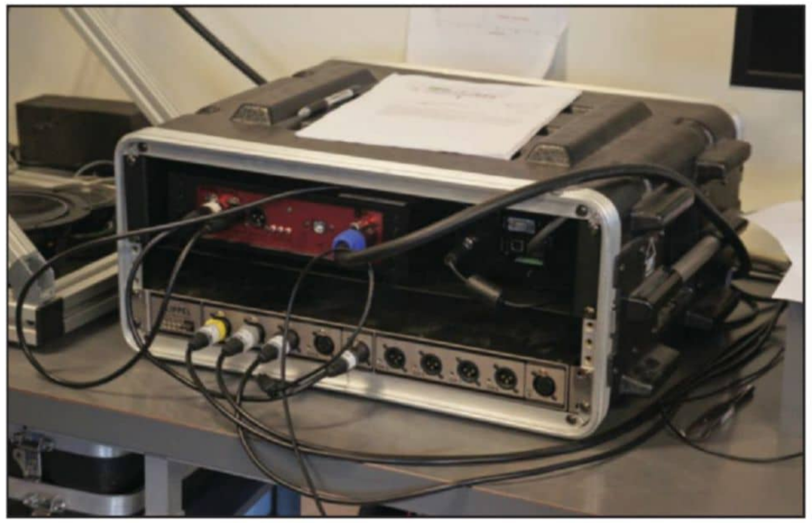
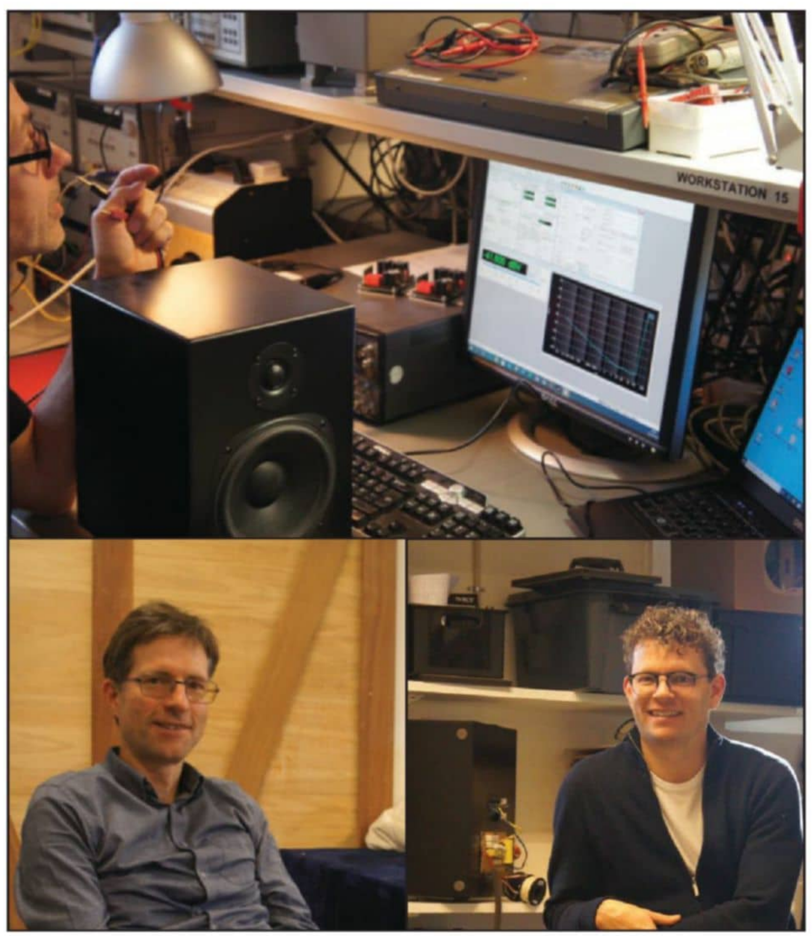


The famous Purifi loudspeaker measurement kit is shown along with a Klippel measurement system. When the Purifi team discovered that they could not get the right measurement information they needed, they developed their own in-house solution.



Lars Risbo, Kim Madsen, and Claus Neesgaard, pictured above, were our hosts during audioXpress' visit to Purifi Audio.

or backward, you can see that there is a surface difference and therefore a source of distortion. This, of course, applies to more position-dependent measurement values that cause distortion. In all these years of loudspeaker research, it is still reasonably accepted that loudspeakers have higher harmonic and intermodulation distortion.

Extensive research efforts seem to show that when you improve one aspect of speaker behavior, another gets worse. The Purifi development team identified and focused on two significant distortion mechanisms that have gone mostly unnoticed: Force Factor Modulation (FFM) and Surround Radiation Distortion (SRD). Together, they explain practically all the low-frequency distortion that currently exists in otherwise well-designed drive units. By identifying these mechanisms, Purifi has been able to link geometrical improvements to audible improvements.

Again, the resulting measurement data, especially the distortion figures, are very good. During the demonstrations, it was very audible that the PTT6.5W04-01A Long Stroke driver will be a very useful resource for loudspeaker designers. As Purifi indicates, this is a long-stroke speaker that is at least a match for, or just better than, many short-stroke speakers with a large cone surface area. And again, because they understand the mechanism, Purifi was able to create a first product that can serve as a basis for other models. Generating a deep bass with a small speaker is particularly interesting.

The company is already planning to develop 3" and 4" models, targeting desktop monitors and soundbars. Perhaps there will be another 5.25" driver, but then there are also plans for 8", 10", and even 12" models —all with the potential to disrupt the speaker world. There is also a new passive radiator version, the PTT6.5PR-01A, which is equipped with the same Neutral Surround technology and intended as a companion for the PTT6.5W04.

Given the team's extensive experience in previous companies and research fields, it is not surprising that the attention to production quality has so much focus. During our visit, there were many products available to verify that attention to detail. In the drivers, a process such as gluing receives the highest attention. Which glue, how much glue, how to apply, but most of all, how to make sure that every bonding process produces the same result. And the team continues to experiment with alternative materials and designs, such as cone materials, even though Purifi prefers to give customers the opportunity to fill in their added value/market distinction.

Overall Impressions

It rarely happens that such a young company releases two products at the same time, which have the potential to change the audio industry. Having that potential, there is quite a lot to do in a short period of time to make it a success. Even to tell the world that they exist and prove their claims, Purifi had to focus on making demo models that can be "seeded" in the right place. The 1ET400A Class-D amplifier module has a connector where all connections can be found. For its own use, Purifi has developed a PCB as an interface between the existing sound sources.

Using two amplifier modules connected to that front end system, the company built a number of red demo housings, named EVAL1, available for the press to review and also for evaluation by OEM manufacturers. All existing Purifi demonstration products can be ordered through an international webshop that has been launched on the company's website. The webshop even allows DIYers to order separate modules and drivers, as long as they are available.

The Purifi website (www.purifi-audio.com) is also a treasure trove of information, including the valuable Blog and Tech sections, with articles that provide insight about what the company is doing, the problems they encounter, and how they have solved them. For those who cannot wait to hear for themselves what the small group of "insiders" are so enthusiastic about, there is also a complete construction description of a demo speaker, complete with crossover.

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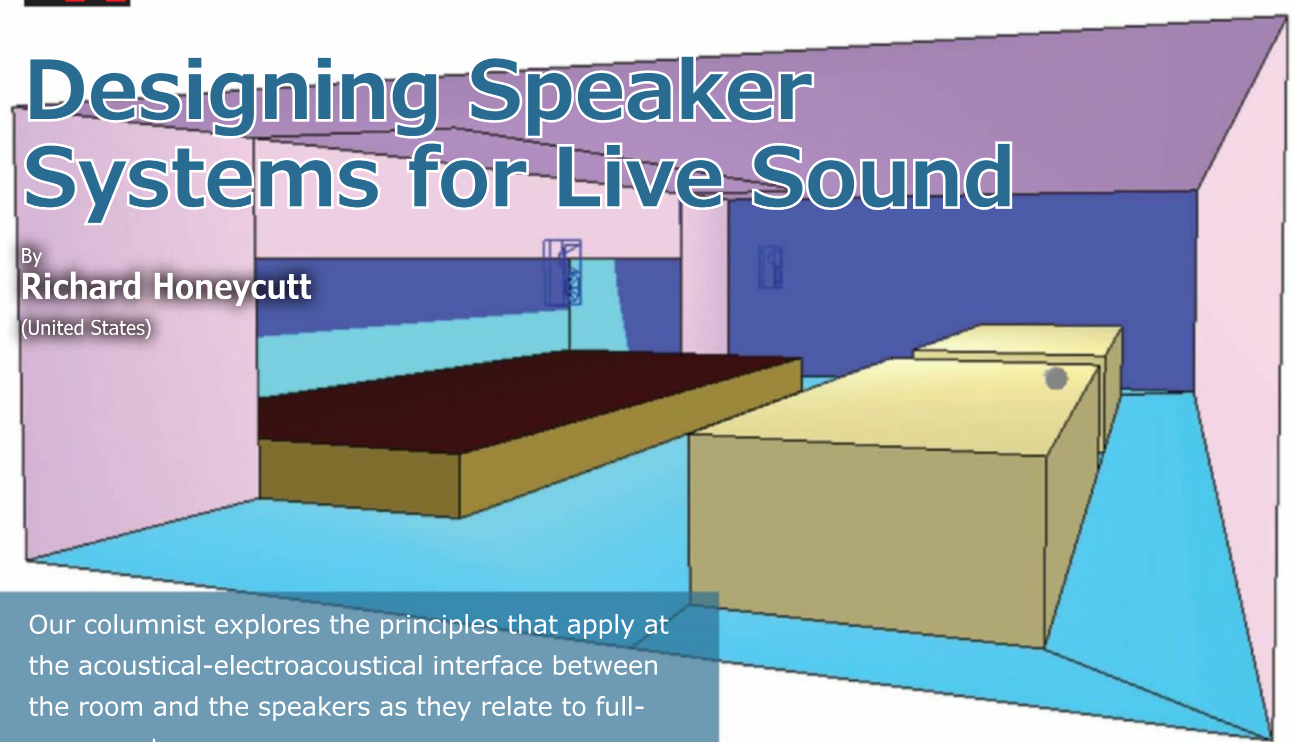

Designing Speaker Systems for Live Sound

By
Richard Honeycutt
(United States)

Our columnist explores the principles that apply at the acoustical-electroacoustical interface between the room and the speakers as they relate to full-range systems.

Photo 1: This computer model shows a small club.

Whether you're setting up a portable sound system or designing a system for fixed installation, certain physical principles apply at the acoustical-electroacoustical interface between the room and the speaker(s). In fact, to a certain extent, public opinion of some speaker designs has been significantly influenced by how forgiving the system is when the proper physical principles are ignored in its installation! We'll look at these principles as they relate to full-range systems—subwoofers bring in other issues that we will not cover in this article.

The primary goals of speaker design, selection, and installation for live sound include:

1. Put the sound where the people are.
2. Provide the proper sound level.
3. Cover the audience evenly in all frequency ranges of interest.
4. Avoid hitting reflective surfaces to the extent possible, in order to avoid annoying echoes.
5. Make sure the acoustical image matches the visual image: a speaker at a podium with his mouth 5′ above the stage floor should not sound like the "voice of God," coming from high overhead; house-left listeners and house-right listeners should hear central sources from the center.

Speaker Positions

The simplest room shape is rectangular, and the "talent" [person(s) talking, singing, or playing instruments] is typically located on one of the short walls, one of the long walls, or centered in the room. **Photo 1** illustrates such a room. The stage takes up most of the left-hand wall, and two large rectangular blocks represent seating in the back of the club. The floor is carpeted; the stage floor is wood on joists; and the walls are gypsum wall board. Two speakers are shown in typical "speakers on sticks" positions. These speakers are specified as having a 90° × 50° (H×V) coverage pattern. **Figure 1** shows the actual octave-band directivity balloons; note the wide variation in coverage with frequency! (The balloons in some adjacent octaves—63 Hz and 125 Hz, 1 kHz and 2 kHz, and the 8 kHz and 16 kHz—are quite similar, so some balloons have been omitted.)

Photo 2 shows the 1-kHz sound pressure level (SPL) map produced by the modeling software. Coverage would be a bit more limited at higher frequencies, and less so at lower frequencies, because of the speakers' variation in coverage angle with frequency.

Reverberation Time

The Reverberation Time (RT) of this small room is quite low—under about 0.6 seconds— but it does vary according to the sound coverage of the speaker: where the direct sound is strongest, RT is lowest. **Photo 3** shows the lower RT in the center of the audience areas—the parts best covered by the speakers. The speakers affect the RT in that the more energy with which they hit the hard walls and ceiling (and the less energy that hits the absorptive audience), the longer will be the RT.

Depending upon the program material, speech intelligibility may be important in a given venue. In a small club, for example, a folk-music act or a poetry reading would benefit from good speech intelligibility. Speech intelligibility is degraded by reverberation, among other things. One metric used to quantify speech intelligibility in architectural acoustics is definition (D-50), defined as the total sound energy within the first 50 ms of the onset of the sonic event, divided by the total sound energy produced by the event. **Photo 4** shows the definition produced by the speakers in the small club. As you can see, even the relatively small difference in SPL at the ends of the seating areas, compared to the centers, results in a significantly lower definition at the ends. Actually, because of the small size of this room and its correspondingly low RT, the definition is pretty good even in the "low-def" places.

If permanently installed speakers were used in this club,

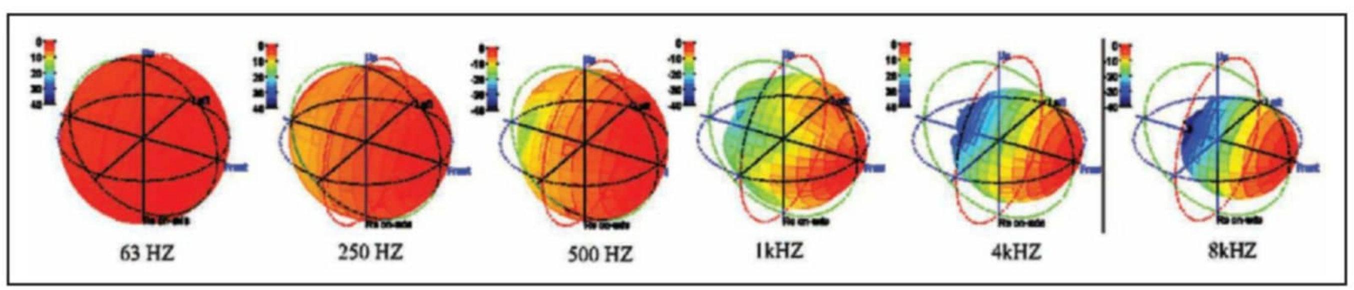

Figure 1: The directivity balloon of the speaker changes significantly with frequency.

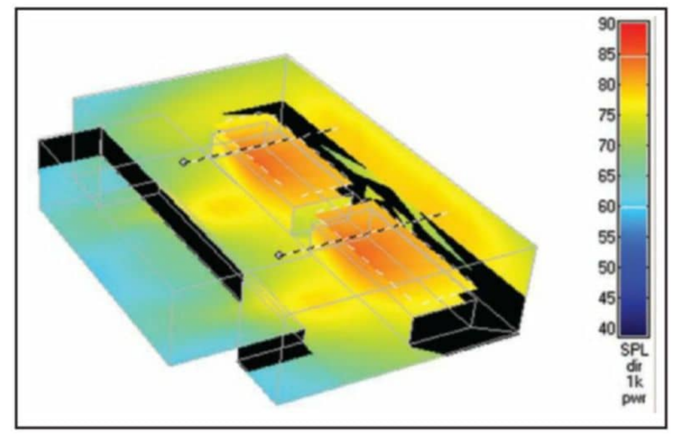

Photo 2: The stand-mounted speakers make the SPL in the middle hotter than at the sides, by about 5 dB.

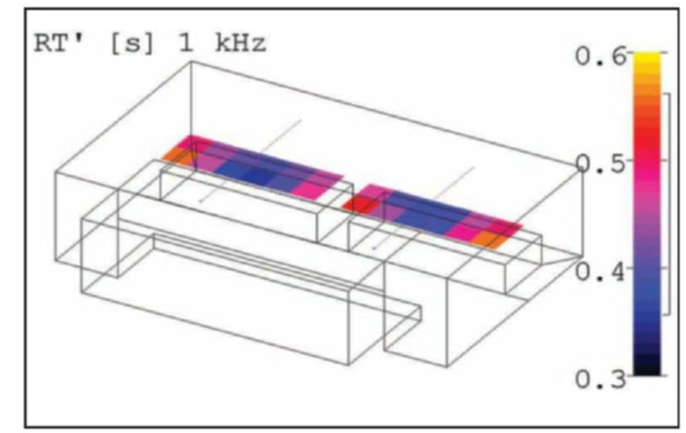

Photo 3: The reverberation time (RT) of the room is lower in the areas best covered by direct sound from the speakers.

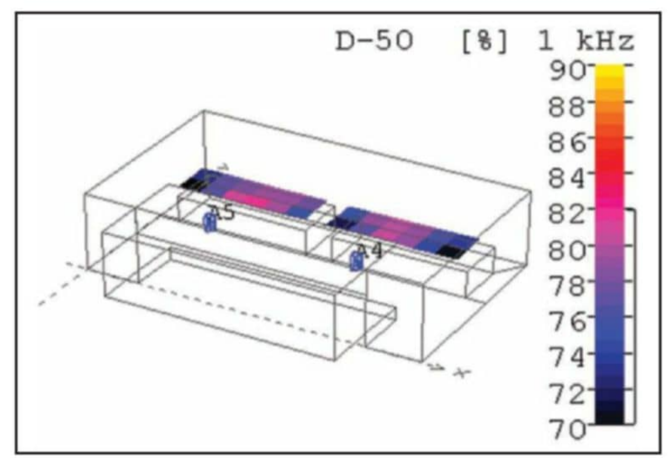

Photo 4: The listeners at the ends of the seating areas will experience lower-definition sound.

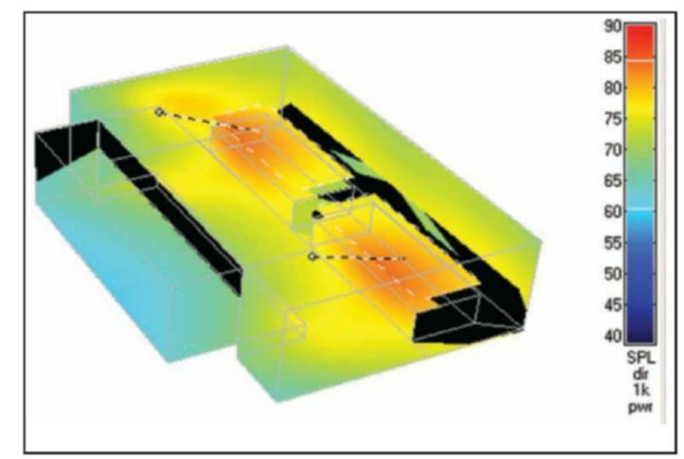

Photo 5: Using installed speakers rather than stand-mounted portable ones makes coverage slightly more even.

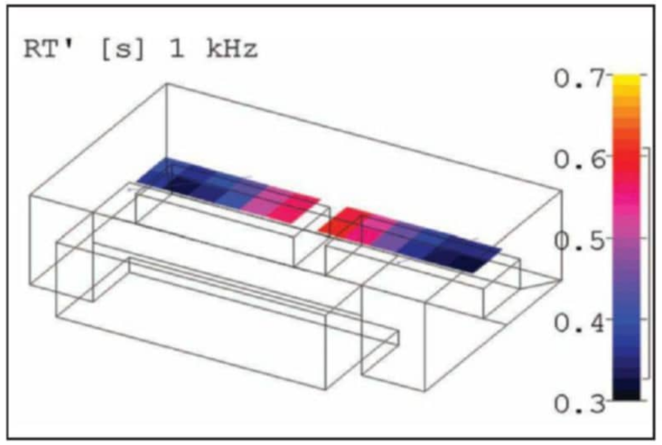

Photo 6: The overall RT is slightly lower with installed speakers.

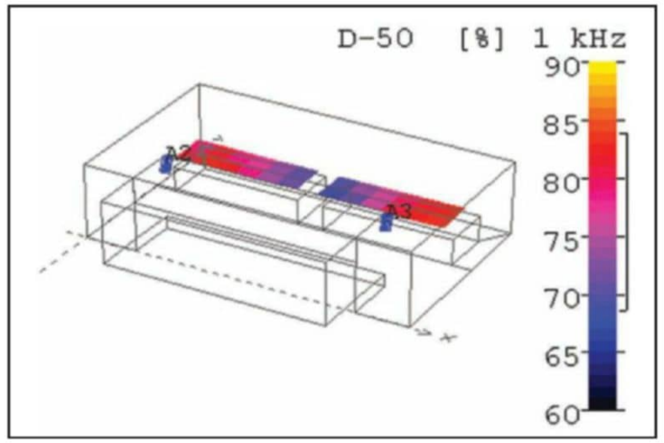

Photo 7: The definition is slightly improved by the slightly lower RT, as you can see by the larger red areas here than are shown in Photo 4.

they would probably be located higher than the stand-mounted ones were, and would be aimed downward toward the audience. **Photo 5** shows the SPL produced by these speakers in the "installed" position. Comparing Photo 2 and Photo 5, note that the downward angling of the speakers makes less of the sound hit the back wall than with the speakers on stands and pointing horizontally.

The difference in mounting position produces a lower RT over more of the seating area, as you can see from **Photo 6**. This also results in a larger percentage of the seats being in areas with improved definition, as shown by **Photo 7**.

Omnidirectional Speakers

At least one manufacturer of omnidirectional speakers advertises its product as being the optimum solution for all venues. It is instructive to compare an omni speaker to the two cone+horn speakers we have been examining. **Photo 8** shows that the omni does indeed give almost as even coverage in this room as the pair of cone+horn speakers. It also covers the room boundaries with sound; we will discuss this effect further.

As mentioned earlier, hitting the reflective room walls and ceiling with more sound increases RT. You can see this effect in **Photo 9**, which shows that the 1-kHz RT approaches 0.8 seconds, as compared to 0.6 seconds with the cone+horn pair. This is a noticeable increase, and would be even more concerning if the room were less fully occupied than the full occupancy the modeling assumed. The detrimental effect on definition can be seen in **Photo 10**: average D-50 is about 61%, as compared to 81% with the cone+horn speakers.

It should be mentioned that as far as sound coverage is concerned, an omni speaker is a viable option in this room. In a very dead room, the increased RT and correspondingly decreased definition might not be an issue. Positioning of an omni is less critical than it is for directional speakers, and of course aiming is not an issue at all! In some ways, an omni can be a more forgiving design.

Single-Speaker Options

One other option exists for using a cone+horn speaker in this small club. Although we haven't mentioned it, both the stand-mount and the "installed" options are subject to comb filtering in the center of the audience area. Naturally, since the omni is a single source, it is immune to comb filtering. But so is the use of a single cone+horn speaker. If a speaker having a very wide horizontal coverage angle were used, it could be mounted near the ceiling in front of stage center. Or it could be mounted in a front corner and angled to cover the audience as well as possible. **Photo 11** shows the sound coverage that would result from using this setup. The sound level is consistent within about 3 dB throughout the audience

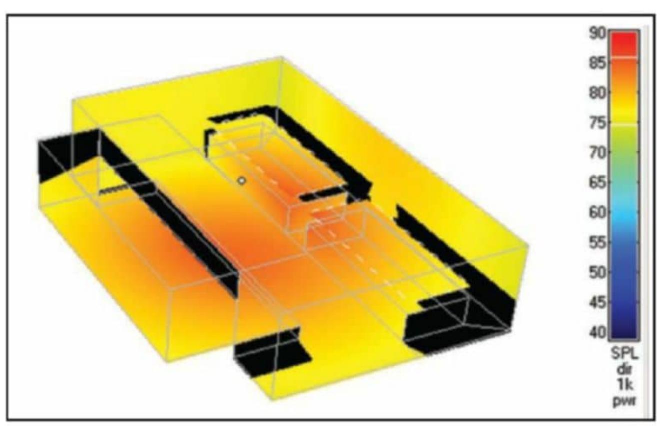

Photo 8: The omni speaker covers the audience—as well as the walls, the ceiling, and the floor—fairly evenly.

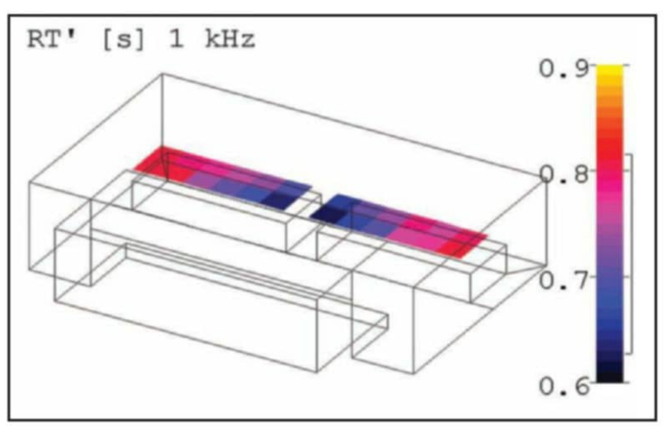

Photo 9: If you note the difference in the color legend, you will see that the omni makes the RT higher.

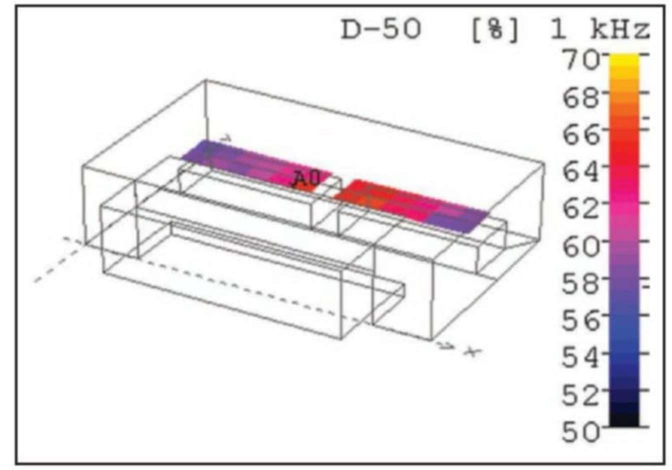

Photo 10: The omni speaker produces a lower-definition sound field in this room.

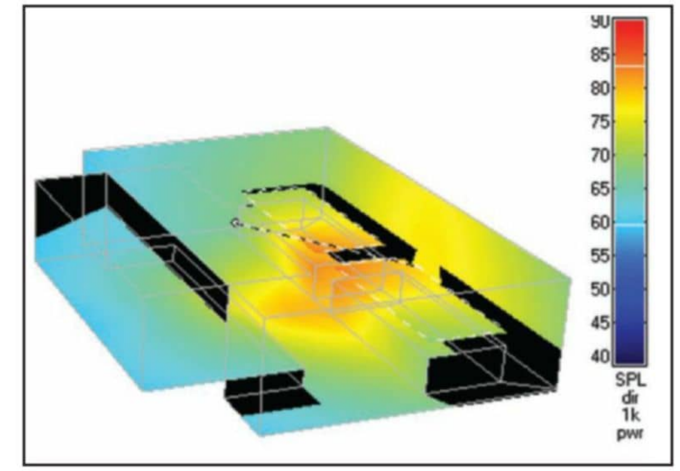

Photo 11: A single cone+horn speaker having a wide horizontal coverage could be used in this small club.

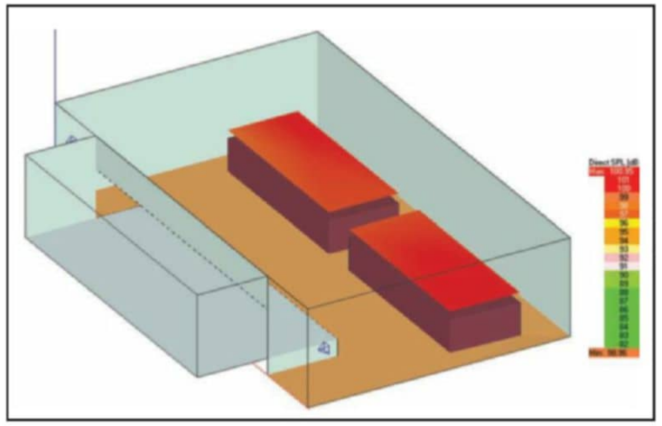

Photo 12: A pair of small line arrays can give very even coverage in this room.

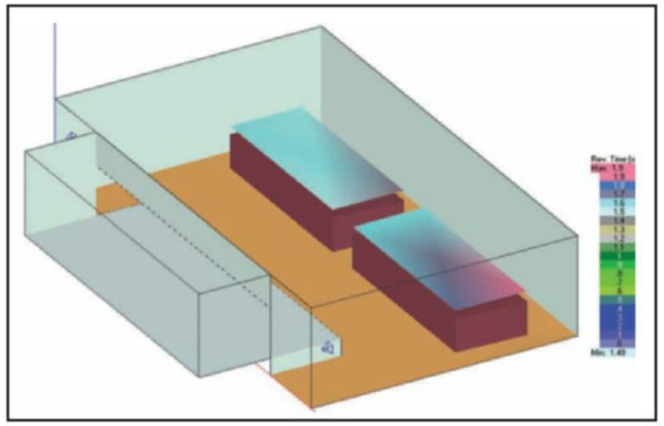

Photo 13: Using line arrays results in a longer RT in this room.

area. The RT and definition are very slightly higher than for the pair of installed speakers.

Line Arrays

Two line arrays are a common solution for a room that is wider than it is deep. **Photo 12** shows how this arrangement would perform in the small club. (This illustration was produced using different modeling software, which accounts for the different appearance.) Most line arrays have so wide a horizontal coverage that a lot of sound hits the side walls, resulting in a longer RT, as you can see in **Photo 13**. One way this manifests perceptually is that the definition is lower (see **Photo 14**).

Meeting the Goals

So far, we've looked mainly at Goal 1 (put the sound where the people are), which is related to Goal 3 (cover the audience evenly in all frequency ranges of interest). However, Goal 3 is strongly influenced by the speaker performance and system EQ. We also looked briefly at Goal 4 (avoid hitting reflective surfaces to the extent possible, in order to avoid annoying echoes).

Goal 2 very often depends upon the available gain before feedback. This is ultimately determined by the ratio of sound from the talent vs. sound from the speakers that enters the microphones. This, in turn, is affected by distance from the talker/singer's mouth to the mic, the distance from the nearest speaker to the mic, the directivity of the speaker(s), and whether the front of the mic is facing the speaker.

If you compare the sound-level plots, you'll see that the omni provides the highest sound level at the mic position, implying the lowest gain before feedback. The central cone+horn speaker is second worst, although its performance could be improved by mounting the speaker higher if the ceiling height permitted.

Goal 5 (proper acoustical image location) is difficult to meet for all audience members if a pair of speakers is used. Listeners in the center of the audience area will hear a centered acoustical image, but those on the right or left of center will hear the image displaced toward the nearest speaker. How distracting the mismatch between the visual image and the acoustical image turns out to be varies from one listener to another.

In this article, we have limited our discussion to a small club. In such a venue, issues such as slap echo, flutter echo, and reverberation are seldom problems, although in larger venues, these gremlins may exert a strong effect upon our choice of speaker design and placement.

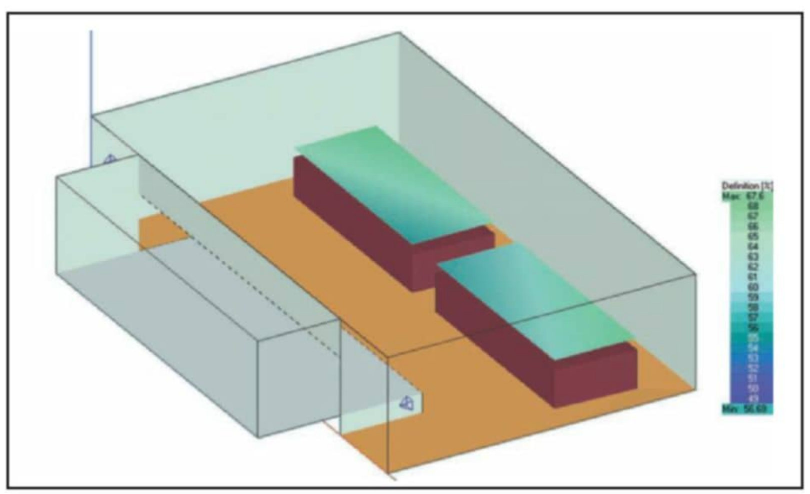

Photo 14: The longer RT reduces definition.

Low Frequency and Popcorn Noise in High-Speed Amplifiers

In this article, Michael Steffes explains how to convert a spectral 1/f noise profile to the industry standard 0.1 Hz to 10 Hz V_{pp} noise along with a typical measurement preamplifier, used to detect low-frequency or the so-called popcorn (burst) noise.

By
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(United States)

Low-frequency noise is one of the more elusive issues in applying high speed op-amps and fully differential amplifiers (FDAs). Converting a spectral 1/f noise profile to the industry standard 0.1 Hz to 10 Hz V_{PP} noise will be described in this article, along with a typical measurement preamp. This same preamp is used to test for popcorn noise with example measurements shown. All of these measurements are input referred to as either voltage or input current noise effects. They then get to the output through the desired gain and resistor values used.

Most input noise modeling operates as a swept frequency spot noise extraction using low noise preamps and spectrum analyzers. These generate the typical curves found in devices such as the OPA827 (a replacement for the industry standard OPA627) as shown in **Figure 1** (see Reference 1). Here, these front-page plots for this junction field-effect transistor (JFET) input device shows no current noise and 0.1 Hz to 10 Hz bandlimited time domain noise over a 10-second sweep.

JFET and Bipolar Amplifiers

All JFET and Bipolar input-type amplifiers show a 1/f corner effect in their input noise. Most complementary metal–oxide–semiconductor (CMOS) input amplifiers do as well with the exception of CMOS chopper amplifiers that have a flat noise response going down in frequency (and no popcorn noise).

One approach to mapping the noise spectral density to expected low-frequency V_{PP} noise is to compute the equivalent flat spot noise value that will integrate to the same noise power over the expected frequency span. This is completely general, but will be applied here to the 0.1 Hz to 10 Hz span commonly shown in many amplifier datasheets. Equation 1 shows that simple computation where E_N is the flatband noise in nV/√Hz, F_1 is the minimum frequency, F_2 is the maximum frequency, and F_{3dB} is the frequency at which the spot noise has increased √2 from the flatband value - the 1/f frequency (derivation on page 7, Reference 2).

$$E_{EQ} = E_N\sqrt{1+\frac{F_{3dB}}{F_2-F_1}\ln\frac{F_2}{F_1}} \text{ (equivalent white noise voltage)} \qquad \text{EQ 1}$$

Testing the OPA827 TINA model for its input voltage noise model shows a flatband noise of 3.8nV/√Hz and a 1/f noise corner at 24 Hz. Putting those numbers into Equation 1 using F_2 = 10 Hz and F_1 = 0.1 Hz yields an equivalent white noise of 13.3 nV/√Hz. To convert that flat spot noise to RMS noise, multiply by the square root of the noise power bandwidth, or √9.9 Hz. This yields an expected RMS noise in that span of 42 nV_{RMS}. **Figure 2** shows this simulated noise integration for the OPA827 set up as a gain of 1 with a 0 Ω feedback resistor—this should just be the modeled input voltage noise integrated from 0.1 Hz to 10 Hz. The 10 Hz value shows a reasonably close match at 40.9 nV_{RMS} validating Equation 1. This curve is continuing up as there is no bandlimiting in this simulation—yet.

To convert this RMS noise to expected V_{PP}, some estimate of crest factor is required. Typical values for noise are 6× to 6.6×. These relatively high crest factors are aimed at predicting a maximum V_{PP} that the actual device will rarely exceed. For any particular device, the 10-second test in Figure 1 will show a different V_{PP} on each sweep—such is the nature of noise. This crest factor choice is somewhat arbitrary as its imputed accuracy is swamped by the larger part to part variation for noise in the 1/f region.

The model delivers a typical profile, but physical devices vary widely on their low-frequency noise profile over frequency. Whatever is calculated here should be taken as very approximate. Using a 6× crest factor would predict 6 × 42 nV_{RMS} = 252 nV_{PP} closely matching the typical datasheet value of 250 nV_{PP} since that specification was also developed using this 6× number. Remember, this input referred so multiply by the noise gain in your application to get an output noise.

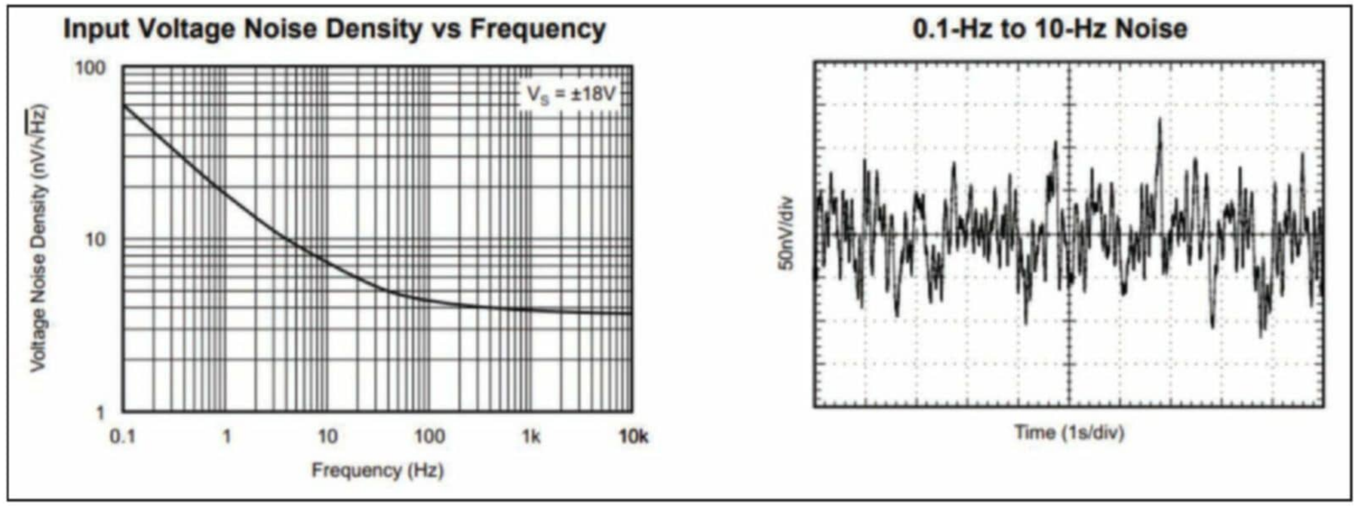


Figure 1: Here is the input spot noise over frequency and low-frequency time domain noise for the OPA827.

Measuring Low-Frequency Noise

By far the most common noise measurements use a spectrum analyzer. In the early days of op-amp development, a 10-second sweep of a 0.1 Hz to 10 Hz bandlimited V_{PP} noise became standard as an easy comparison metric. For that, a high gain, bandpass filtered, noise preamp is required. The early Burr-Brown low-frequency noise preamp used popcorn-screened OPA627s. Later work (see Reference 3) went to the more recent OPA827 device. Both of these devices have their own 1/f noise characteristics and possibly popcorn noise. One of the more recent gain of 100 V/V three-stage active filter designs has been adapted to an even more recent OPA388 (see Reference 4) CMOS chopper amplifier shown in **Figure 3**. CMOS chopper amplifiers have an intrinsic flat noise spectrum going down in

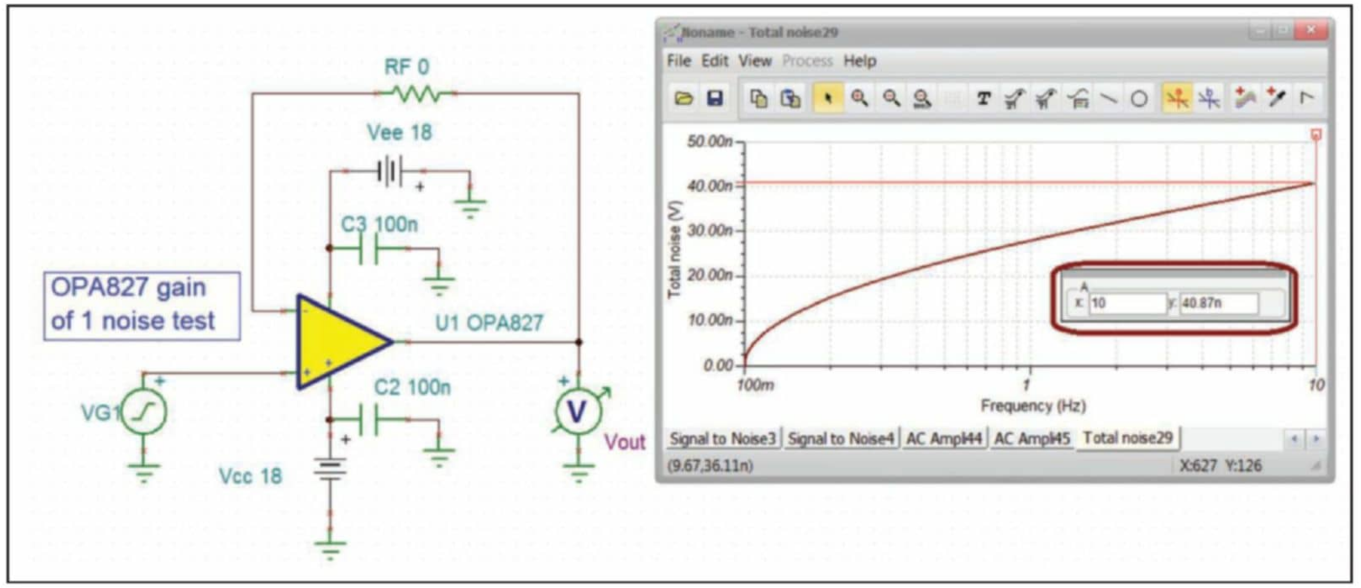
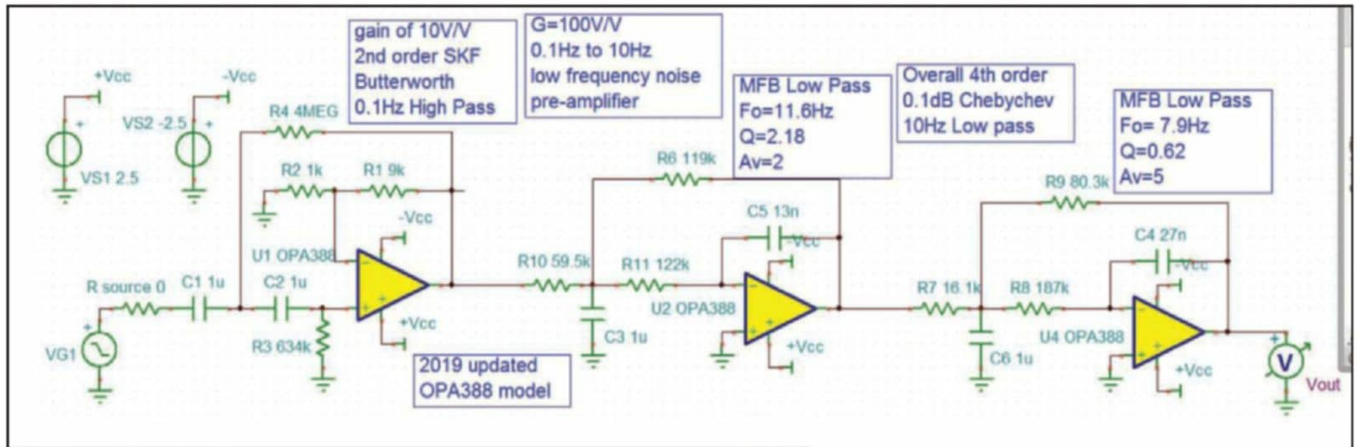


Figure 2: This shows the integrated noise over the 0.1 Hz to 10 Hz span for the OPA827 TINA model at gain of 1 V/V.



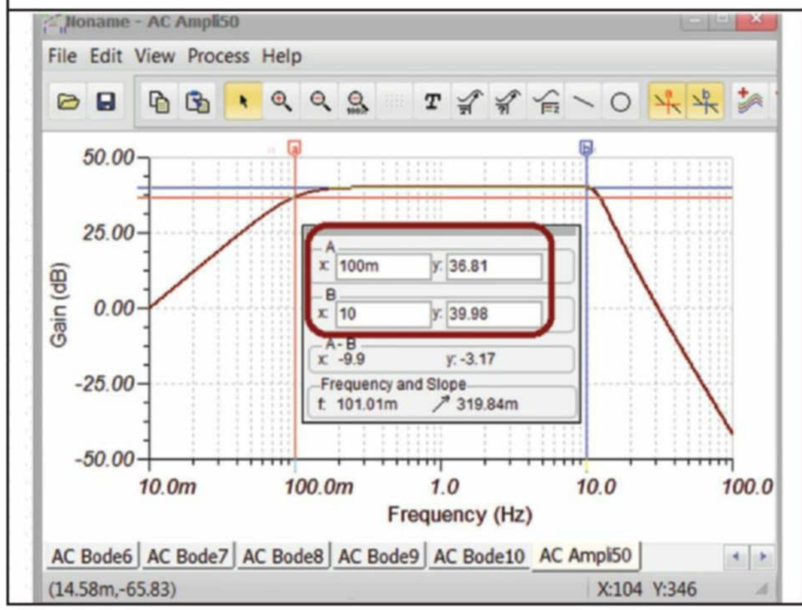


Figure 3: Here we show an updated gain of 100 V/V 0.1 Hz to 10 Hz active filter for low-frequency noise measurements.

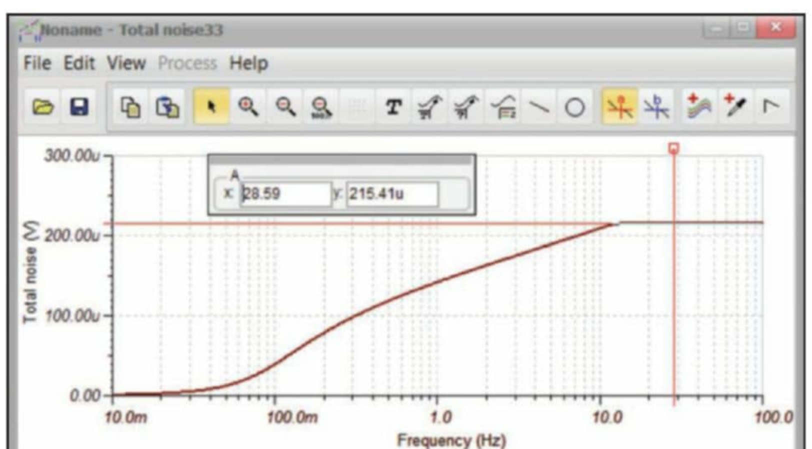


Figure 4: The integrated noise for the OPA827 is set for gain of 51 V/V followed by the OPA388 based gain of 100× active filter noise preamp.

frequency and are themselves popcorn free. This relatively high-speed device offers a flat 7 nV/√Hz input noise with 10 MHz gain bandwidth product. This noise preamp uses a first stage gain of 10 V/V second-order Butterworth high-pass filter at 0.1 Hz. This blocks off any DUT DC offset for later stages. The original designs followed that with a gain of 10 V/V fourth-order Butterworth filter at 10 Hz. Figure 3 shows the most recent low-pass stages implementing a sharper cut-off 0.1 dB fourth-order Chebychev at 10 Hz. The second-order high-pass is rolled off about -3 dB from the mid-band gain of 40 dB at 0.1 Hz. The sharper cut-off 0.1 dB Chebyshev is actually flat at 40 dB gain through 10 Hz then rolls off at a fourth-order rate (-80 dB/decade).

Using this OPA388 chopper amp gives a lower integrated noise through the filter span for this noise preamp itself over earlier implementations. Running a broadband output integrated noise simulation yields only 21 μV_{RMS}. This input refers as only 210 nV_{RMS}.

The device under test (DUT) need only come in with at least 10× this level for this preamp to yield a good measurement. Going back to the OPA827 noise integration at a gain of 1 shows a 41 nV_{RMS} number in Figure 2. To exceed the noise preamp input integrated noise floor by 10× requires the DUT be set to a gain ≥51 V/V. Adding that OPA827 at a gain of 51 V/V in front of the noise preamp gives the total output integrated noise swept over a wider span shown in **Figure 4**. The integrated noise is going flat at a 215 μV_{RMS} level indicating the fourth-order low-pass filter is cutting off all higher frequency noise. That 215 μV_{RMS} input refers (divided by a gain of 5100 V/V) to give 42 nV_{RMS} input referred—closely matching Figure 2. Using a 6× crest factor, that 215 μV_{RMS} noise out of the

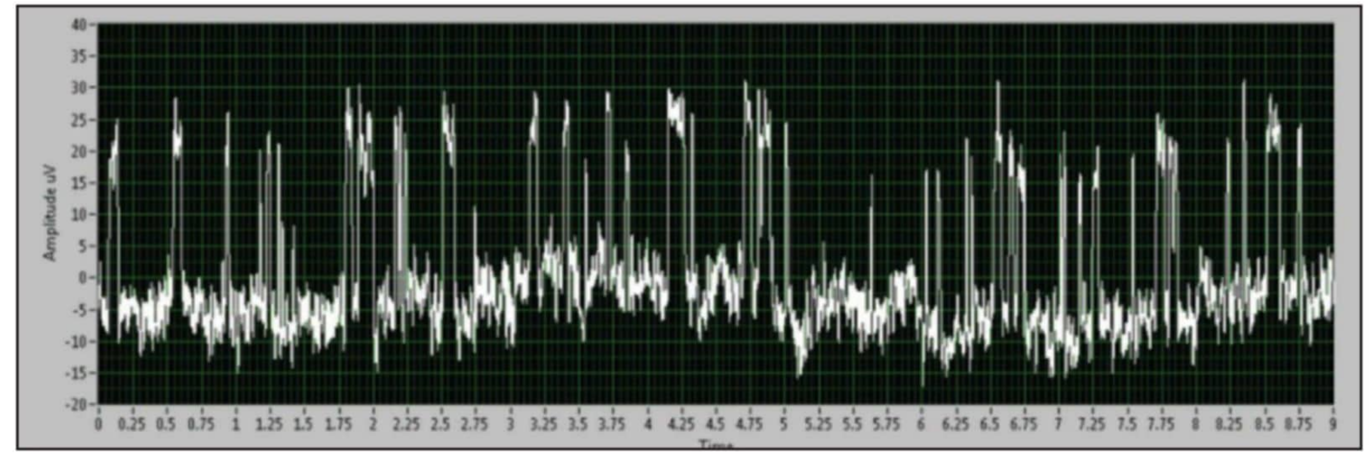


Figure 5: Here is a graphic representative popcorn noise in a returned THS4531A.

preamp will appear as approximately 1.3 mV_{PP} noise—easily measurable. Increasing the DUT gain will increase both the accuracy of the measurement and the measured V_{PP}.

Want Butter with that Popcorn Noise?

So popcorn noise was supposed to be an historical oddity—right? Well not so much—it still appears in many monolithic amplifiers—just at much lower levels than the early days that gave its name coming through speakers. While it may be at an imperceptible level on the speaker driver side, it could pose a limitation on very high dynamic range measurement mic digitizer channels. Where needed, this same low-frequency active filter preamp is used to measure this effect.

For FDA differential output popcorn noise, a low-noise differential to single-ended stage is first needed to get single ended into the noise preamp. Ideally, that would be a wideband chopper based Instrumentation Amplifier (INA). There are a few low-frequency ones, but the recent INA818 (non-chopper) appears to have suitably low 1/f corners (see Figure 22, Reference 5) for this application. Just screen that particular INA818 for popcorn noise itself using the popcorn free preamp of Figure 3.

Recent high-speed amplifier developments have shown a low, but non-zero, incidence of popcorn noise in characterization. Each wafer has a variable incidence of popcorn noise where most sites are popcorn free. Those that are not show a surprising randomness in frequency, duration, and amplitude for these discrete steps in apparent input offset voltage (or current in some cases).

Figure 5 shows a particularly virulent case of popcorn noise in a returned THS4531A low power FDA (see Reference 6 and Reference 7). This input referred plot shows the relatively low (10s of μV) level for these input offset voltage shifts.

Using the low-noise, high-gain active filter, preamp of Figure 3, it is possible, but time consuming, to screen these outliers out. Some production test flows add a 2-second screen for this issue. While helpful, any device with a greater than 2-second pop period might be a test escape. This phenomenon is not limited to Texas Instruments (TI) amplifiers, it also appears across a range of bipolar input amplifiers in the industry. Use the low-frequency noise preamp described here to perform your own validation testing (ideally across multiple units and wafer lots).

References

[1] "Low Noise, High Precision, JFET-Input Operational Amplifier," Texas Instruments OPA827 Datasheet, revised July 2016, www.ti.com/lit/ds/symlink/opa827.pdf.

[2] M. Steffes, "Noise Analysis for High Speed Op-Amps," Burr-Brown Application Note, October 1996, www.ti.com/lit/an/sboa066a/sboa066a.pdf.

[3] A. Kay, "0.1 Hz to 10 Hz Noise Filter," TI Designs, June 2013, www.ti.com/lit/ug/slau522/slau522.pdf.

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Build a Fully Balanced Transconductance Preamplifier

A few years ago, Dimitri Danyuk shared his complex design for a transconductance preamplifier with a balanced JFET input and single-ended output. Now, he has designed an open-loop transconductance amplifier with less parts and easily found through-hole parts.

By
Dimitri Danyuk
(United States)

A few years ago I designed a transconductance preamplifier with a balanced junction field-effect transistor (JFET) input and single-ended output. The preamplifier has an high output impedance and is loaded by a volume control potentiometer with very low resistance. The preamplifier sound is transparent with a good width and depth without the smallest hint of listener persuasion. The preamp itself is rather complex and was built with surface-mount parts and is not easy to construct by untrained enthusiasts.

So I asked myself, if it would be possible to make an open-loop transconductance amplifier with less parts and from readily available through-hole parts. I chose a fully symmetrical topology, which can be found in the equally famous and rare Blowtorch preamplifier by Curl-Thompson-Crump (CTC) Builders. This preamplifier is expensive and well-regarded among a minuscule group of owners. The topology of the preamplifier gain block is shown in **Figure 1**.

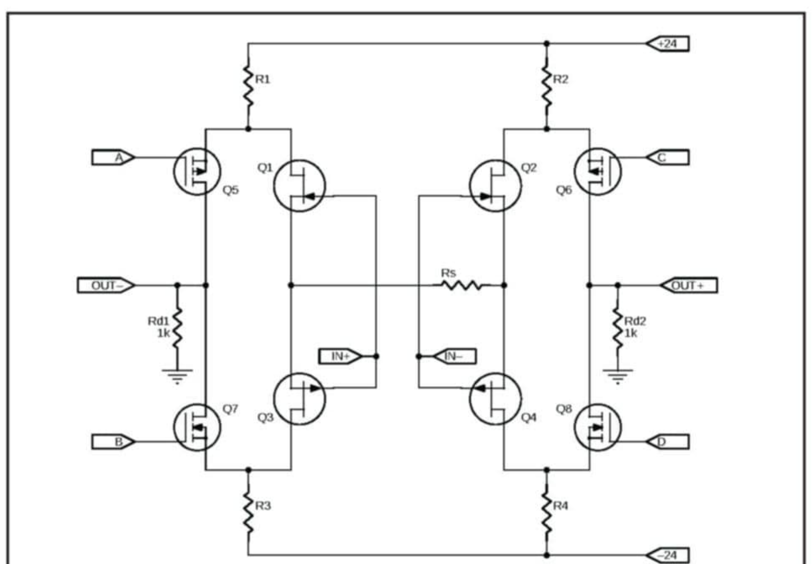

Figure 1: This is the Blowtorch preamplifier gain stage topology.

The Topology

The circuit follows Vendetta Research's SCP1 MC phono stage complementary folded cascode topology. The Vendetta Research's SCP1 MC phono stage is single ended. With the addition of an identical mirrored input stage (Q2, Q4) to the original one (Q1, Q3), we get a fully symmetrical balanced input/balanced output topology.

The second step is to disconnect source networks from the ground and tie them together with resistor Rs. JFETs with the highest possible Idss (2SJ73V/K146V) were used in the Blowtorch. This allows linear operation of the stage. MOSFETs (2SJ76/K213) and JFETs were connected as folded cascode stages, in the same manner as in the Ayre V3 amplifier. (For more information about this topic, see Resources.)

The voltage between inputs is almost equal to the voltage drop on source resistor Rs. The value of Rs is chosen considerably higher than transconductance of Q1, Q3 and Q2, Q4. Source resistor Rs decreases JFET stage transconductance with a correspondent increase in linearity of transfer characteristics. Devices Q5-Q8 act as common gate stages with unity current gain. Common gate stages maintain stable voltage on the drains of the input stage, thus reducing distortion associated with nonlinearity of JFETs output characteristics.

My realization of the symmetrical balanced transconductance gain stage is shown in **Figure 2**. I chose the amplifier's input stage to be comprised of readily available n- and p-channel JFET matched pairs (LSK489/LSJ689). LSK489/LSJ689 devices have desirable qualities of low voltage noise, low current noise, low input capacitance and high input impedance.

JFET pairs work as a self-biasing source followers, being loaded with bipolar transistor common emitter stage (Q3, Q4, Q11, and Q12). Four bipolar transistors (Q9, Q10, Q13, and Q14) are connected as diodes and supply bias to the bipolar junction transistor (BJT) stage (Q3, Q4, Q11, and Q12). Each transistor, connected as a diode, has a thermal contact with the correspondent amplifying device. The voltage drop across diodes (Q9, Q10 and Q13, Q14) also sets quiescent current through JFETs. The overall transconductance of the stage is set by R18.

The value of R18 establishes collector currents in Q3, Q4, Q11, and Q12 being proportional to the input signal. The collector currents are equal without the input signal. Currents through R14, R15, R25, and R65 split between Q3, Q4, Q11, Q12 and Q5-Q8. Resistors R14, R15, R25, and R65 set quiescent current through common base devices (Q5-Q8). Having essentially unity current gain and low distortion, the common base devices Q5-Q8 shield the gain transistors Q3, Q4, Q11, and Q12 from voltage changes on the output terminals. Resistors R19 and R28 function as load resistors. I decide to use the same value for load resistors (2 × 1 kΩ) as in the Blowtorch. The gain of the symmetrical balanced transconductance gain stage is equal to 4 (12 dB).

The servo circuit is shown in **Figure 3**. The servo circuit has two channels, similar to the Blowtorch

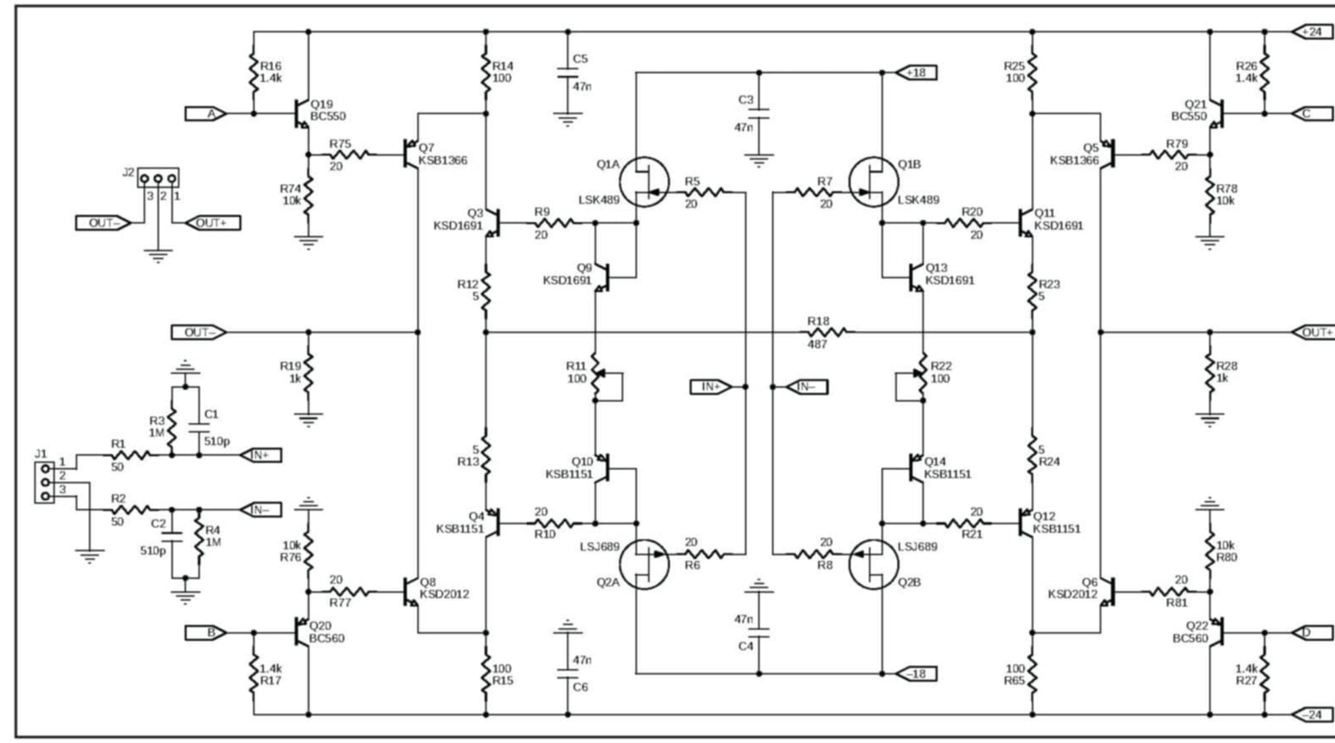

Figure 2: This is the symmetrical balanced transconductance gain stage.

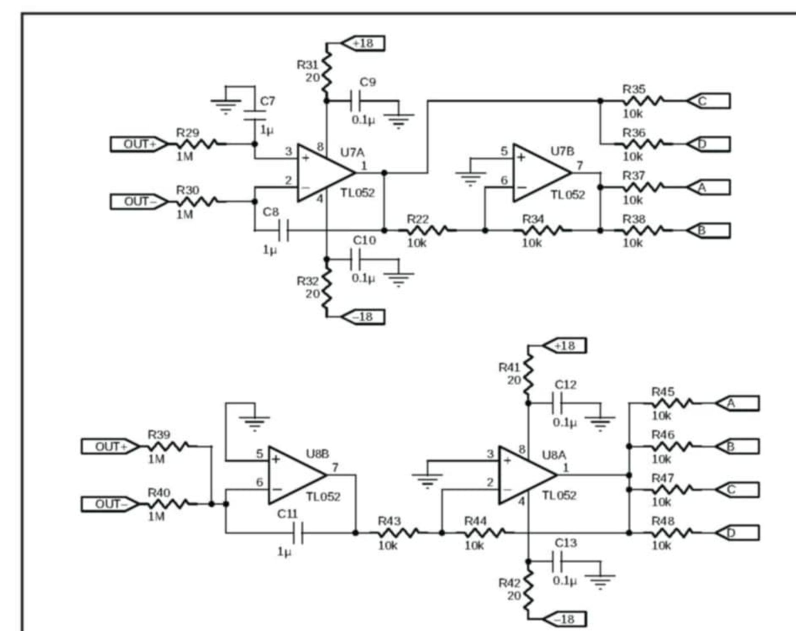

Figure 3: The DC servo circuit has two channels.

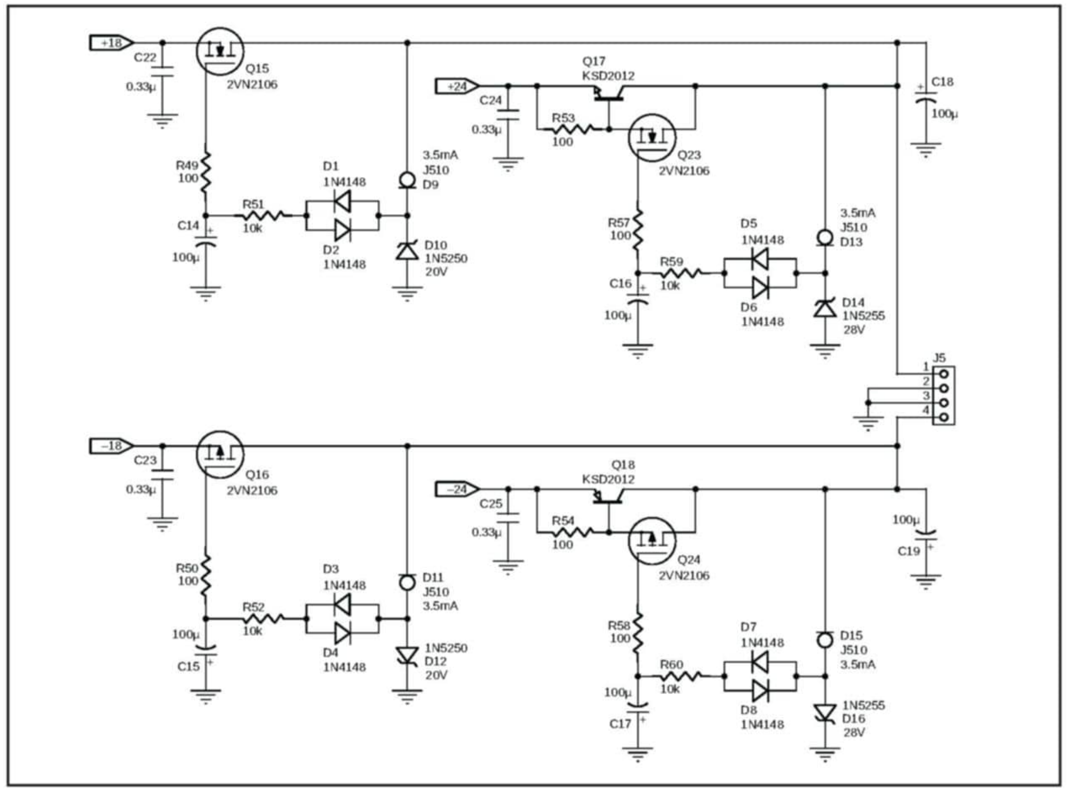

Figure 4: The power supply is based on conventional source/emitter followers with Zener diodes as references.

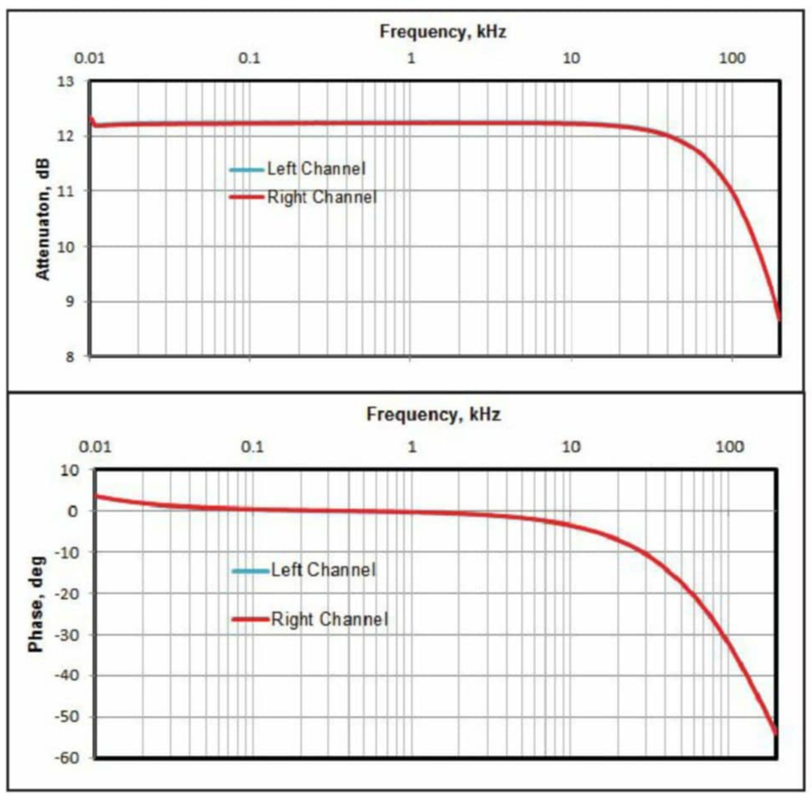

Figure 5: The amplifier's frequency and phase response

servo. Common mode servo channel (U8) senses the DC output voltage with respect to ground and makes it equal to zero, while differential servo channel (U7) maintains zero DC voltage between outputs. The power supply circuit is shown in **Figure 4**. It is based on conventional source/emitter followers with Zener diodes as references.

The schematic and PCB design made in ExpressPCB format is available on the *audioXpress* web server. A dozen of boards were manufactured according to these design files, stuffed, and tested.

The Amplifier's Performance

The amplifier's objective performance is very good. The bandwidth extends above 150 kHz with the phase shift less than 10° at 20 kHz. The most important measurement results are shown in **Figures 5-9**.

The distortion figure remains constant with the signal frequency up to 5 kHz. There is a 10 dB increase in distortion when the signal frequency rises up from 5 kHz to 20 kHz. The typical distortion spectrum is shown in Figure 8 and Figure 9; the even harmonics are largely suppressed. I can say that the distortion figures are excellent for the circuit without overall feedback.

I measured the noise density at 1 kHz around 38nV/√Hz, which corresponds to the noise of 90 kΩ resistor. This is considerably higher than

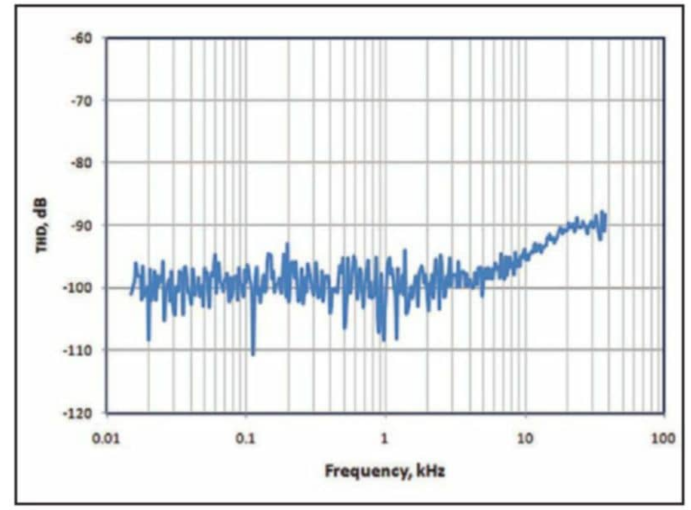

Figure 6: Total harmonic distortion at 4 V_{RMS} output (1 V_{RMS} input)

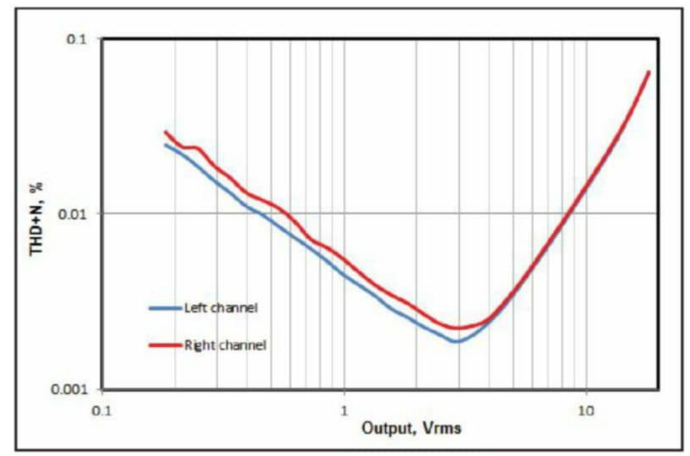

Figure 7: Total harmonic distortion plus noise (THD+N) at 1 kHz.

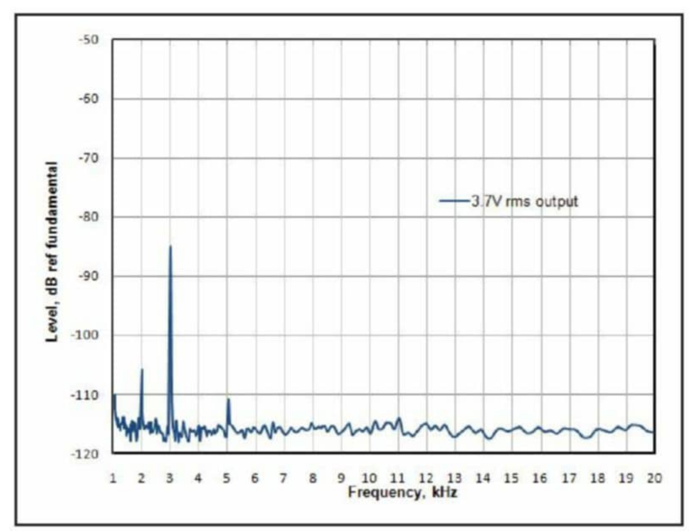

Figure 8: Distortion spectrum for 1 kHz sine wave, 4 V_{RMS} output, 1 V_{RMS} input

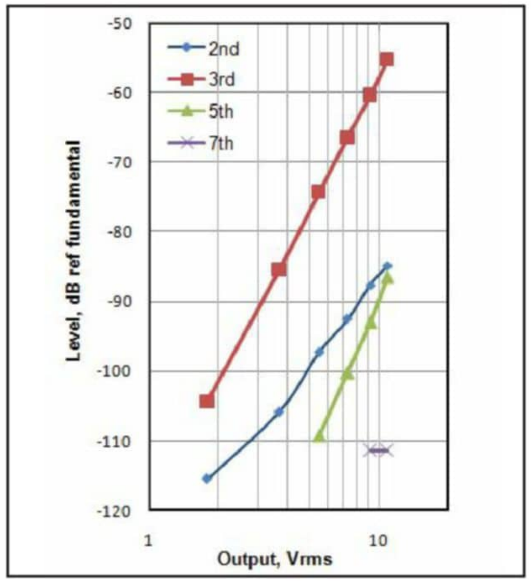

Figure 9: Individual harmonic distortion (order of harmonic as parameter)

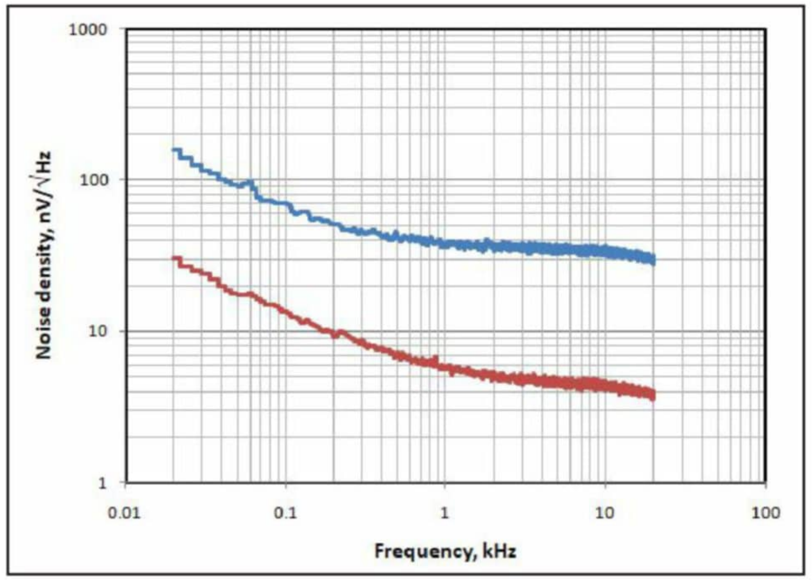

Figure 10: The input noise voltage spectral density (the red line is with additional 47 µF capacitors across R16, R17, R26, and R27)

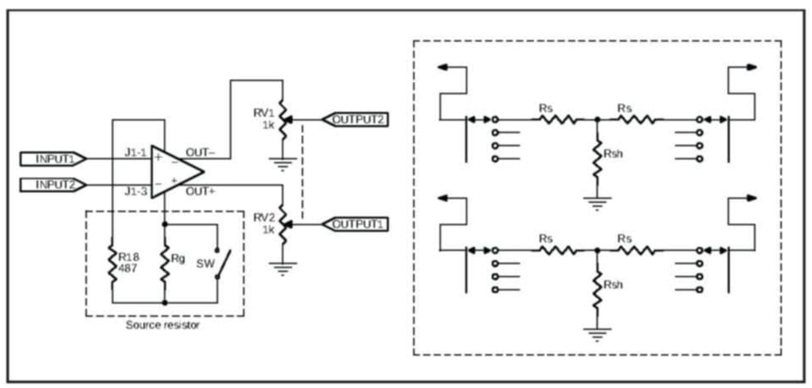

Figure 11: This shows the recommended volume control arrangement.

the noise, produced by input JFETs and gain setting resistor R18. Evidently, the cascode devices (Q5-Q8) contribute noise since they have small value resistors (R14, R15, R25, and R65) in emitters. Voltage noise of bias dividers (R16, R17, R26, R27, R35-R38, and R45-R48) being multiplied by factor of 20 (2 × R19/R14) applies to the output terminals (see **Figure 10**). If resistors R14, R15, R25, and R65 are replaced with the current sources, then folded cascode devices experience source degeneration and contribute noise to a much less degree. Simplicity is an important consideration in a design, but over-simplicity is harmful.

Volume Controls

The preamplifier can be equipped with a conventional balanced volume control (dual gang logarithmic potentiometer) in front of it. The better solution is shown in **Figure 11**, where the dual gang logarithmic 1 kΩ potentiometer Rv1, Rv2 operates as a load for the transconductance stage (resistors R19 and R28 are not populated). One can also use a dual ladder step attenuator, a dual "T" pad attenuator, or a dual shunt volume control, preferably with rotary switch with silver coin contacts. I would recommend that builders implement the switchable gain setting resistor R18 (as shown in Figure 11) if 12 dB of gain is too high. Additional switchable series resistor Rg reduces the overall gain to desirable level.

Additional Notes

I populated the boards with conventional high-grade parts: CMF/CCF Vishay/Dale and MB* Vishay/Beyschlag resistors in audio path, Wima MKP2 and Nichicon audio-grade capacitors. I prefer the sound of transistors with large dies in circuits without overall feedback. I use devices with high current ratings (e.g., KSB1366/KSD2012 and KSB1151/KSD1691). I also get good results with TTA004/TTC004 and TTA008/TTC015 pairs.

After the build, I attempted to reduce the noise level by shunting R16, R17, R26, R27 resistors with 47 µF aluminum electrolytic capacitors. The noise dropped from 38nV/√Hz to 6 nV/√Hz at 1 kHz (see Figure 10). These capacitors are not captured on the schematic shown in Figure 2. The voltage noise spectral density for supply rails is shown in **Figure 12**. The shape of the curves is typical for MOSFET devices.

Subjectively, this open-loop high-quality transconductance amplifier has very good transparency and stage. I sincerely hope that building, modifying, and upgrading this design will lead to an exciting journey into the world of open-loop audio circuits.

Author's Note: audioXpress*' Technical Editor Jan Didden suggested checking the noise on the power supply rails. Thank you Jan.*

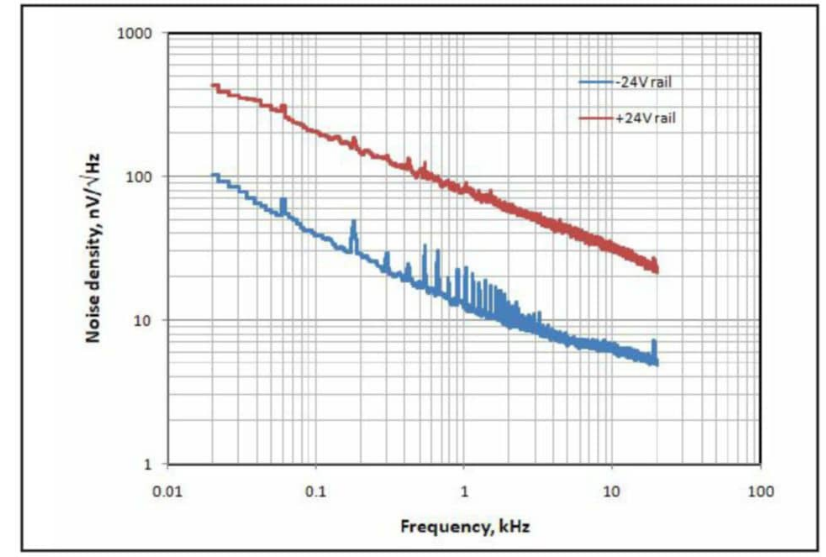

Figure 12: This shows voltage noise spectral density for the supply rails.

Project Files

To download the bill of materials (BOM and other files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Resources

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About the Author

Dimitri Danyuk has been the principal hardware engineer at Harman luxury audio since 2013. He graduated with honors from Kyiv Polytehnic in 1985. His passion is to help audiophiles in all aspects of audio design. His articles have appeared in a number of magazines including the *Journal of the Audio Engineering Society*, *IEEE Transactions*, *EDN*, *Electronic Design*, etc. Danyuk also provides consulting services for various businesses.

Hunting the Elusive Transient

In this article, our longtime author describes a few different types of drivers, systems, and tests that can help reveal the significance of transients. Despite inventive digital tests, there is still lots of room for better correlation with how we hear. It may also lead to new methods of testing.

By
Roger Russell
(United States)

What is a transient? In acoustics and audio in general, a transient is a high amplitude, short-duration signal and is often at the leading edge of a waveform. We are exposed to all of these sounds each day–whether it is a complex wave shape such as music, noise, voices, bells, or even hand claps. What has all this got to do with speakers? They are required to reproduce transients for accuracy. Although emphasis is often not widely placed on speaker transient response, it is often considered to be as important, if not more important, than frequency response.

A speaker system can have excellent frequency response, good dispersion and low distortion but still not have good dynamics. Instead of reproducing peaks in program material, the sound can still be dull and lifeless because it cannot correctly reproduce a transient signal from the amplifier. In addition, transients are not just limited to high frequencies. Transients can be found at lower frequencies and be just as significant for a bass drum as for a triangle and beyond. A few different types of drivers and systems are described that can help to reveal the significance of transients.

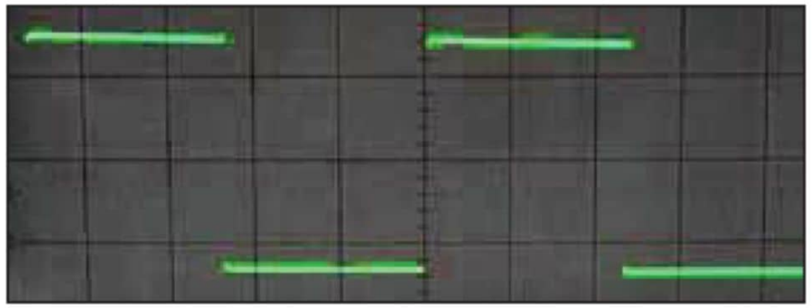

Photo 1: This 400 Hz square wave is at the power amplifier output.

Square Waves

The word transient might bring to mind a square wave (see **Photo 1**), which has been used in amplifier testing to demonstrate wave shape, ringing, and so forth, or even used as a marketing tool. The 400 Hz square wave shown in Photo 1 is at the power amplifier output. The components are the fundamental plus the odd harmonics (one-third of fundamental amplitude for the third harmonic plus one-fifth of the fundamental for the fifth harmonic and so on). The more harmonics that are added, the more the squareness improves. However, there are diminishing returns for the leading edge as it approaches a perfectly square corner. Many good amplifiers today might exhibit good square wave response for selected frequencies. A frequency of 400 Hz is chosen in this simple example because there is sufficient bandwidth of audible frequencies on either side that will contain enough harmonics and subharmonics to create a reasonable-looking square wave.

However, no musical instruments can produce a perfect acoustic square wave. Mass of the moving parts prevents the instant acceleration that is needed. Even when playing a recording through speakers, any further transient information could be lost. Although relatively subtle, improving a signal containing transient information can definitely be heard by the listener.

Tone Bursts

The General Radio 1396-A Tone burst generator (see **Photo 2**) is actually a gating device that allows various repeating signals to be transmitted for different periods of time. Tone-burst tests have been available for many years to show how well a speaker can perform. Many years ago, emphasis was placed on tone burst tests for loudspeakers. Around 1960, I recall going on an Audio Engineering Society (AES) tour of the Consumer Reports facility in Mt. Vernon, NY. During the tour, I noticed that Larry Seligson, a project engineer working there, was using a 1396-A Tone Burst Generator to test loudspeakers. Typical tests can be made in a reflection-free environment such as an anechoic chamber that simulates a free field. **Photo 3** shows one cycle of a gated 400 Hz sine wave. The length of time for one complete Hertz is 2.5 ms equivalent to a wave length in air of 2.81′ (33.7″).

One Hertz of a gated sine wave may not be adequate for a complete evaluation, such as ringing or overshoot. Several Hertz may be needed to verify the complete response, as shown in **Photo 4**, but keep in mind that a short burst minimizes any room reflections that might otherwise interfere with the measurement. A display of many Hertz crammed together in one burst might make the display look more favorable but can hide the real transient information.

Shaped Tone Bursts

The question arises about the degree of accuracy in reproducing a tone burst. The instant on-and-off comparison presents a problem when gating a signal. A listening test reveals an audible clicking sound that can be heard even when the trigger level is set to a zero crossing. It can also be displayed on a spectrum analyzer (see **Figure 1**).

With a conventional tone burst generator, a considerable number of unwanted harmonics are generated. These extend to sub-audible frequencies, and excite resonances that have nothing to do with the actual test frequency as mentioned by Siegfried Linkwitz (see Resources).

A shaped burst gradually increases the amplitude of each Hertz of the test waveform and then decreases it again. It can reduce the unwanted harmonics by using a one-third octave cosine waveform envelope shown in **Figure 2**. For a 5 Hz burst, either side of the center frequency is now down by 30 dB. The response to a standard tone burst can then be compared to a shaped burst and reveal how accurately drivers respond without interference. Linkwitz shows that a response using the maximum value of the shaped bursts can be measured in ordinary closed rooms and have a very high correlation to free-field measurements. It also can be used for high power bursts.

A More Sophisticated Design

Typically, drivers are mounted one above the other on the same plane or on a front board. The sound from each driver could be expected to arrive to the listener at the same time. Although the speed of sound is the same in air for all the drivers, the effective point of radiation is not the same. Repositioning the drivers in a new stepped cabinet can compensate for the different arrival times to the listener (see **Photo 5**) and restore some of the transient information.

The acoustic center is the effective radiation point often found near the plane of the driver voice coil. I separately measured the acoustic center at 1 m directly in front of each driver in a reflection-free environment. I used a Tektronix storage oscilloscope that includes a delay line. It is triggered by the

Photo 2: The General Radio 1396-A Tone burst generator is actually a gating device that allows various repeating signals to be transmitted for different periods of time.

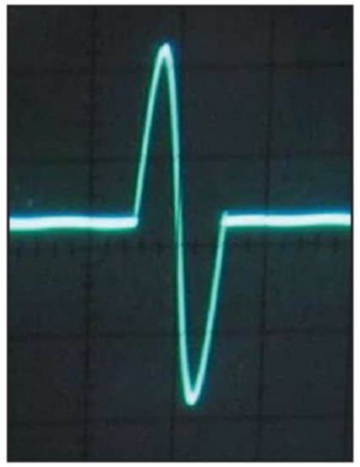

Photo 3: This is one cycle of a gated 400 Hz sine wave.

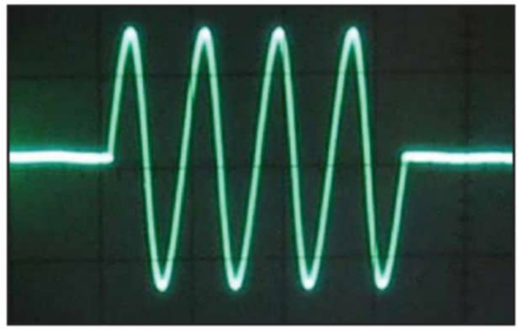

Photo 4: Several Hertz may be needed to verify the complete response.

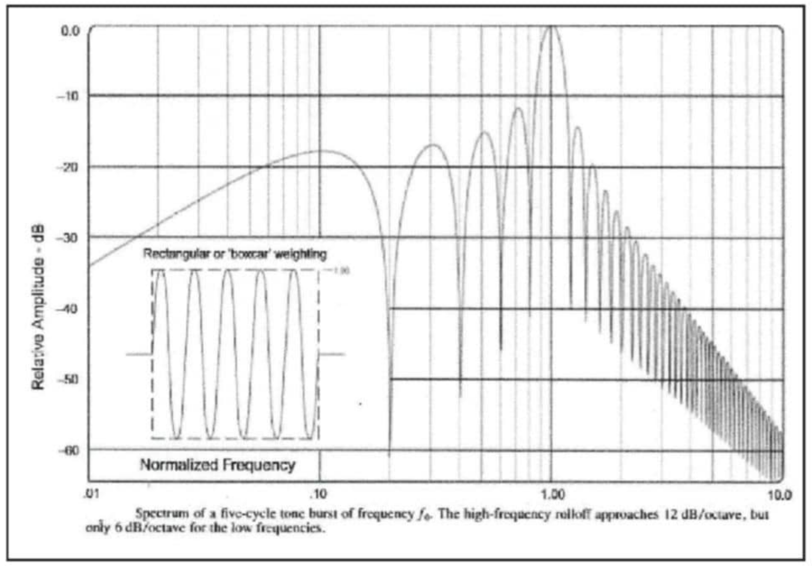

Figure 1: An audible clicking sound that can be heard even when the trigger level is set to a zero crossing. It can also be displayed on a spectrum analyzer. (Image courtesy of Audio Engineering Society, aes.org)

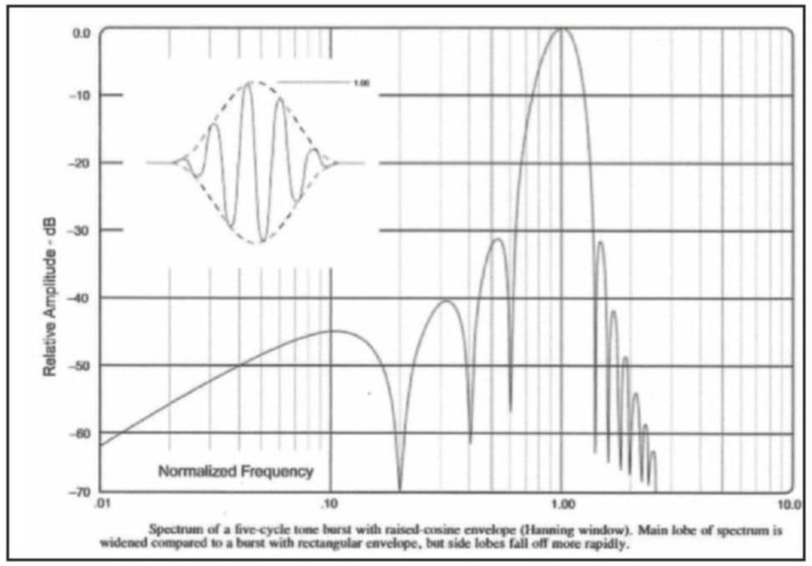

Figure 2: A shaped burst can reduce the unwanted harmonics by using a one-third octave cosine waveform envelope. (Image courtesy of Audio Engineering Society, aes.org)

About the Author

Roger Russell graduated from Rensselaer Polytechnic Institute (RPI) with a bachelor's degree in electrical engineering. He was a student member in the Audio Engineering Society (AES) and is now a life member. He is proud of RPI for its advancement in combining science and music, giving a fresh look in the relation between the two. His interest in sound and music began in grade school with learning to play the violin. He also spent time in high school building amplifiers and speaker systems to improve reproduced sound.

As an adult, he was a senior engineer in audio products at the Sonotone Corp. and Director of Acoustic Research at McIntosh Laboratory. Retired now and living in Florida, he continues with research on hearing and speaker design and has written many articles about it.

microphone output and the time recorded. Generally, the arrival time is shorter for tweeters and longer for woofers. To compensate with the new cabinet, the tweeter is mounted further back and the woofer is the closest. The mid-range is located part way back. I used this data to design and build a three-way time-compensated system that is based on the acoustic arrival time for each of the three drivers. The improvement of transients and overall sound is very noticeable. However, a slowly increasing error can be heard as you listen to the system further off axis, whether up, down, or sideways. Optimum placement for the systems is where the tweeter height is opposite from the listener's ears when sitting and the systems are toed in to face the listener.

Coaxial Drivers

These older drivers actually had the right idea years ago, whether by intension, economics, or space savings. For example, the 1950 Sonotone CA-15 coaxial speaker shown in **Photo 6** has the tweeter assembly and supports mounted in front of the woofer. This arrangement places the tweeter acoustic center in a more favorable position with respect to the woofer. First-order crossover components (the coil and the capacitor) are attached behind on the frame of the woofer. The RF-475 Coaxial Speaker (see **Photo 7**) has a horn tweeter used for the high frequencies. The tweeter voice coil is in a favorable position behind the woofer voice coil.

Photo 8 shows the Altec 604C duplex speaker. This 15″ coaxial speaker was in demand by recording studios for many years after it was introduced. Meanwhile, several other coaxial drivers were introduced that did take greater advantage of time alignment, such as the Stephens 5106 coaxial speaker, the Tannoy Dual Concentric, and the University 6201 coaxial speaker. They all have the tweeter voice coil located in a favorable position behind the woofer voice coil making the tweeter acoustic center align closer or behind that of the woofer. This can contribute to improved time alignment. Of course, the two separate drivers still require a crossover network, however simple.

Recently, companies such as KEF have what is called a Coincident Driver, where a small tweeter is placed in the throat of the mid-range or woofer. This is an alternate and less expensive method to keep acoustic radiation more like a coherent source. In addition, some car systems, due to space-saving necessities, have a small dome tweeter attached to the pole-piece at the center of the woofer.

Others, such as the 8″ extended-range driver shown in **Photo 9**, have a small conical-shaped whizzer and is attached to the same voice coil as the woofer. This presents a very economical way of extending the highs and is referred to as having a mechanical crossover. Some also come with a solid-shaped "phasing plug" mounted on the center pole piece but no data is available on the location of the acoustic center(s).

Another Path to Transients

Photo 10 shows a unique design found for transients in a single, long column composed of 25 drivers, which I discussed in a 2006 *audioXpress* article (see Resources). These wide-range units are all identical and take advantage of a 2-5/8″ effective cone diameter for exceptional mid- and high-frequency response plus wide dispersion. The weight of the moving parts for each driver is close to a dime (2.5 g or 0.08 oz.). Although not measured, the acoustic center of such a small driver can show relatively little variation in arrival time. A distinct listening advantage is that the same sound appears to come from the area closest to your listening height. It is explained by the precedence effect, also known as the Haas Effect.

Another bonus is that the combined cone area of all the drivers is equal to that of a 16″ woofer. Then, with a little help from electronic equalization prior to a power amplifier, low-frequency response is extended to 20 Hz and the highs to 20 kHz. Some peripheral beliefs are that good bass notes cannot be reproduced by small drivers and that long columns produce response effects known as comb

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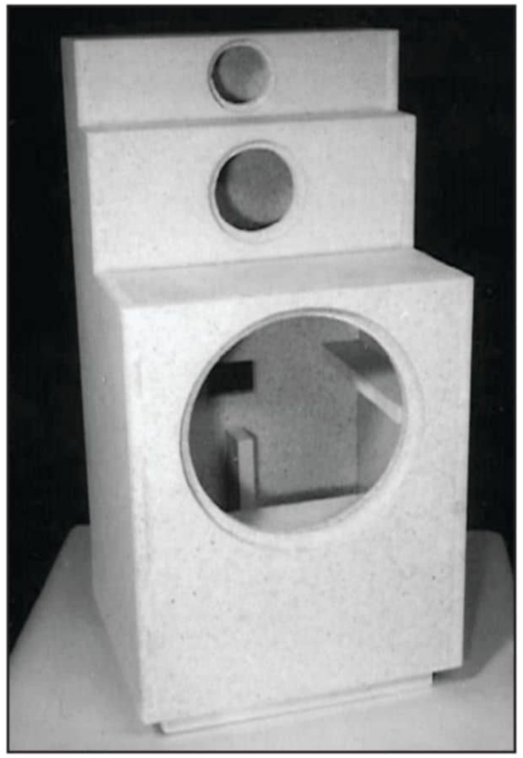

Photo 5: Positioning the drivers in a stepped cabinet can compensate for the different arrival times to the listener and restore some of the transient information.

Photo 6: This 1950 Sonotone CA-15 coaxial speaker has the tweeter assembly and supports mounted in front of the woofer.

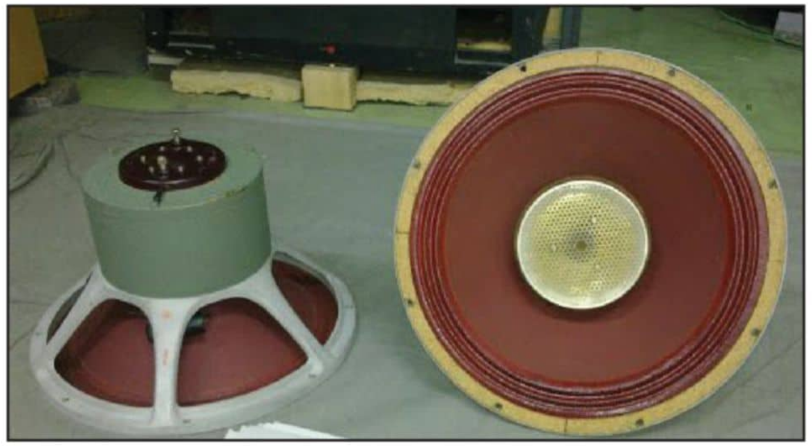

Photo 7: The RF-475 Coaxial Speaker has a horn tweeter used for the high frequencies.

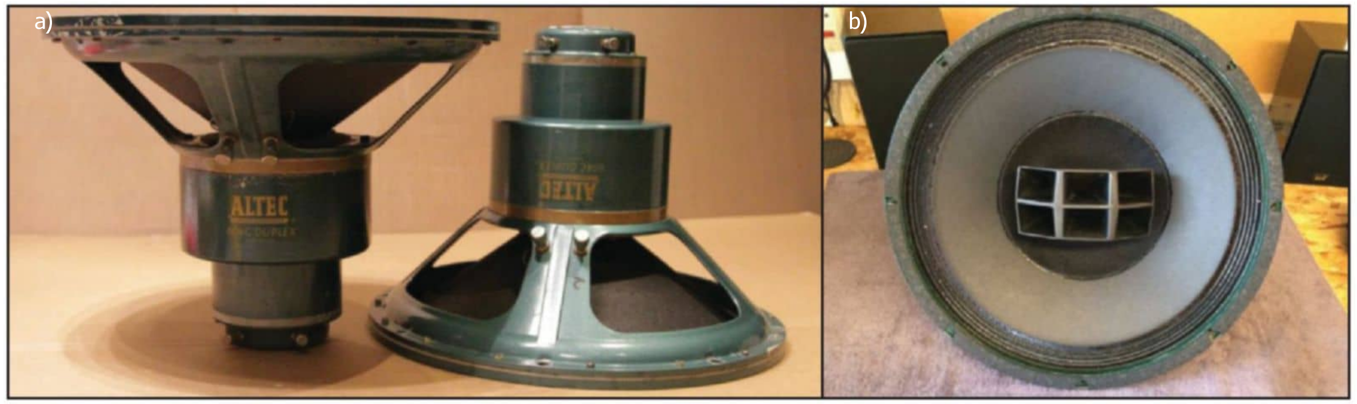

Photo 8: The Altec 604C duplex speakers was in demand by recording studios many years after they were introduced (a). The speakers have the tweeter voice coil located in a favorable position behind the woofer voice coil making the tweeter acoustic center align closer or behind that of the woofer (b).

Photo 10: This is a unique design found for transients in a single, long column composed of 25 drivers.

Photo 9: This 8″ extended-range driver has a small conical-shaped whizzer attached to the same voice coil as the woofer.

filtering. However, in practice this is proved to be false. No crossovers are needed. The transients are outstanding. After listening to a pair of these systems, some opinions are that the sound is like electrostatic speaker systems at their best. Electrostatics are often associated with excellent clarity and transient response.

Summary

A few different types of drivers, systems, and tests are described that can help to reveal the significance of transients. Although other systems can measure well in conventional tests, such as response, distortion, and dispersion. Nevertheless, clarity and liveness are subtle characteristics of a well-designed driver or a well-designed system that can capture much of that elusive transient. In addition, transients are not limited only to high frequencies.

Transients can be found at lower frequencies and be just as significant for a bass drum as for a triangle and beyond. Despite inventive digital tests, there is still lots of room for better correlation with how we hear. It may also lead to new methods of testing to be available for the inventor.

The Haas Effect

After hearing the first signal arrive at the listener, the brain will suppress any later signal (e.g., an echo) for a time up to about 30 ms or 40 ms even if the second arrival is as much as 10 dB higher. The inhibition is called time or temporal masking. A classic example to demonstrate temporal masking is to stand in front of a hard surface (e.g., a brick wall, which is away from other reflecting surfaces that might interfere with what you hear). By clapping your hands and gradually moving back, you will reach a point where you can begin to discern two separate sounds, your initial handclap and the echo. I used two small boards because my hands got tired of clapping. When I tried this, I measured 17′ to the wall. The total distance, representing the time it takes for the sound to travel to the wall and back again, was 34′ or 30.6 ms.

Powering Your Circuits (Part 6)

Power Transformer Rating, Selection, Fusing, and Safety

By
Ed Simon
(United States)

This is the sixth and final article in this series on ways to power circuits, which has covered battery power, AC power supplies, DC power regulation, and diodes. In the conclusion of his series, Ed Simon covers the unavoidable topic of transformers and safety issues.

The question often comes up as to what size transformer is required to power a known load. Transformers are rated in volt amps (VA). This is not quite the same as watts! A watt is the voltage times the current times the phase angle between them. If you understand that, it is the current that ultimately will melt the transformer wire even if it is almost completely out of phase with the voltage, then you can understand why the rating is not the same. So knowing the output voltage and the VA rating, you can easily calculate the maximum current draw.

A rule of thumb is for a dual-diode center-tapped power supply to use 1.2 times the current draw times the transformer's secondary voltage to get the VA rating required. For a full-wave bridge make that 1.8! Now if you are doing a high voltage power supply and using a choke, that will lower the output voltage and reduce the correction factors required to 0.7 and 1.0.

Since there is variation in load during use, transformer construction and resulting efficiencies, and you were very frugal, you might get away with a transformer up to 10% smaller. The 8% variation in equally rated coils shown in Figure 9 from Part 3 of this article series (*audioXpress*, April 2020) should give you a good hint at transformer tolerances! But be sure you really do know the maximum load.

If you are designing high-end audio products, you probably want to size it larger, both for the reduced noise and tighter regulation. How much larger is your choice, there are no hard and fast rules. If you feel the heat, see smoke, or get an outright transformer failure, you probably made a bad design choice.

Many older electronics were designed for an AC line voltage of 110 V, so with more modern AC distribution it is common to see secondary voltages higher than shown on the design schematics and sometimes even higher than the rated voltage of the filter capacitors! This will also show added line distortion from the saturation at the AC line voltage peaks.

Power Supply Safety

The most important issue in any power supply is safety. The R-core unit flunked this test out of the box! It had three input leads, one for common, one for 120 V operation and one for 220 V. The 120 V lead was green! This color is used in the US for the safety ground wire. So to avoid any issue, the lead was removed. Most transformers will show that they are made in compliance with safety standards and bear a UL or other testing lab certificate of listing. Be wary of very cheap transformers, as they often do not meet any safety standards. Small batch custom transformers are often not listed due to the extra cost, so be sure you know who is actually making those transformers.

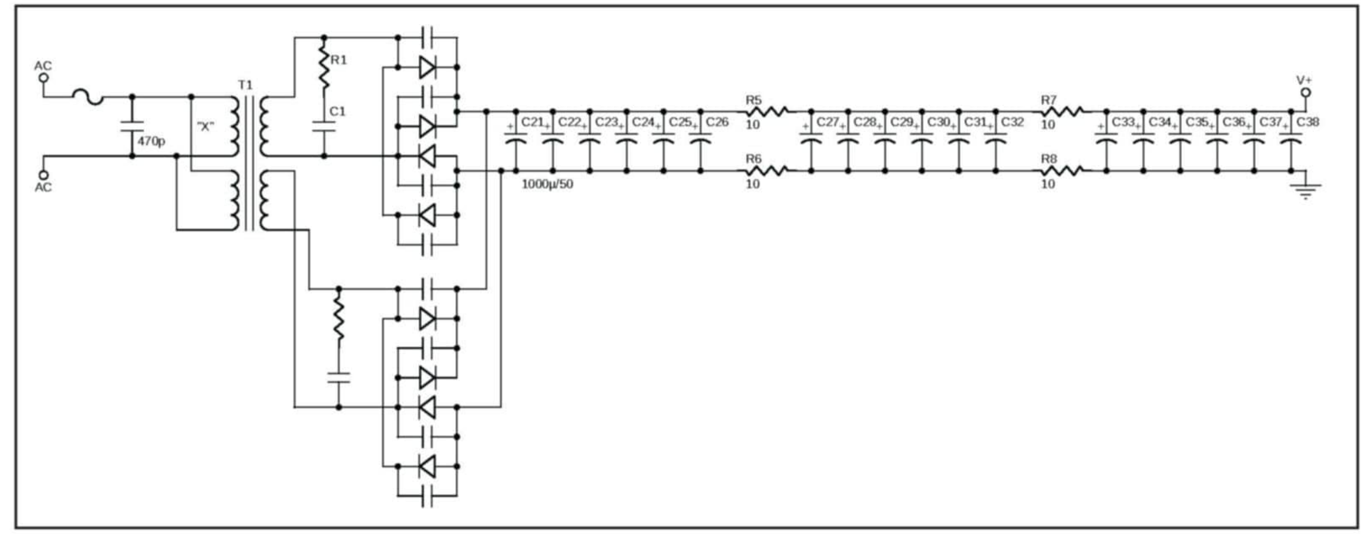

Figure 1: I built this power supply as an external supply for a microphone preamp.

Fuses

The next safety issue is selecting a fuse that will open if the circuit malfunctions but does not open under a starting surge. The easiest and most accurate way to install a fuse is to place it in line with the unit and apply a 200% rated load to the secondary. The fuse should blow in under 10 seconds. If it blows instantly, you tried too small a value. If it did not blow in under 10 seconds, the fuse has too high a current rating. Trial and error will get you to a nice safe fuse value. Those faint of heart can start with low values and work up. With a current probe and modern digital oscilloscope you can, of course, measure several starting surges and use the fuse manufacturers' datasheets to pick a value. Or, you can just use a fuse with a current rating equal to 80% of the transformer's VA rating divided by the line voltage.

Since a toroidal transformer start-up current is influenced by the actual phase at turn on, you will have to repeat the switch-on test many times to be sure you have the right fuse rating.

For very small transformers, you might get away with a normal "Fast Blow" fuse, but usually a medium delay (MDL) type is best. For audio power amplifiers or other large loads you might want to use a "Slow Blow" fuse. Of course, be sure the fuse has a voltage rating greater than the line voltage. Many of the fast blow 1/4" × 1 1/4" round fuses are only rated for 250 VAC up to 10 A. Slow blow may only be rated up to 8 A at 250 V. Most fuses do not have a DC voltage rating. When a fuse blows, there is an arc that should be confined to inside the fuse. This limits the maximum current a fuse can stop. Typical ratings are 10,000 A at 125 V and for the same fuse only 35 A at 250 VAC. Interrupting an AC current is easier than a DC current as the current and voltage drop to zero during the normal power cycle. This gives the arc a chance to extinguish.

You may also find that some fuses are sold without a UL or similar rating. These do cost less, but are probably only a good choice for test fuses.

The other issue with fuses is that they must have some resistance in order to work. Often this is actually greater than the resistance of the AC power line all the way from the generating plant through all the wiring, including your power cord and connectors! I measured some small fuses and found that the resistance increased as the current through the fuse increased. It was not a linear change. A fast blow fuse rated at 0.5 A started off with a resistance of 1.24 Ω at 0.025 A and increased to 1.93 Ω at 0.5 A! You can expect the resistance at an AC outlet to drop less than 5% from no load to full load. So a 15 A, 120 V outlet should exhibit no more than 0.4 Ω of resistance. In that case, the fuse would cause more voltage loss than all of the AC power wiring!

The worst fuses for voltage drop are ones rated around 0.5 A. Depending on the fuse type and make, this can be as high as 5 V. Fortunately for most fuse sizes and types, it is only around 0.25 V of loss. This data can be found in the fuse manufacturers' datasheets.

Examples

Figure 1 shows a power supply I built as an external supply for a microphone preamp. Note the capacitors (C1, C2) across the transformer secondaries, the damping resistors (R1, R2), the use of multiple small filter capacitors instead of a single large one, and the resistors to improve the filtering. There is also one new trick—the secondaries are wired out of phase with each other. In **Figure 2**, you can see a comparison of the AC noise, measured from rail to rail of the power supply, with the secondaries wired in and out of phase, as well as with the power supply unplugged.

A few other details not shown in just the power supply schematic is that at the point of load (the

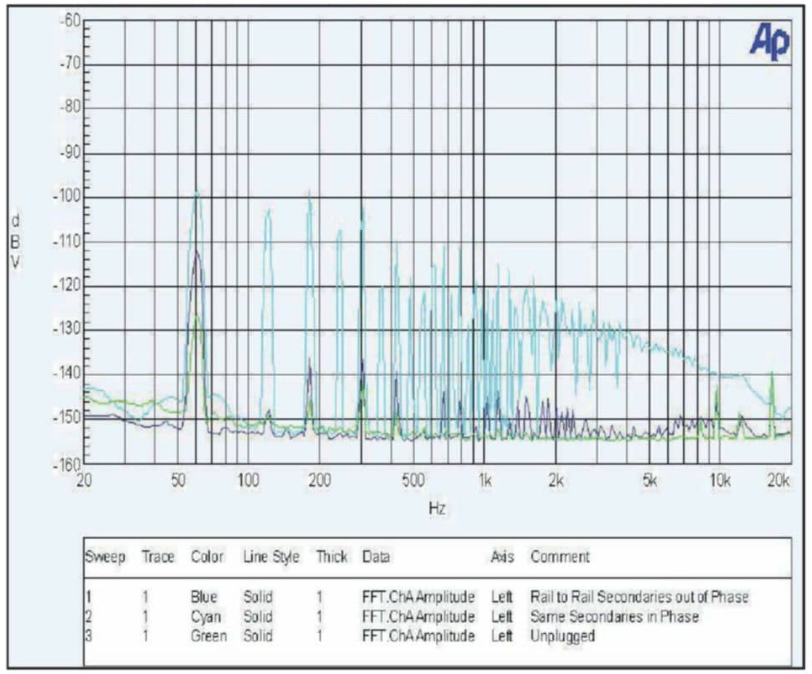

Figure 2: This graph compares the power supply measured from rail to rail for AC noise, between the power supply unplugged and with the secondaries in or out of phase.

mic preamp card) a large (1000 µf) capacitor is placed from + rail to − rail to allow both rails to swing evenly from an offset load. Of course, there are also small bypass capacitors at several load points. As you can see, using all of these techniques provides a much quieter power supply.

Having reviewed using linear power supplies to energize your projects, we will look at providing protection for your circuits in a future *audioXpress* article.

About the Author

Ed Simon has a BSEE from Carnegie-Mellon University. He was employed by Cameradio, a regional electronics parts distributor from 1973-1975. In 1975, he started Edward Simon & Co. He has published more than 40 articles and taught courses for the NSCA, Carnegie-Mellon University and the University of Pittsburgh among others. His company specializes in sound and communication systems, specializing in systems for public facilities, houses of worship, and allied engineering. The business has completed more than 400 church installations, and 150 other installations including shopping malls, teleconferencing facilities, performing arts centers, television studios, and radio stations.

The Development of Hollow-State Live Sound Amplifiers

Richard Honeycutt reflects on the early inventions that formed the foundation of the live sound equipment we have today for our concerts, festivals, and public address systems.

Photo 1: This Vocal Air modulated airstream driver, similar to the one used by Gaumont in 1900, is capable of producing the loudest amplified sound of any speaker even today. (Photo courtesy of Jim Hunter, Manager of Engineering, Operations, and Resident Historian of the Klipsch Museum of Audio History in Hope, AK.)

By
Richard Honeycutt
(United States)

When a medicine is used for something other than its primary intended use, such a treatment is called an "off-label" application. Off-label applications are found in many other contexts as well. The telephone is a good example. Both Alexander Graham Bell's wife and his mother were profoundly deaf, so he set out to invent an electrical device to help them hear—we would now call his objective a hearing aid. The first off-label application of his efforts was the great-grandmother of the small device that most of us carry in our pockets to communicate with others remotely or to perform many other second-level off-label tasks, now called a cell phone.

The second off-label application of Bell's invention was public address (PA) systems. Before the year 1900, persons speaking at large public gatherings had to depend upon their own vocal projection in order to be heard. In that year, Leon Gaumont and the Gaumont Film Company gave the first public demonstration of electrically enhanced sound projection in Paris, France. This Chronomégaphone was a device that provided synchronized sound playback from a disc recording with the visual image from a projector. The speaker horn was driven by a modulated compressed air driver (see **Photo 1**).

The Beginnings of the Loudspeaker

On Christmas Eve 1915, Edwin Jensen and Peter Pridham, founders of Magnavox, demonstrated the electrodynamic loudspeaker that Jensen had invented by projecting concert sound to a crowd of 50,000 in Golden Gate Park in San Francisco, CA. The *San Francisco Bulletin* commented: "Great invention made of [sic] Californians solves many problems. The slender tone of a single violin heard about a mile away. The sound [sic] operatic Luisa Tetrazzini's voice reverberating throughout the stadium, and a piano solo resembling the chimes of Westminster Abbey played by Colossus of Rhodes."

In 1919, President Woodrow Wilson undertook a speaking tour to promote the establishment of the League of Nations, and one memorable speech was made to an audience estimated at 50,000 people gathered in what was then known as City Stadium in San Diego, CA. The President spoke from a glass enclosure, and his voice was projected by a Magnavox sound system. Although details on the amplifier used are scarce, it was said to be a three-stage amp using a parallel pair of output tubes rated at 500 V plate voltage. The maximum output power is uncertain. However, the Jensen speaker, which today we would call a compression driver mated to a curved horn, probably had an efficiency of about 25%, making possible intelligible sound projection over a mile, even with an input power almost certainly under 50 W.

Western Electric, having seen the potential of the loudspeaker, offered the "loudspeaking telephone," which, combined with an amplifier and a microphone, made up a rudimentary public

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Photo 2: This WE 7-A amplifier is a part of the collection of Ernie Hite. (Photo courtesy of the Charlotte ARC, http://lilienthalengineering.com/100-amplifiers-chapter-1)

address system. It was a Western Electric PA system that provided sound reinforcement for the 1921 inauguration of President Warren G. Harding. The inaugural address was transmitted by AT&T Long Lines from its originating location at the Tomb of the Unknown Soldier in Washington D.C.'s Arlington National Cemetery to Madison Square Garden in New York and to the Civic Center and the Plaza in San Francisco, CA, allowing more than 150,000 people to hear the address in real time.

Western Electric's 7-A Amplifier

Although details about the system are hard to come by, it is reasonable to assume that the amplifiers used were similar to the Western Electric 7-A amplifier (see **Photo 2**), whose schematic is shown in **Figure 1**. Although the 7-A did not enter full production until 1922 (some sources say as late as 1924), it is likely that the amp used for the Harding inauguration was at least similar in design.

The WE 7-A was a two-stage all-triode transformer-coupled amplifier using a single-triode preamp, a transformer phase splitter, and a push-pull output pair. The tubes were type 216A (see **Photo 3**). These triodes used 5 V to 6 V filament supplies, and were generally designed to operate with a 130 V plate voltage (160 V rated maximum), drawing a plate current of 6.5 mA with the control grid biased at -9 V. These conditions would produce a quiescent plate dissipation of 0.845 W.

In Class-A operation, the maximum available output power would have been about 0.28 W, but in the push-pull configuration, a pair of 216As could theoretically put out about 2.1 W. With the high efficiency of the loudspeaking telephone receiver used as a speaker, this would radiate a reasonably loud sound, albeit the fidelity would have been limited by the ~500-Hz horn cutoff.

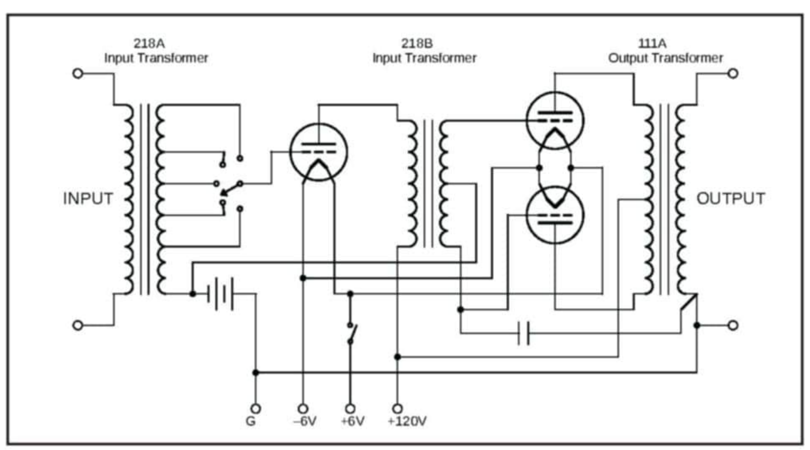

Figure 1: The WE 7-A amplifier was used both as a repeater amplifier for long telephone lines and as an amplifier for sound to be reproduced by loudspeaking telephones.

Photo 3: The WE 7-A used three 216A vacuum triodes.

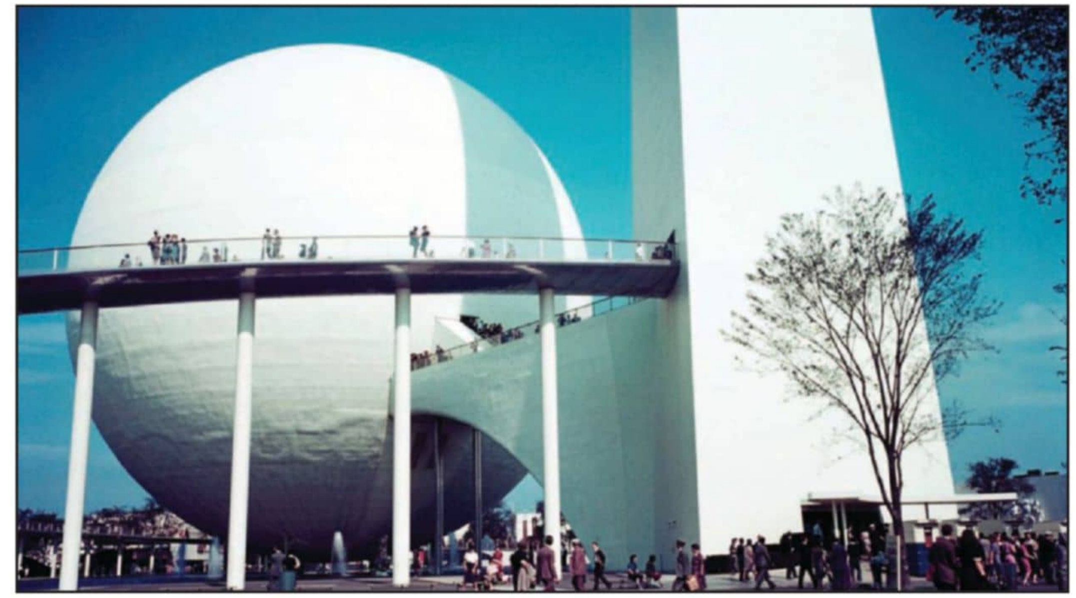

Photo 4: The Perisphere of the 1939 New York World's Fair was served by two large sound systems. (Photo courtesy Peter Campbell/ Corbis via Getty Images)

A similar Western Electric system may have been the one provided by the Chesapeake and Potomac Telephone Co. to support Calvin Coolidge's inauguration in 1925.

Other Memorable Sound Systems

The New York World's Fair in 1939 included three memorable sound systems. Two were involved with the "Perisphere," a 200'-diameter sphere (see **Photo 4**) housing a futuristic diorama of a utopian city named Democracity. The inside of the Perisphere was served by six RCA theater speakers, each driven by a pair of 50 W power amplifiers using 6L6s in a push-pull configuration. Outside the Perisphere, an enormous horn speaker was concealed beneath the pool surrounding the Perisphere. The speaker system designed by John Volkman of RCA was driven by 24 50-W amplifiers using 845 tubes running in Class A.

The third system was installed at the "Lagoon of Nations" (see **Photo 5**). This feature at the fair was a display using water, fire, smoke, light, and sound. As many as 1400 streams of water sprang into the air, illuminated by floodlights of various colors. The sound had to seem to come from the air above the center of the fountain, and the system had to cover as many as 100,000 people at an average distance of 250'. A special speaker system designed

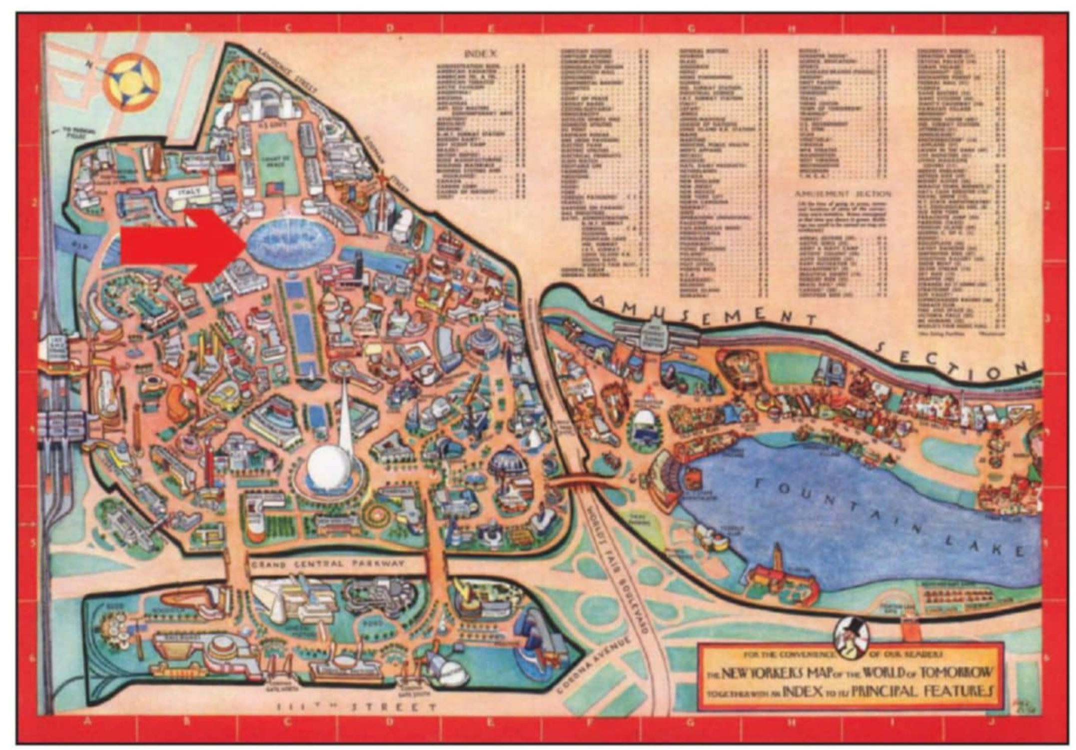

Photo 5: This layout of the 1939 World's Fair shows the Perisphere near the bottom and the Lagoon of Nations marked by a red arrow.

by Rudy Bozak of Cinaudagraph was custom-built for this application.

The speaker systems were housed in four fireproof chambers or "igloos" located among the fountains. Each igloo contained two low-frequency horns providing directional control down to 28 Hz, plus four high-frequency multicellular horns. The music was performed live in a remote studio using a stereo arrangement of Western Electric cardioid microphones.

The igloos were located so as to present a stereophonic aural image to each listener within the 360° arc surrounding the lagoon, so that the virtual locations of the orchestral instruments could be distinguished. Thus, each igloo was fed by a power amplifier driving a single low-frequency horn and two high-frequency horns. The amplifiers were made by Bludworth. The schematic of each amplifier is shown in **Figure 2**. (The amplifier schematic was provided by Jim Hunter, Manager of Engineering, Operations, and Resident Historian of the Klipsch Museum of Audio History in Hope, AK.)

The amplifier uses balanced circuitry throughout, with a voltage amplifier utilizing 6SF5 and 6C5 triodes driving balanced 2A3 drivers with a 250 V B+, which in turn drove a transformer phase splitter feeding a push-pull output pair of type 838 triodes. Each amplifier also provided a separate output for an 8" monitor speaker for the low-frequency channel, the high-frequency channel, and the full-range signal. The 838 tube was designed by GE to be used as a Class-B modulator tube having a maximum plate voltage of 1250 V, a maximum plate current of 175 mA, and a maximum plate dissipation of 100 W. The bulb was 2 5/16" in diameter and 7 7/8" long. **Photo 6** illustrates a modern use of the 838.

Many Technological Advances Created Today's Pro Sound Industry

Numerous authors have pointed out that the home stereo ("hi-fi", as we used to call it) industry as well as the pro sound industry owe their beginnings to the motion-picture industry. While this is true, it should also be mentioned that several companies

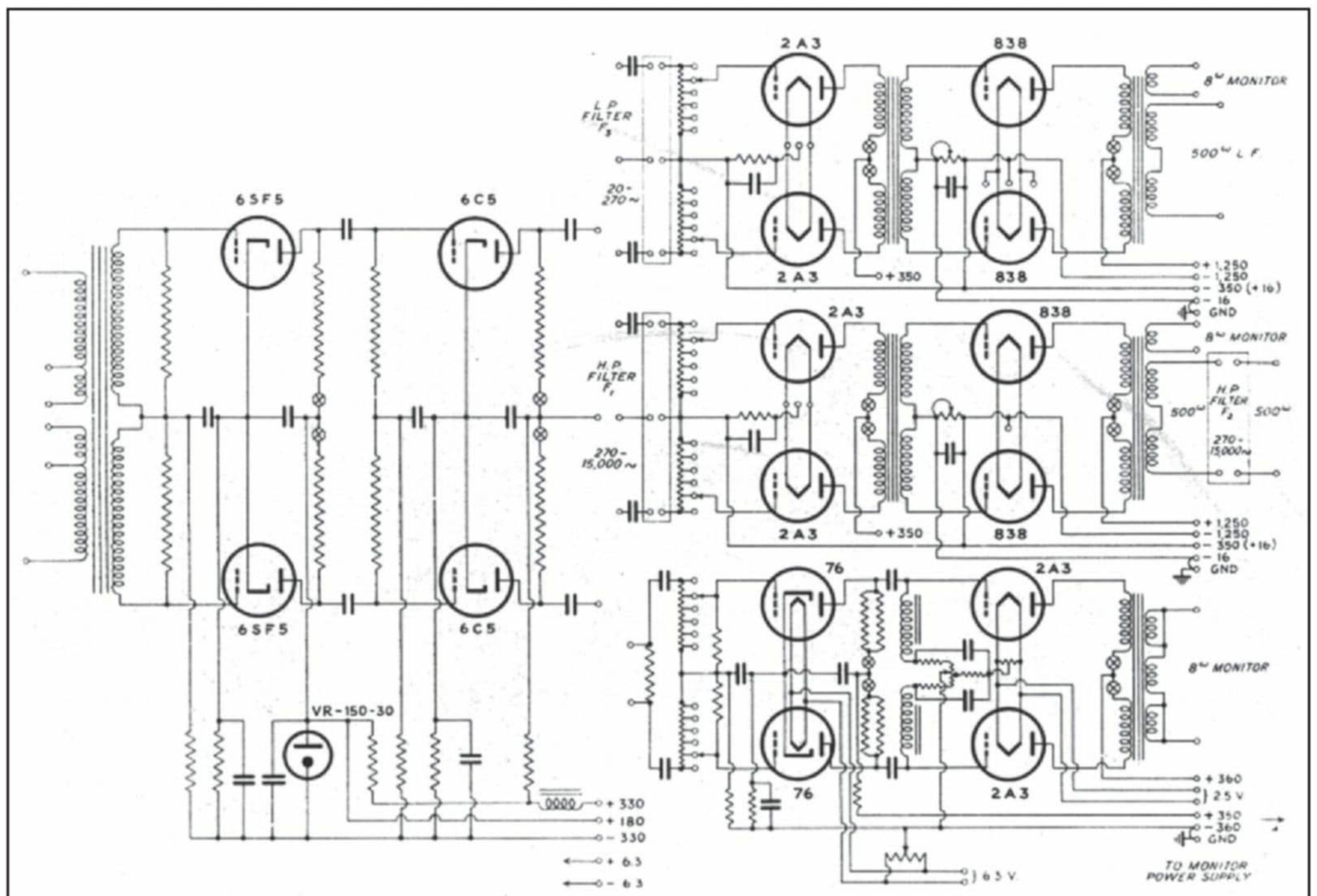

Figure 2: The Lagoon of Nations sound system used four of these amplifiers that produced an output of 500 W each.

Photo 6: These left- and right-channel single-ended audio power amplifiers illustrate the use of the 838 tube, and show its size in comparison with a 5998 driver triode and miniature dual triodes 12AT7 and 12BH7. (Photo courtesy of Michimori Hirokuni)

including the telephone industry—especially Western Electric—and the work of independent inventors (e.g., Edwin Jensen) contributed to the beginning of the pro sound industry.

As the live-sound industry grew in the years between 1900 and 1940, as in following years, its needs often pushed the limits of available electronics technology, such that custom-designed amplifiers were occasionally needed.

The Bludworth amplifier used in the Lagoon of Nations exhibit at the 1939 World's Fair and the presumably pre-production Western Electric 7-A amplifier used in Harding's inauguration are examples of this process.

Even during the Woodstock festival, the 10,000 W sound system was mostly powered by 350 W McIntosh MC3500 tube amplifiers, producing 11 kW of quiescent heat below the stage. This "side effect" of tube amps required the use of bagged ice along with multiple cooling fans to keep the system operating!

While 21st-century live sound events no longer use hollow-state power amplifiers, I hope you have enjoyed this trip in the "Way-Back Machine" to review some major milestones of the past!

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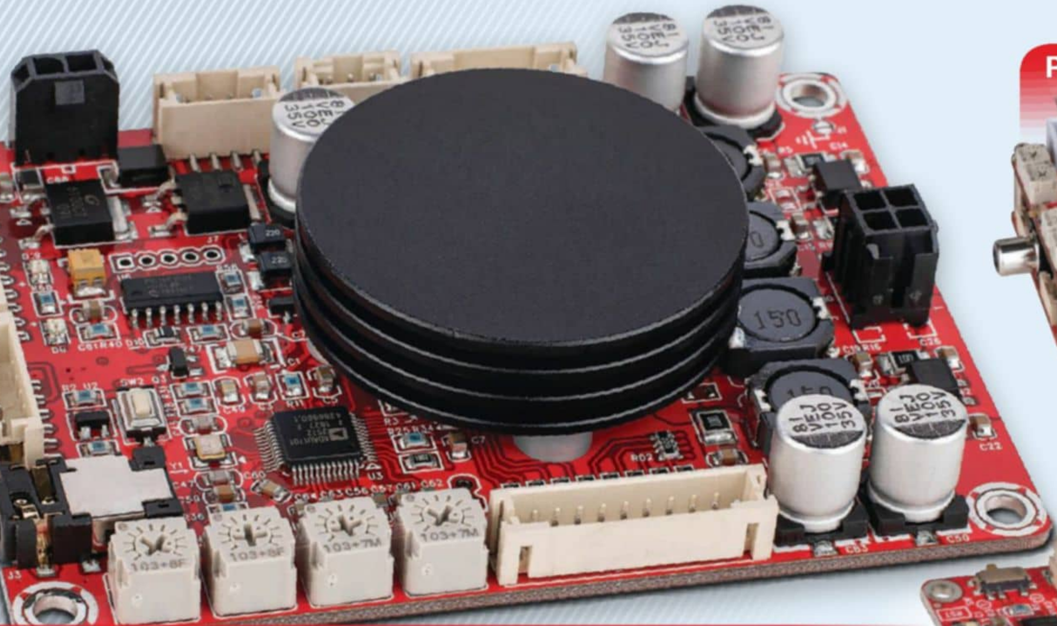

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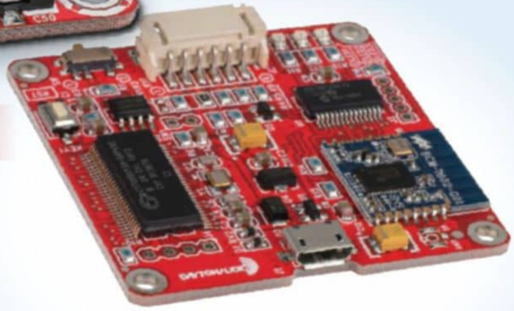

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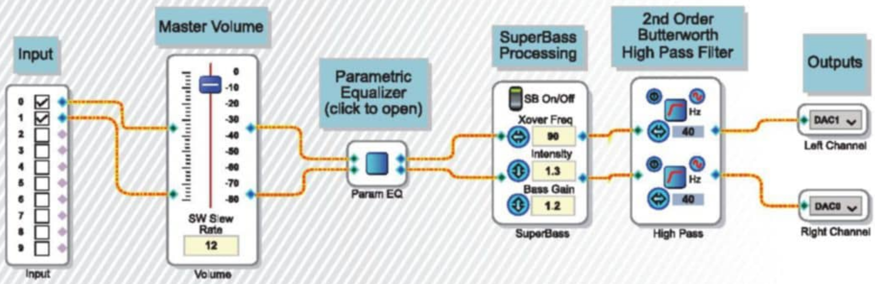
2 channel SigmaStudio project for the DSPB-250 (with Superbass processing).



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