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Why Smart Power Audio Amplifiers Are Becoming Important to Smartphones

By Eric Eklund (Cirrus Logic)

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Class D for Denmark

This edition of *audioXpress* focuses on audio amplifier technology and includes an article from Ward Maas in our Amplifier Series—this time dedicated to an Italian company, Powersoft. Anyone familiar with the professional audio market will immediately recognize Powersoft as one of the leading developers and manufacturers of power amplifiers, now also expanding into other essentially related areas, such as new types of transducers and network connected systems. The article focuses on Powersoft's expanding range of OEM solutions, which with the trend toward fully integrated and active loudspeaker systems is an increasingly important business.

Previously, the Amplifier Series has profiled several other OEM Class-D solutions and provided the history of those companies, including ICEpower, Pascal, and Merus Audio (acquired by Infineon), which have a common denominator: they are all from Denmark. Yes, there are other leading companies from other neighboring countries exploring Class-D amplification technology, such as Hypex and Axign—also profiled in *audioXpress*' Amplifier Series, to which we could also assign another "D" for Dutch. We could even add "D" for Deutschland where Infineon Technologies originates. But enough with the "D" country analogies, the truth is, something really interesting is happening in the Kingdom of Denmark.

As *audioXpress* reported online, a new all-star Danish company was recently born, resulting from the association of a leading team of pioneer researchers and entrepreneurs in the audio industry, with a common denominator in the origins of the Class-D revolution, and much more. The company is called Purifi Audio, and it has already announced that it is on a mission to revolutionize both loudspeaker drivers and audio power amplification. Purifi Audio's work is referenced in the Audio Amplifier Market Update included in this edition.

The company was co-founded by Bruno Putzeys, the inventor of UcD (Philips) and Ncore (Hypex), together with Lars Risbo, who pioneered direct switching PCM-PWM audio amplifiers as founder of Toccata Technology, later acquired by Texas Instruments (TI). The third co-founder is Peter Lyngdorf, a name that is almost synonymous with the Danish audio industry, founder and owner of DALI, Lyngdorf Audio and Steinway Lyngdorf, among other companies.

With the three co-founders, we find as co-owners the names of Carsten Tinggaard, the founder and owner of PointSource Acoustics (2009), previously involved with Tymphany Denmark, Peerless, Vifa and Scan-Speak brands; Claus Neesgaard, another Toccata/TI R&D mind; Søren Poulsen, former lead of the audio power IC division at TI; Kim Madsen, also a lead engineer from TI in audio amplifications; and Morten Halvorsen, a Senior Acoustic Engineer at PointSource Acoustics, who did pioneering work on force factor modulation, new loudspeaker measurement equipment, and related simulation models. Together, this team holds more than 50 key patents in Class-D amplification, loudspeaker design and related measurement technologies.

audioXpress recently interviewed the Purifi Audio team, and we soon intend to explore even more of their work, which can be considered as "the next step" in the evolution of both Class-D amplification and loudspeaker design. At the origins of Purifi is the remarkable work developed at PointSource Acoustics since 2009, establishing the Roskilde company as the Danish industry's loudspeaker reference lab, which is now also the headquarters for Purifi Audio.

And this doesn't happen by chance.

The team brings with it all the lessons learned from the team members' previous work at leading companies in the field. But they also benefit from the leading ecosystem of the Danish audio industry, which leads the field since, in 1915, Peter Jensen (1886-1961), invented electronically amplified sound through his discovery, the loudspeaker. For a better understanding of Denmark's contribution toward the audio industry in general, I highly recommend "The Danish Loudspeaker 100 Year Anniversary," book (www.audioxpress.com/article/the-danish-loudspeaker-100-year-anniversary-book), which also acknowledges all the work of the Danish Technical University (DTU), and other engineering schools, at the origin of many of Denmark's companies involved in microphones, loudspeakers, audio amplifiers, audio test and measurement, and even hearing aids. Today, Denmark leads the audio world.

J. Martins
Editor-in-Chief

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COLUMNISTS

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

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

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

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

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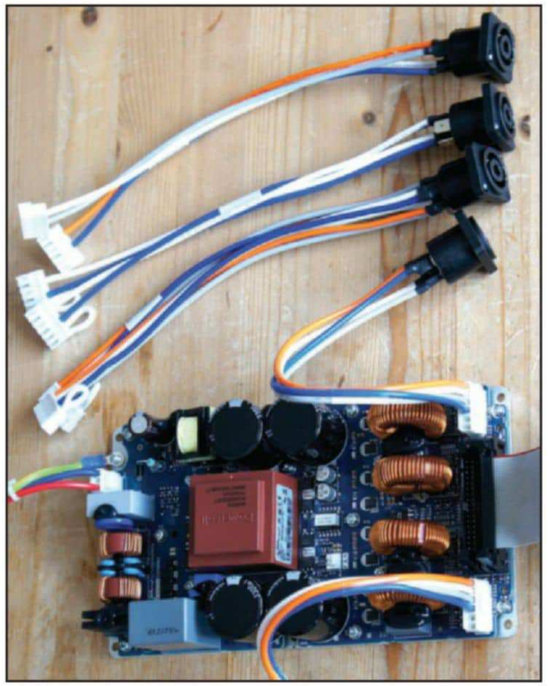

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Flexible Power for Active Systems

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The transportation of audio over IP is profoundly flexible, yet the ability to retain controllability over all these IP connected devices remains a frontier for discussion. This article considers approaches to matching the flexibility of AoIP network structure with new technologies and standards for control and monitoring user interfaces.

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Market Update—Audio Amplifiers

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By J. Martins

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By Eric Eklund

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Acoustic Design of Music Rehearsal Rooms

Whether you're considering a rehearsal room for a choir, band, orchestra, or garage band, certain principles apply to the design of a successful space.

Photo 1: A small music rehearsal room (Photo courtesy of RPG, Inc.)

By
Richard Honeycutt
(United States)

In years past, design principles for music rehearsal rooms were often not considered, with the result that musicians, singers, and conductors were not able to hear the ensemble well, or perhaps the rehearsals were uncomfortably loud—possibly dangerously so. Studies of hearing loss among musicians have concluded that not only those playing amplified music, but also musicians in bands and orchestras may suffer occupational hearing loss. In instrumental groups, especially percussionists and those sitting in front of brass sections—particularly trumpets and trombones—may be at risk.

There are six aspects of rehearsal-room acoustics that require attention in order to achieve truly good acoustics:

- There must be enough acoustical absorption that the sound levels do not become too high during rehearsals.
- Acoustical absorption and diffusion must be properly distributed to prevent flutter echoes.
- The sound reflections from the walls and ceiling must be great enough that musicians can hear each other well, and must be sufficiently diffuse that each singer or musician can hear all others in the ensemble.
- The floor must not be too acoustically absorptive, so that the room is not too dead.
- There must be enough floor area per person so that the musicians or singers are not bunched together.
- The Reverberation Time (RT) should be optimum for the type of ensemble. This requirement, combined with the floor-area requirement, determines a minimum volume (ft^3 or m^3) per person.

Photo 1 shows a small rehearsal room in which these aspects have been considered. We will examine each of these aspects.

Sound Level Control

Acoustical absorption is the process by which sound energy is converted into heat. Usually, this occurs by friction, as when vibrating air molecules in a porous material transfer energy to fibers or cell walls in the material. Acoustical absorption is measured in sabins—a metric sabin corresponds to one square meter of perfect acoustical absorber, and an Imperial sabin, to one square foot of perfect absorption. In this article, we will consider "sabin" to mean "Imperial sabin." Every material provides a certain amount of acoustical absorption: wall, floor, ceiling materials, furniture, and people in the room all contribute to the total number of sabins in the room.

The absorption provided by a certain piece of material depends upon the area of the material and the acoustical absorption coefficient α (alpha). Alpha has long been defined as the fraction of incident sound energy absorbed by a unit area of a material.

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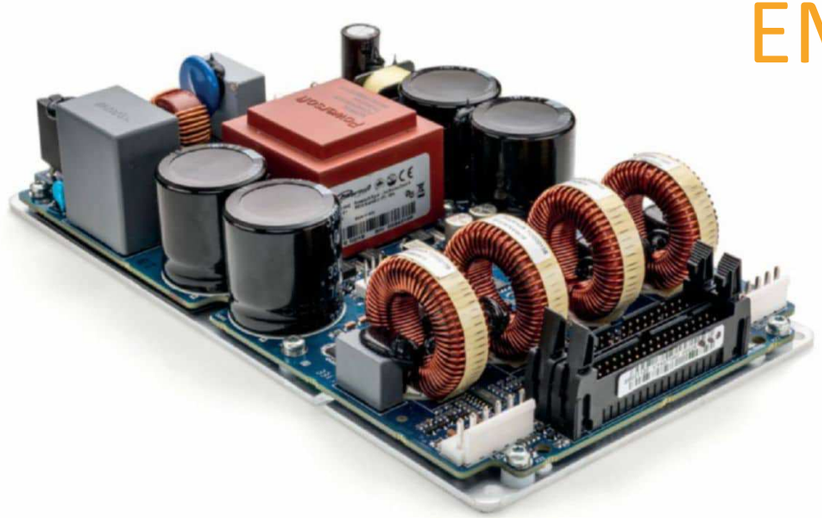

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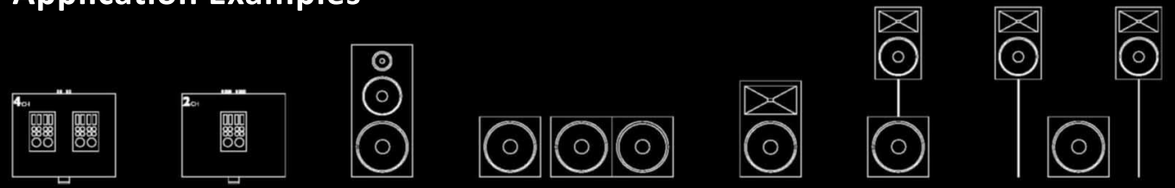
Application Examples



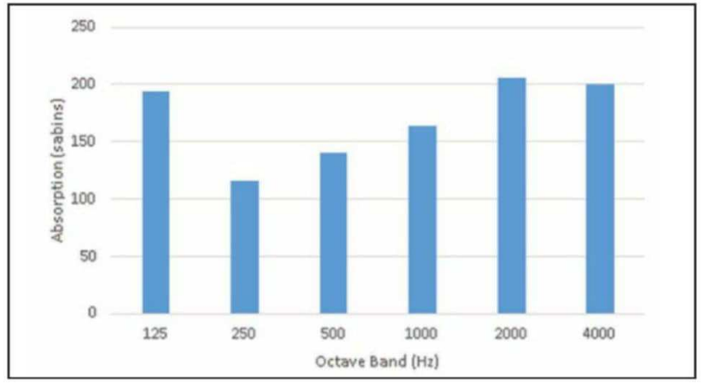


Figure 1: Total acoustical absorption in a small room

While this definition is approximately correct, it does lead to certain mathematical errors, as discussed in previous Sound Control articles. These have to do with the rigorous application of W. C. Sabine's statistical acoustics. However, considering α as a measure of the fraction of incident sound that is absorbed, specified in Imperial sabins per square foot or metric sabins per square meter is satisfactory for a qualitative understanding. This having been said, I will mention that it is possible to measure α values ranging from near zero to about 1.4, depending upon the size and shape of the material sample being tested. A truly accurate prediction of sound level in a rehearsal room would require wave analysis, such as Boundary Element Analysis (BEA) or Finite Element Analysis (FEA). But statistical methods such as Sabine's provide good engineering estimates, and are commonly accepted.

Based upon statistical acoustics, the amount of acoustical absorption needed in a rehearsal room depends upon the sound sources and the total amount of acoustical energy released in the room. A singer produces less sound energy than an instrumentalist, and an acoustic instrumentalist typically produces less than an amplified instrument. So it turns out that to keep sound levels safe and comfortable, a choir rehearsal room needs a minimum of 16 sabins per person; a band or orchestra, a minimum of 25 (preferably 30 or more); and amplified instruments, 35 or more, depending upon the amplifier settings. These amounts of absorption must be effective in the 500 Hz, 1 kHz, and 2 kHz octave bands since acoustical absorption of a material varies with frequency.

Figure 1 shows the total acoustical absorption in octave bands in a small rehearsal room 20′ wide × 20′ long × 18′ high. The walls and ceiling are concrete; the floor is vinyl tile, and the occupancy is 10 middle school students. No acoustical treatment of the walls or ceiling has been applied. As you can see from the graph, total absorption in the 500-Hz band is 140 sabins, which amounts to 14 sabins/person. For a small chorus or band of any kind, more absorption is needed, at least in the 500-Hz band (pitches near middle "C"). How much of and what kind of absorption will be affected by other factors, as we will see.

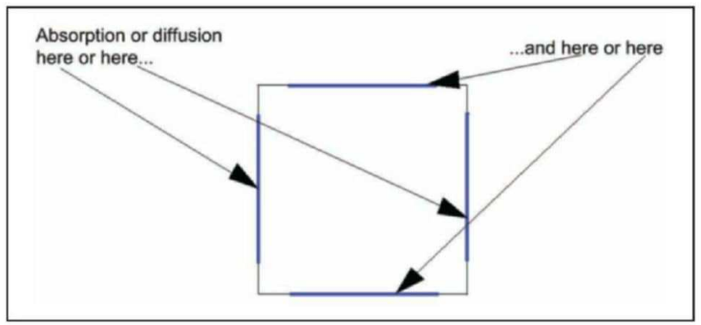


Figure 2: Absorption or diffusion locations to prevent flutter echo (plan view)

Distribution of Acoustical Absorption and Diffusion

The minimum time delay required for a person to perceive a reflection ("the Haas limit") is stated variously as somewhere between 30 ms and 80 ms. This corresponds to sound traveling between 34′ and 94′. The figure 50 ms (56.75′) is often used as a convenient approximation. A sound source located near one wall of our 20′ × 20′ × 18′ room will travel a maximum of 40′ (~35 ms) before returning to the source.

With completely hard walls, the sound will re-reflect from side to side many times before being completely absorbed. These repeated reflections are called "flutter echoes," and they can impair the sound quality in a number of ways—thus, they should be prevented if possible.

One way of doing so involves avoiding acoustically reflective surfaces opposite each other in the height range of the source. For a singer, this height range covers from about 3′ (singer seated) to 7′ (very tall singer standing). Some sort of absorptive material should be placed on one wall of each opposing pair, especially on the area near the center of the walls.

Another way to avoid flutter echo is to use acoustically diffusing materials in those same locations (see **Figure 2**). The absorption or diffusion should extend from about 3′ to 7′ above the finished floor. It is not necessary that the entire area illustrated in Figure 2 be covered with absorption or

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Photo 2: This barrel diffuser redirects sound that strikes it. (Photo courtesy of Acoustic Surfaces, Inc.)

diffusion; the material can be applied in blocks with gaps of no more than a foot width between blocks. Refer back to Photo 1 to see examples of absorptive (curtains), diffusive (wood elements), and reflective (walls at corners) materials combined on the walls of a rehearsal room. And as we will discuss later, some absorption will need to be placed on the ceiling.

Proper Reflection Strength

Reflected sound that arrives at a listener's ear within the Hass limit after the initial sound is perceived as part of the same acoustic event as the initial sound, but the reflection causes an increase in sound level. Ideally, the members of a musical ensemble should hear the reflected sound at a high enough level to enable them to play in tune and in time. The reflection strength is determined by the average α of the surface (wall, floor, or ceiling) from which the sound reflected, and the distance from the sound source (singers or musicians) to that surface and back to the person's ears. It is preferable that the wall reflection strength and the ceiling reflection strength be approximately equal to promote the singers' and musicians' feeling that they are acoustically localized within the group, not under it or at some distance away horizontally. Equalizing wall and ceiling reflection strengths also helps participants to "hear across the group"—to hear the entire ensemble rather than only certain members.

Photo 3: These pyramidal diffusers redirect sound in two dimensions. (Photo courtesy of RPG, Inc.)

In Figure 2, our 20′ × 20′ × 18′ room contains two blocks of acoustical absorption, each having a maximum area of 14′ wide × 4′ high: a total area of 56 ft^2. If the material used is 1″ thick fiberglass panels, the average α in the 500 Hz, 1 kHz, and 2 kHz bands is 0.94. If we ignore the acoustical absorption of the concrete wall, the average α of each treated wall would then be the total absorption of the wall divided by the wall area: α avg = 0.94 × 56/(20 × 18) = 0.146. We will need to add acoustical absorption to the ceiling in order to prevent it from being totally reflective (to bring its average α in line with that of the walls, taking distances into consideration). Often the proper average α for the ceiling is attained by using a suspended ceiling made of a mix of absorptive (fiberglass, mineral wool, or mineral fiber) and reflective tiles (gypsum), along with diffusive elements (see **Photos 2–4**), which are likely to be acoustically reflective. Further information on acoustically absorptive and diffusive materials can be found in the Sound Control articles that were published in the December 2013 and February 2014 issues of *audioXpress*.

Proper ceiling heights to provide balanced wall and ceiling reflections are usually 16′ to 20′ for choral rooms, and 18′ to 22′ for instrumental rooms. When retrofitting existing rooms having higher ceilings, one can install a lower suspended ceiling or a suspended absorptive/reflective/diffusive plane ("cloud") at the proper height.

Floor Coverings

Rehearsal room floors should not be carpeted, since carpet absorbs primarily high frequencies, giving the room a blanketed sound. Vinyl composition tile or wood is preferable. The acoustical absorption and scattering provided by furniture and people generally prevent floor reflections from having to be taken into account. The exception is that if choral risers are used, they may be carpeted, since most sound striking the carpet is ultimately absorbed by people.

Floor Area per Person

Aside from personal comfort, there is an acoustical reason for providing sufficient floor area per person in a rehearsal room. If singers or musicians are "packed like sardines", each person will hear mainly the sound of the nearest neighbors, rather than being able to hear across the group. This makes it difficult for members to sing in balance. The preferable area per person is about 20 ft^2 for choral rehearsal, and 30 ft^2 for instrumentalists.

Reverberation

At first thought, it would seem that the ideal RT for a rehearsal room would be the same as that for the room in which the ensemble typically performs. There are several reasons that this is not so. First, few groups perform in only one venue, so matching the RTs of multiple performance venues becomes impossible. Second, the ideal RT for a particular type of music depends upon the room size, and the ear/brain system automatically calculates approximate room size based upon the delay experienced by early sound reflections. So a (typically) smaller rehearsal room with the RT of a larger performance hall would not sound right. Third, the greater RT of the performance venue can make it difficult for the conductor to hear small details in the music during rehearsals. For example, our 20′ × 20′ × 18′ room had a volume of 7200 ft^3, and would ideally have an RT of about 0.95 seconds if used as a rehearsal room. It turns out that a good rehearsal room should include a minimum of 300 ft^3 of volume for each singer, or 400 ft^3 for each instrumentalist.

Pulling It All Together

Since sound level, reflection strength, and RT all depend upon room dimensions (floor area and enclosed volume) and acoustical absorption, the six aspects of designing good rehearsal rooms are interconnected. Although we will not get fully into the math and physics of acoustical design, we can see how our 20′ × 20′ × 18′ room will serve. As mentioned, the floor area is 400 ft^2, allowing adequate space for 20 singers or 13 instrumentalists, based upon the floor area requirements presented earlier. The volume is 7200 ft^3, which is enough space for 24 singers or 18 instrumentalists. Thus in this room, maximum comfortable occupancy is limited by the floor area.

Photo 4: This product combines diffusion and absorption. (Photo courtesy of Jocavi, Inc.)

The Sabine formula for RT is:

RT=0.49 × V/(total absorption in sabins).

Thus, we can solve for total absorption based upon the target RT of 0.95 seconds. The answer is 371.4 sabins. Sound-level control, as discussed earlier, requires a minimum of 16 sabins per person; a band or orchestra, a minimum of 25 sabins (preferably 30 or more); and amplified instruments, 35 sabins or more, depending upon the amplifier settings. For the maximum-size choir, this comes to 320 sabins for 20 singers or 325 sabins for 13 instrumentalists, or more for amplified instruments. Note that to obtain the preferred RT, a bit more absorption is required than would be needed just for sound-level control. The techniques of controlling reflection strength and wall/ceiling reflection balance are beyond the scope of this article.

But now you can understand the multiple interacting factors involved in designing a successful music rehearsal room!

Oh, yes, if you're concerned about room modes in this room, you may be interested in knowing that the Schroeder frequency is 108 Hz, so unless the music contains significant fundamentals below A2, modes will not be a problem.

Editor's Note: All audioXpress articles from 2001 to present can be found on the aX Cache, a USB drive available from www.cc-webshop.com.

Resources

R. Honeycutt, "Acoustical Diffusion and Scattering," *audioXpress*, February, 2014.

R. Honeycutt, "The Oldest Tool in the Modern Acoustician's Toolbox," *audioXpress*, December 2013.

Power for All Audio Needs

By
J. Martins
(Editor-in-Chief)

The Lyngdorf MXA-8400 is an eight-channel power amplifier, and the most powerful product from the company, with a peak output power of no less than 3.2 kW. It is also one of the first products to integrate the new Purifi Audio self-oscillating Class-D technology.

For this market update on audio amplifiers we have selected a few highlights from multiple segments, in an effort to illustrate the state of the industry in terms of audio amplification. Following our May 2019 issue almost entirely dedicated to tube amplification, we will focus mainly on solid state and several cutting-edge approaches in all amplification segments, including OEM modules, amp plate solutions, and the evolution of Class-D topologies—an area that is seeing increasing activity.

A lot is happening in audio amplifier technology, even if it is not always reflected in commercially available products. With many silicon companies focusing on high-volume portable and mobile platforms, and product categories (e.g., smart speakers, where the most exciting innovations are taking place), the more conventional applications segments, from high-end home audio to multichannel home theater, studio to sound reinforcement, and professional audio installation have been relatively quiet in the last 12 months. But there is clearly a crossover and there are some exciting developments on this front.

Incremental High End

In the high-end segment, we continue to see some leading amplification brands introducing "improved recipes" of their company's own topologies, focusing on increasing the specs of some very high quality products, even if only slightly. While the leading consumer electronic brands focus on features such as integrated DACs and adding support for immersive formats or wireless protocols in integrated solutions, the fundamental amplification stages have not changed so much. One exception is that the audio industry benefits from much more efficient and sophisticated chips and amplification platforms that can be found in most mainstream products. And Class-D technology is evolving fast and getting better every day. Of course, the leading audio companies continue to perfect their Class-A amplification designs and we have seen a few interesting recent announcements.

Celebrating its 30th anniversary in 2019, British manufacturer Chord Electronics has launched Ultima, a new state-of-the-art flagship amplifier featuring the latest circuit topology of John Franks'—the company's owner and Chief Designer. The first all-new design in three decades from Chord, the Ultima mono amplifier and the matching preamp is now available as a set (pre and 2× mono power amps) for £90,000, or approximately $116,000 USD. The interesting part of Ultima's technology is that it is based on a technical paper by Dr. Malcolm J. Hawksford (emeritus professor at Essex University), which was taken and refined by Bob Cordell. John Franks, then took the theory and developed it to a much higher level; a progression of Cordell's advanced dual-feed-forward error-correction topology incorporating Franks' own advanced concepts in ultra-high-frequency power supplies, which have been a cornerstone of the company's proprietary-technology amplifiers since its inception in 1989.

The Ultima mono amplifier unit has an incredible 64 proprietary MOSFET power devices that demand the most advanced and refined drive circuitry, suitably devoid of sound-degrading components. The drive circuitry features an ultra-low-distortion, high-voltage amplifier operating at very high speed: 300 V per microsecond technologies with advanced feed-forward and feedback compensation techniques. It also benefits from Chord Electronics' TO3-style dual die lateral matched MOSFETs, which are continuously monitored with output stage error-correction circuitry and feature fully balanced input circuitry with DC-tracking servos and full temperature compensation.

Just recently, Chord introduced more affordable models with the same technology, the new 750 W Ultima 2 (£18,360 each), and the smaller 480 W Ultima 3 (£11,000 each).

From Bryston, another leading name in high-end amplification, we have not seen much activity in new amp stage designs. But the Canadian company did announce a collaboration with StormAudio, the French premium audio hardware manufacturer of multichannel amplifiers for commercial and home cinema, which was converted into an independent business two years ago.

StormAudio developed audio electronics that support all existing sound multichannel and immersive formats, such as Auro-3D, Dolby Atmos and DTS:X. Both companies are now leveraging their respective design and manufacturing expertise to create new products. And that's exciting.

As a byproduct of the cooperative effort, Bryston has already unveiled its SP4 surround processor based on the acclaimed StormAudio platform, and StormAudio introduced two-channel and four-channel amplifiers based on Bryston's own Cubed Series Technology.

Bryston has demonstrated past success partnering with respected brands such as Harman, PMC, Axiom, and others, and the cooperation with StormAudio promises to significantly impact the home theater and custom integration markets.

Jumping to another stalwart of high-end audio, we have seen the introduction of Dan D'Agostino's Relentless Monoblock amplifier. The Relentless Monoblock amplifier, the latest manifestation of D'Agostino's passion and desire to always go further, is able to drive any speaker to its best possible performance, no matter how demanding the load it may present.

Chord Electronics Ultima state-of-the-art flagship amplifier

With its 5.5 kW power supply feeding roughly 100 output devices, the Relentless Monoblock easily delivers 1,500 W into 8 Ω—and when connected to a 220-V outlet, it doubles its output to 3,000 W into 4 Ω and 6,000 W into 2 Ω, all while maintaining the same musical composure it offers at a mere 1 W.

The Relentless Monoblock employs an all-new, fully complementary topology based around a precision input stage that maintains an essentially perfect balance between the signal's positive and negative components. Any distortion artifacts that appear on one half of the signal will also appear on the other half, and thus cancel each other. Gain circuits operate in the current domain, assuring the Relentless Monoblock's performance does not fluctuate no matter what the demands. Global negative feedback is not employed, nor is it required as the amplifier's open-loop linearity eliminates the need for this performance-robbing technique.

The core of the power supply is a transformer that uses multiple sections running in concert, all potted and encased in a metal shield. This design reduces transformer noise to near zero and protects the audio circuitry from the transformer's magnetic field. The transformer feeds a custom-designed rectifier circuit and a 600,000-µF/100-V capacitor

The Bryston SP4 Surround Processor shares technology from StormAudio.

bank. A special holder mechanically grounds each capacitor so it won't vibrate. Thermal, short circuit, and DC protection are included in a non-invasive but comprehensive defense against any external system faults.

The entire chassis of the Relentless Monoblock is machined from solid aluminum, and on the front panel there's a new trademark power meter that measures more than 8" in diameter. The meter's range and response automatically adjust to the level of the incoming signal, making it easy to read no matter how quiet or loud the musical passage is.

In keeping with the tradition begun with the Dan D'Agostino Audio Systems Momentum amplifiers, the Relentless Monoblock offers a single XLR balanced input.

Professionally Focused

Looking at the other extreme of the market, one of the most interesting trends in the professional audio segment is the evolution toward networked solutions, a trend that is also going hand in hand with a strong market consolidation—with Powersoft and QSC leading the pack. At the same time, targeting less sophisticated, run-of-the-mill professional audio applications, there was a strong fragmentation with many small companies—US-based, European, and Asian—spreading fairly-standard Class-AB and Class-D audio amplifiers, and grabbing market share from the former market leaders—Music Group (that acquired Lab Gruppen) and Harman Professional.

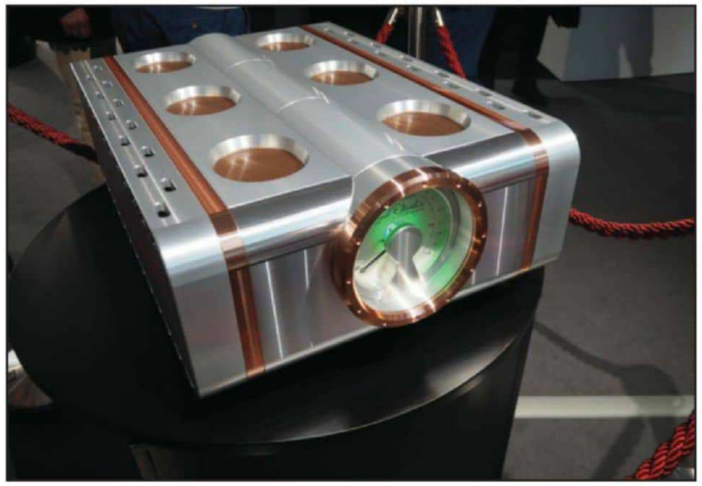
According to one of the most complete market reports dedicated to this segment, published in September 2018 by Futuresource Consulting, the professional audio power amplifier market continues to grow. Its global market value in 2017 was roughly $650 million USD, covering rental, portable sound, installed leisure, installed commercial and cinema sound applications, with the two install categories currently being the fastest growing.

Futuresource Consulting predicts that professional audio amplifiers will continue to grow at a compound rate of more than 5% a year until 2022. The report confirms that the market is extremely competitive and many speaker manufacturers are increasingly building active speakers with built-in amplifiers, while there's an influx of low-cost Asian brands. In the installation segments, the target audience is highly price-sensitive, and companies (e.g., Harman's Crown, d&b audiotechnik, and L-Acoustics) are trying to influence amplifier purchases by aligning them closer to loudspeaker platforms and systems, asserting that their speaker systems cannot be used with third-party brand amplifiers. In the touring and rental markets, this type of strategy has been particularly successful for d&b audiotechnik and L-Acoustics, and a fundamental part of the philosophy from rising challenger Bose Professional.

Also, as connectivity continues to expand as a key part of any professional audio integrated solution, Dante connectivity is increasingly used as a key feature by many amplifier brands, while d&b audiotechnik, L-Acoustics, and Meyer Sound continue to promote their own proprietary AVB-based systems, which might eventually become interoperable if the consortium-lead Milan protocol becomes a reality for those companies. But even QSC is increasingly cooperating with Audinate on Dante-compatible solutions, while Powersoft is now basically expanding Dante to all its latest range of advanced amplifiers.

In the last few months, Powersoft announced the release of its new T Series line, an ultra-reliable and efficient amplifier platform designed with the rental market in mind, with four models now available. The new platform, builds upon the credits of the flagship X Series, but it effectively combines unparalleled flexibility for small to mid-size reinforcement applications and benefits from an intense R&D effort.

The new advanced amplifier designs combine the vast experience of Powersoft's production department with an expanded R&D team that now includes the highly experienced Klas Dalbjorn, a former Lab Gruppen and Lake product research manager, appointed Product Manager in January 2017. With renewed energy, Powersoft is building a new generation of amplifiers with next-generation processing and networking capabilities, at very competitive prices. All T Series models—available in two- and four-channel iterations with 3,000 W



The Dan D'Agostino Relentless Monoblock amplifier, the most powerful high-end design from the industry pioneer

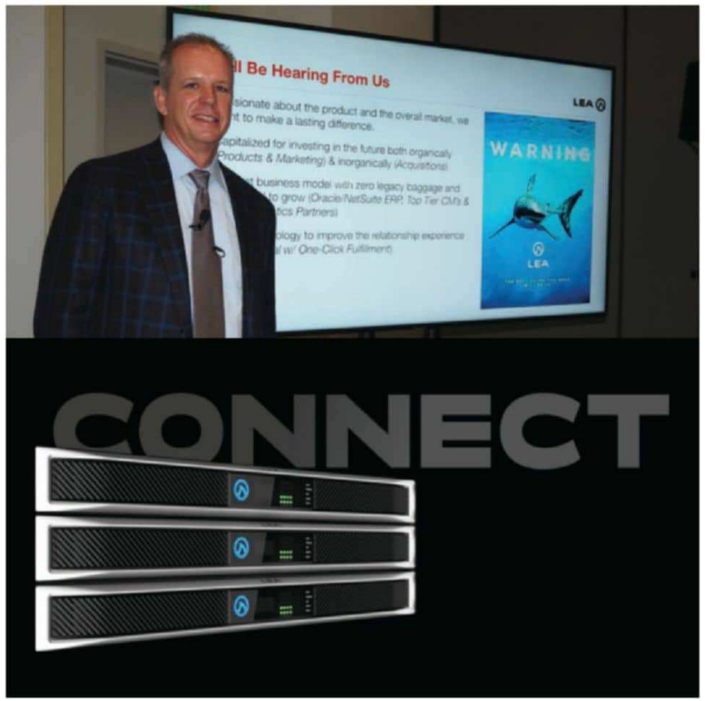
and 6,000 W in a single rack unit—can deliver the same high peak voltage and can drive single 8 Ω/16 Ω cabinets at full SPL. The high voltage can also be used for power sharing between channels for applications such as bi-amped loudspeakers or subwoofers with passive tops.

Another more recent offering from Powersoft is the new X4L, a four-channel amplifier specifically designed for peak voltages and to feed the most power hungry PA arrays and subwoofers. The X4L was designed to deliver up to 300 V (peak) to the loudspeakers, and with the ability to deliver the same or more power per loudspeaker while driving more cabinets in parallel when compared to other amplifiers, which will save users both money and truck space on portable applications. And the X4L amplifier also has versatility on its side, with five output connectors to allow easy connection without adapters for many of the expected configurations, such as channels 1 and 2 driving a cardioid sub solution, while channels 3 and 4 drive some two-way front fill in parallel.

From QSC, the Costa Mesa, CA-based leading amplification company, we have also seen the announcement of new network power amplifiers for its expanding Q-SYS networked ecosystem. A pioneer of the networked-distributed amplification concept, QSC introduced its CX-Q Series audio amplifiers with four- and eight-channel models that combine the most advanced QSC amplification technology with native network transport, control and monitoring capabilities and cross-platform Dante-networked solutions, developed in cooperation with Audinate.

As a native peripheral of the Q-SYS Ecosystem, the new CX-Q Series network amplifiers can take advantage of network routing capabilities as well as advanced processing, control, monitoring and loudspeaker output load. These amplifiers also assist Q-SYS in deploying signature QSC technologies like Intrinsic Correction, which maximizes QSC loudspeaker performance and protection.

The CX-Q Series network amplifiers feature a highly efficient, Class-D hybrid powertrain design built upon the dependable PL380 PowerLight amplifier platform, combined with two of QSC's latest amplifier innovations—FlexAmp and FAST (Flexible Summing Amplifier Technology) that combine to offer power distribution options in a single amplifier—essential in a networked audio amplifier system. FlexAmp allowing for asymmetric output channel loading, and FAST allowing combined channels to deliver either higher voltage loads (up to 200 V_{RMS} output) or higher current loads (up to 35 A). Together, these technologies help expedite installation while decreasing overall system cost by reducing wasted power and channels.



Blake Augsburger is back with a new amplifier company, LEA Professional, soon to be launching the new networked ConnectSeries amplifiers.

New Amplifiers from LEA Professional

The market leaders in this field will soon be challenged by a new company launched earlier in 2019. Founded on the premise that amplifier technology has headroom for innovation in terms of performance, networking and user interface, LEA Professional is led by Blake Augsburger, former president of Crown International and President of Harman's Professional Division for 15 years, together with a team of proven executives, engineers, and sales/marketing experts from the pro audio community.

LEA Professional is now about to launch its first products in the market but already combines a vast expertise in the pro audio amplification segment, apart from having some very clear ideas about the market opportunities at hand.

During the company's presentation at NAMM 2019, Augsburger described the strategic vision for the company, based on his view that a stratification has occurred in the market. "For amplifiers there is a complex high-end, and a commoditized low-end. At the high-end you see expensive, highly sophisticated, networked professional audio technology and at the low-end, inexpensive, undifferentiated prosumer-grade audio technology. We see an opportunity in the middle: to provide channel partners and integrators with high-performing, cloud configurable and yet affordable and easy-to-use audio technology that rocks."

Based in Northern Indiana, a traditional hub of amplifier and AV innovation and former home base for Harman's Crown, LEA is "a startup with deep pedigree," as Augsburger describes it. The engineering, sales/marketing, and executive teams feature most of Crown's experts in their fields, all with considerable accomplishments in terms of product innovation and business transformation and growth.

The first physical manifestations of Augsburger's strategy is a series of professional grade two-channel and four-channel amplifiers for small to medium scale installations. Featuring DSP, wired or Wi-Fi connectivity, webpage configuration and control, the LEA Connect Series are described as "the very first 'IoT enabled' professional-grade amplifiers and will provide end-users with best in class sonic performance, ease of use and bulletproof reliability." The LEA Connect Series, which has been detailed on the company's website, will offer a six-year warranty with advanced replacement to provide the assurance that professionals require from a new brand. The Class-D amplifiers are being partially manufactured in Costa Rica and assembled in the US, and will be available in Network-agnostic models and Dante-connected versions, all featuring a built-in web-browser, direct LoZ and HiZ (70 V or 100 V) selectable per channel, a proprietary Smart Power Bridge topology, 96 kHz Analog Devices DSP and universal Switch Mode Power Supply with PFC.

As Augsburger stated, "We want to be known for having the very best performing, best connected and best looking amplifiers in the market but we're also determined to win on price. We can do that because our capabilities are of a twenty-year-old company but our overhead is that of a startup!"

Resources

Bryston | www.bryston.com
Chord Electronics | https://chordelectronics.co.uk
Dan D'Dagostino | www.dandagostino.com
LEA Professional | https://leaprofessional.com
Powersoft | www.powersoft-audio.com/en
QSC | www.qsc.com
StormAudio | www.stormaudio.com/en

Hypex Growth

Founded in 1996 by Jan-Peter van Amerongen as a supplier of plate amplifiers for live sound loudspeakers, Hypex Electronics quickly became an established supplier of amplifier platforms, to OEM hi-fi manufacturers, but also reaching out to the studio products and demanding professional audio segments in residential and commercial installation. Since Hypex introduced its own range of class leading NCore products, the company expanded its catalog with both general-purpose amplifier modules, used now by many big name brands, from DIY audio to extreme high-end applications.

Hypex's recent focus in general-purpose power amplifier/supply module development, allowed the company to extensively expand its portfolio. Since 2017, the company has updated its entire range of OEM DSP modules and announced a completely new series of Fusion Amp (FA) plate amplifiers.

The FA plate amplifier range includes eight different models, ranging from the low power single-channel FA251 (1× 250 W) model, to two channel models in 125 W, 250 W, and 500 W configurations, to high power three-channel solutions like the F503 (2× 500 W + 100 W), applicable in a variety of pro audio applications.

The Hypex Fusion Amp plate amplifier range includes eight different models.

All the new FA amplifier plates feature superior AD and DA conversion and combine analog XLR+RCA inputs, analog XLR through, USB (HFD compatible), built-in programmable presets, volume potentiometers, and high-level input in the one channel models, with limiters, signal and clip indicators, and over-temperature protection. The multichannel models also feature digital inputs (optical, AES/EBU and S/PDIF) and digital outputs (AES/EBU and S/PDIF).

Since the first FA releases, Hypex has received lots of market feedback and decided to add the option to ensure the plates can also be operated with remote control for volume control and source selection and introduced a separate IR-receiver module.

With the NCore OEM module range, Hypex has also expanded the options in the NCxxxMP series, incorporating a low power standby power supply in accordance with current ErP regulations, together with a highly efficient switch-mode power supply (SMPS), all on the same module with the Class-D amp. These are used in a wide range of audio applications and provide flat frequency response irrespective of load impedance, nearly frequency independent distortion behavior and very low radiated and conducted EMI. Control is based on a phase shift controlled self-oscillating loop taking feedback only at the speaker output.

The main SMPS providing the power for the amplifier is a compact, high power, highly efficient, regulated half bridge

converter with synchronous rectification on the main output rails, ideal for powering Class-D audio amplifiers. To achieve universal mains input compatibility the SMPS features an automatic input voltage doubler. The Hypex Channel Extension connector provides for an additional low power tweeter amplifier channel.

The most recent in the series is the NCAS500MP—a high efficiency two-channel (400 W and 100 W RMS at 4 Ω) solution for active speakers (the additional letters "AS"). This module will soon be joined by the new NCAS1000MP with two 800 W RMS and 200 W RMS channels, providing a strong foundation for two-way designs.

The low-weight modules feature auto-switching 115 V/230 V universal mains operation, advanced over current protection, and external control operation, ideal for studio monitors, hi-fi systems and demanding installation speakers.

In the Ncore (NC) OEM catalog, Hypex provides the highest quality audio power modules that are able to challenge many Class-A designs, combined with minimum energy consumption. The range includes the NC500 (550 W at 2 Ω), the NC1200 (1200 W at 2 Ω) and the powerful NC2K (2000 W at 2 Ω), able to support a 48 A current, all offering differential audio input and up to ±98 V operation. All models also offer extremely low distortion over frequency and power range, low noise and extensive, microprocessor-controlled error protection, being ideal for high quality stereo and multichannel amps, active loudspeakers for recording and mastering studios. Operation is based on a non-hysteresis fifth-order self-oscillating control loop taking feedback only at the speaker output.

The Hypex NCAS500MP module is a high efficiency two-channel solution for active speakers.

Hypex Ncore OEM modules

The Hypex NC500 OEM and the NC2K are unbuffered amplifier modules leaving the implementation of an input buffer up to the manufacturer.

Hypex | www.hypex.nl

Reaching Out with ICEpower

ICEpower's Class-D amplifier modules are a powerful portfolio available for OEM manufacturers looking for a cutting-edge amplification platform. The ASP, ASC, and ASX modules are compact and efficient audio solutions that combine great sounding Class D amplifiers with switch mode power supplies (SMPS). Together with the ICEmatch and ICEbricks modules, ICEpower offers solutions for any type or level of application.

Recently, ICEpower introduced its most powerful module in the portfolio, the 1200AS, designed for professional sound reinforcement applications. The ICEPower 1200AS module is based on ICEpower's recent ICEedge Class-D chip set, enabling superior sound quality and an ultra-low noise floor (less than 30 µV—130 dB SNR). Its high audio qualities also make it a superior choice in high-end audio amplification for demanding consumer applications.

The module is available as the 1200AS1 model, with a single-channel amplifier, and the 1200AS2 with a dual-channel amplifier, both channels capable of 1200 W into 4 Ω and 1400 W into 3 Ω. Both modules feature a new universal mains SMPS with power factor correction (PFC), with ErP and Energy Star-compliant standby functionality, regulated auxiliary power supplies, wake on signal sense, and a DC-bus output.

According to the Danish company, reliability has also been a key focus in the 1200AS development. As with all other ICEpower products, the 1200AS modules goes through a comprehensive functional test before delivery to market. And another key argument is its low cost,

ICEpower 1200AS, currently the most powerful amp module from the Danish company, is designed for professional sound reinforcement applications.

The new ICEpower 200AS1 module for guitar amplifiers, bass amplifiers, and subwoofers

allowing developers to maximize power and performance.

Another recent product from ICEpower is the new 200AS1 module, specifically designed for guitar and bass amplification. The latest member of ICEpower's AS series, the 200AS1 was designed with a rock-steady regulated power supply of 300 W with universal mains, combined with a 1× 200 W (4 Ω) Class-D power module.

The 200AS1, together with its smaller siblings, the 100AS1/2, have been uniquely designed for guitar amplifiers, bass amplifiers and subwoofers without the need for forced cooling. The DC-bus output also allows powering a separate low power tweeter amplifier, for two-way applications.

Utilizing the same topology, both the 200AS1 and 100AS have the exact same input gain, mechanical dimensions, mounting hole pattern, electrical interface and connectors, making it easy to create a series of products using the same power platform.

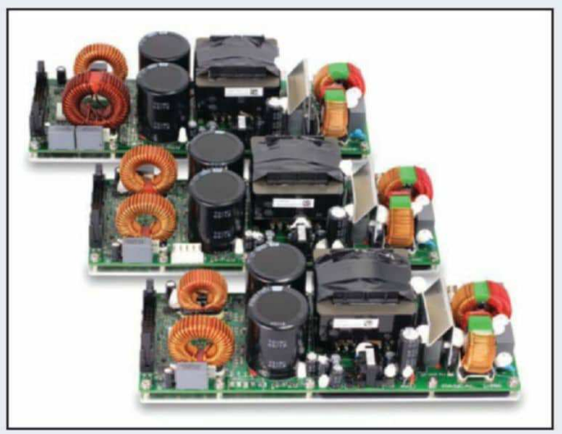
ICEpower |www.ICEpower.dk
Profusion | www.profusionplc.com

Pascal Pro Power

While those in pro audio manufacturing are increasingly aware of Pascal's rise in becoming a leading provider of high-power amplifier electronics to the industry, the company is also expanding to both more demanding high-quality applications.

The Copenhagen, Denmark-based company recently expanded its catalog with a new range of dedicated amp modules, with features and performance optimized for integration within power amplifier designs, across the breadth of live sound and commercial and custom installation applications. A range of six modules, based on the company's proprietary UMAC Class-D technology provides the highest power density/lowest form factor ratio, and enables both constant voltage (100 V/70 V) and low impedance output mode operation and a wide range power configurations; for half rack 1RU to full 19" 2RU form factors.

These modules are equipped with comprehensive protection features and provide Energy Star-compliant auxiliary power outputs. Advanced real-time readouts of protect/mute, temperature and clip signals are accessible for DSP and network control implementations, and I/O stages. At the same time, the modules use UREC PFC power supplies, enabling universal AC mains operation anywhere in the world.



The new L-PRO2 Series Class-D modules from Pascal provide 50% more power from the same form factor as the S-PRO series.

Depending on the combination of modules used within a power amplifier, it is possible to drive any share of the total power capability to any output stage. This offers installers and integrators complete freedom to drive exactly the required amount of power precisely where it is needed, addressing asymmetrical power requirements from the same power amplifier.

All these latest modules feature Pascal's plug-and-play 26-pin interconnect, making them inter-connectable to a single front-end I/O/DSP board design. A single front-end module can thus be used in multiple power amplifier design configurations, reducing the required engineering work across multiple product design projects.

Pascal also released its new L-PRO2 series amp modules, which provide 50% more power from the same form factor as its existing S-PRO series, using Pascal's proprietary UMAC Class-D technology. The series includes a new standard in power density for Class-D power amplification, as the world's smallest 1500 W RMS amp module.

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

Pascal Audio | www.pascal-audio.com

Powersoft LOTO DSP Plates

Powersoft recently expanded its OEM components range with the introduction of the new LOTO DSP one-two input/four output Advanced DSP board. The new LOTO advanced processing board was specifically designed to provide advanced DSP functions to any powered product. It comes in a Basic and Advanced version, both featuring the same characteristics of Powersoft's high-end X-Series platform, including parametric raised cosine filters, custom FIR and IIR equalizers, as well as TruePower, RMS and peak limiters, active damping control, and live impedance all completely supported and managed via Powersoft's ArmoníaPlus software.

Powersoft LOTO Basic and Advanced DSP plates

LOTO Advanced further increases routing options by adding 2× AES3 digital input streams through XLR connectors, and the support of Dante digital audio networking architecture, with two inputs and two outputs.

Compatible with the Digimod PFC2, PFC4, IpalMod, Litemod family, and MiniMod4 OEM amp modules from the Italian brand, LOTO boards are ideal for any application with high processing and digital audio networking requirements, and offer a DSP solution for top-level multi-way systems and line arrays where complete control and premium performances are needed, all easily controlled via Powersoft's ArmoníaPlus software suite.

The DSP-Lite ETH board is another new addition to Powersoft's OEM range. This is an Ethernet-based one-two input/three output processing board designed for the LiteMod Series, MiniMod4, and DigiMod PFC2/PFC4 amp modules, integrating an extremely compact interface panel compatible with mono-in/link-out or stereo-in configurations.

Compatible with Powersoft Integration Kit platform, the built-in Ethernet port allows access to the DSP-Lite ETH processing capabilities directly from a PC running ArmoníaPlus software, to easily program and store up to four presets that can be selected by the end user.

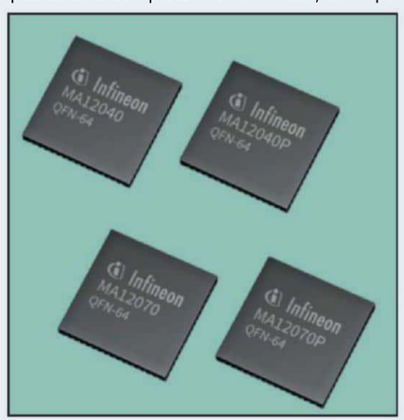
Powersoft | www.powersoft.com

Infineon Merus

Throughout the years, Infineon has worked with audio manufacturers across the world, helping developers with best-in-class products backed by advanced technologies. The company's already successfully established audio portfolio was recently complemented by the latest high-performance Class-D amplifier solutions, following the acquisition of Danish-sensation Merus Audio, founded in 2010 by Hans Hasselby-Andersen and Mikkel Hoyerby. In its September 2018 edition, *audioXpress* published a complete profile in its Audio Amplifier series.

The now rebranded Infineon Merus range of energy-efficient integrated Class-D Multi-Level amplifier solutions is an important contribution to minimizing heat and design space, and maximizing audio performance and battery playback time for smart home and battery-powered speakers. The Infineon Merus integrated multilevel audio amplifier ICs can outperform traditional Class-D audio amplifiers with products that are cooler, smaller and lighter, more robust and flexible.

The current Infineon Merus portfolio includes fully integrated single-chip audio solutions as well as discrete audio solutions with highly scalable driver and power MOSFET combinations, suited for applications with output power levels from 20 W to 2000 W per channel. The multilevel audio amplifier ICs use additional on-chip MOSFETs and capacitors to produce outputs with a higher signal granularity, allowing for higher switching frequencies and/or multiple output signal levels—typically up to five voltage levels.



Infineon Merus ICs

The Merus technology positively impacts power consumption, solution size, audio performance, electromagnetic interference, and BOM cost, while allowing potential LC filter removal, low THD+N, and cooler operation.

Even in idle and near-idle mode, traditional Class-D amplifiers continue to have a lot of internal switching activity, which increases power consumption. The Merus integrated ICs use scalable signal "granularity" to keep the power loss extremely low. Due to the proprietary circuits' architecture, there is virtually no switching loss measurable in idle mode, and in realistic audio playback situations, the solution reduces power consumption in AC input and in battery powered audio applications.

Infineon | www.infineon.com
Merus | www.infineon.com/audio

Purifi Audio, Taking It Further

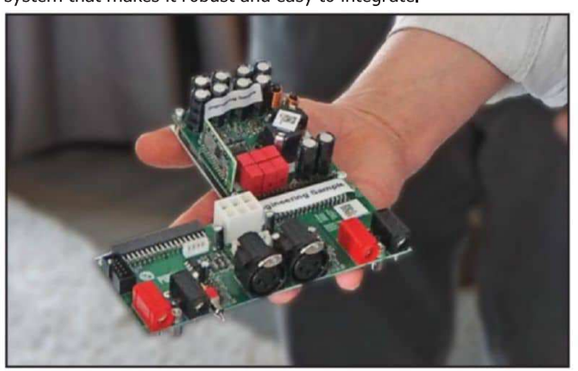
A new Danish company, founded by Bruno Putzeys, Lars Risbo, and Peter Lyngdorf was recently announced. The company is Purifi Audio, and as the founders' names involved in the venture imply, they have been developing Class-D amplifiers with record-breaking performance "unmatched by amplifiers of any operating class." The goal of the new Danish project, they add, combines work on Class-D amplifiers and loudspeaker drivers, specifically intending to reduce distortion in both, with "magnitudes lower intermodulation distortion across frequency and SPL."

Purifi Audio was born out of the research work of Class-D pioneers Bruno Putzeys (of Philips, Hypex and Kii Audio fame) and Lars Risbo (Toccata Technology and Texas Instruments), according to which, their new "Eigentakt" or self-clocking amplifier technology achieves a roughly 20 dB improvement over previous generations.

The result of years of research deepens into the self-oscillation behaviors in Class-D amplifiers with algorithms that, when applied in control loops, improve existing designs by an order of magnitude or more. "The Eigentakt technology is unique," explains Putzeys. "Amplifiers carrying the technology will be exceedingly indifferent to speaker load, volume level, and signal content, resulting in negligible THD and IMD levels, and will feature exceptionally clean clipping." According to measurements by Purifi, the Eigentakt circuit is expected to outperform any audio amplifier known, regardless of technology or class, with THD and IMD quoted as below 0.00017% at all frequencies and power levels, and the frequency response in the audio band remains within ±0.01 dB under all load conditions.

The first working platform to be unveiled by Purifi is the 1ET400A, a 400 W Class D single-channel, analog-input, amplifier module. The technology is said to be completely indifferent to frequency, level and impedance variations, with measured intermodulation distortion significantly below any other technology. The 1ET400A module's specifications detail extraordinary low noise levels ~11.5 µV A-weighted with over 94% efficiency at an output power >400 W with 1% THD, or ~25 A output current, featuring <0.00017% THD+N (-116 dB) at 100 W and 4 Ω, and ~131 dB(A) dynamic range with a 12.8 dB gain.

As the company details, Purifi's continuing research into nonlinear control theory has produced the first mathematically exact large signal model of self-oscillating amplifier controllers. This breakthrough allows complete optimization of the feedback circuit and improves performance by at least an order of magnitude over existing implementations. The amplifier module further incorporates a comprehensive protection system that makes it robust and easy to integrate.



The new Purifi Audio 1ET400A amplifier module

The Purifi-Audio team shown (left to right): Bruno Putzeys, Lars Risbo, Claus Neesgaard, Morten Halvorsen, and Carsten Tinggaard.

These circuits and methods enable a high loop gain (>75 dB) in the entire audio band, corresponding to an unprecedented 110 MHz Gain-Bandwidth Product, with negligible intermodulation distortion (IMD), a very good measure for how well an amplifier handles complex signals. THD remains extremely low at any frequency and any power level right until clipping, which according to Purifi translates into a total lack of sonic signature, and an ability to reproduce any type of music without preference for genres or production style.

The 1ET400A module also features a high power supply rejection ratio (PSRR), which places no particular demands on the power supply quality, and will be able to handle difficult loudspeakers with ease thanks to its load-invariant frequency response and negligible output impedance. Purifi also details a controlled, second-order low-pass response, enabling a very flat audio-band response with a sensible, 60 kHz bandwidth, and reduced sensitivity to out-of-band noise from DACs, reducing the requirements on the DAC reconstruction filter and allowing better behavior with outboard DACs.

The Purifi 1ET400A's compact size and claimed high reliability will enable a broad range of premium audio quality applications. The module was specified for very low idle losses and reduced electromagnetic interference (EMI), allowing little to no effort to pass regulatory tests. The enormous loop gain allows relaxed timing of the power MOSFETs without degrading audio performance, as Purifi also states. Finally, the overall implementation/integration of the 1ET400A module will save time and cost for system integrators, with an architecture that completely eliminates heterodyning in multichannel applications.

Since the company was able to demonstrate the working principles and the implied advantages of the self-oscillating approach, the market response has been enthusiastic. The first licensing agreements were confirmed, starting naturally with Peter Lyngdorf's company, Lyngdorf Audio.

The 1ET400A amplifier module was effectively used in a prototype of the latest Lyngdorf MXA-8400 amplifier, an 8× 400 W Class-D amplifier shown at the Integrated Systems Europe (ISE) 2019 show. Next to be confirmed was an agreement with NAD Electronics (part of the Lenbrook group of companies). NAD intends to create a new amplifier platform for future, yet-to-be-announced products.

At the time of closing of this edition of *audioXpress*, Purifi had already updated the specifications for its 1ET400A amplifier module, which can be found at www.purifi-audio.com.

Why Smart Power Audio Amplifiers Are Becoming Important to Smartphones

It's not unusual to see friends gathered around a sideways-turned phone to play a video game or share the viewing and listening experience while watching videos on the phone's external speakers.

By
Eric Eklund
(Product Line Manager, Cirrus Logic)

As more users disconnect from traditional cords, the need for smart power amplifiers is becoming more critical, particularly for smartphone manufacturers who want to differentiate their products.

There seems to be no end to the amount of content people are consuming over their smartphones, tablets, or other portable devices. As smartphone screens become larger and brighter and offer a better overall visual experience, consumers are experiencing that mobile content in new ways—not just via their headphones. It's not unusual to see friends gathered around a sideways-turned phone to share the viewing and listening experience while watching videos on the phone's external speakers.

Over the years, for many phone designs screen sizes became larger—but the audio experience hasn't always kept up, especially while in "loud" speaker mode. As a result, more high-end phones are offering stereo as an added feature, often with a microspeaker positioned at the top and the bottom. Now, when users turn their stereo-enhanced mobile phones on its side for viewing (i.e., landscape view), there is a speaker on either side to help create a better "stereo" listening experience.

This trend toward stereo audio in smartphones, and other audio-playing devices, shows that audio amplifiers are playing a more important role in smartphone design. As a result, audio amplifiers are seeing a big jump in both unit volume and technology (see **Figure 1**). According to SAR Insight & Consulting's "The Global Market for Standalone Speaker Amplifiers," market growth report, "Market shipments for audio amplifiers have grown 3× from 2010 to 2018, due not only to the increasing shipments of audio playing devices but also the increasing number of channels per device."[1]

The report also states that stereo speakers have become widely used in the high-end smartphone market (phones priced above $550), especially since the 2016 introduction of stereo in the iPhone 7 mobile digital device. Since then, many other flagship smartphones have adopted stereo speakers in their products as well. The analyst firm also expects stereo sound to extend into mid-tier phones (defined as those between $200 and $500) estimating that nearly 600 million amplifiers shipped for use in the mid-tier market in 2018, with an anticipated rise to almost 700 million by 2023.

Growing Importance of Boosted Audio Amplifiers

However, the sweeping trend to include stereo in smartphones does not necessarily ensure a high-quality audio experience. For this to happen, the microspeakers should also be accompanied by a smart-boosted audio amplifier, which enables driving the speaker to maximum loudness while protecting the speaker from damage.

Smartphone audio designers should ensure a better, louder, clearer audio sound with the use of high-voltage audio amplifiers, speaker monitoring analog-to-digital converters (ADCs), and advanced algorithms that keep microspeakers within their safe operating conditions. Also, two speakers naturally mean two amplifiers. So size, efficiency, battery current management, and audio performance are twice as important. Thus, as smartphones go stereo, the role of the previously unheralded audio amplifier is becoming more elevated.

A design checklist to ensure you get the most from your smart-boosted audio amplifiers includes:

- Size—Board space is always a premium when packing features into a portable device, and one option is for two microspeakers to be placed at the top and bottom of the phone. Audio amplifiers must be smaller and more integrated, especially now that there are two. To address the size issue, Cirrus Logic used 55 nanometer technology to design its newest CS35L41 smart-boosted amplifier with DSP, making it easier for OEMs to incorporate a two-speaker design without sacrificing space for other new features.
- Power Efficiency—The SAR Insight & Consulting study also confirms that Class D has all but replaced Class-AB amplifiers in many markets due to the increased efficiency of Class-D audio amplifiers. However, even more power efficiencies can be achieved in the design. For example, the 11-V boosted Class-D amplifier from Cirrus Logic also uses a Class H DC-to-DC converter to boost the supply with higher efficiency over Class G, while additional battery management systems predictively adapt to changing audio, speaker and battery conditions to regulate battery current and save power. This underscores the changing technology and "boosted" role of the boosted audio amplifier (see **Figure 2**).
- Speaker Protection—According to SAR Insight & Consulting, the use of speaker protection algorithms from an embedded DSP within the amp has become more common in a number of applications where small form factor speakers are used (e.g., smartphones). These algorithms enable higher output without damaging the speaker. For example, the Cirrus Logic audio amplifier pushes over 5 W of power to a speaker rated at 1 W. This newer class of boosted audio amplifiers can push the power needed for higher excursion to produce louder, clearer audio, especially over speakerphone, without damaging the ultra-small micro speakers. SAR Insight's report further states that "the use of speaker protection in mid-range smartphones has grown rapidly from 2013 as OEMs look to maximize speaker performance and use this alongside other enhancements (e.g., high-end CODECs) to differentiate their designs."
- Audio Enhancement—Even with all these features, not all audio enhancement is created equal. To produce better clarity and enhanced stereo audio to match those vibrant, larger screens and visual content, advanced algorithms within an integrated DSP should also address audio design issues (e.g., equalization, compression and psychoacoustic enhancements to fine tune audio playback).

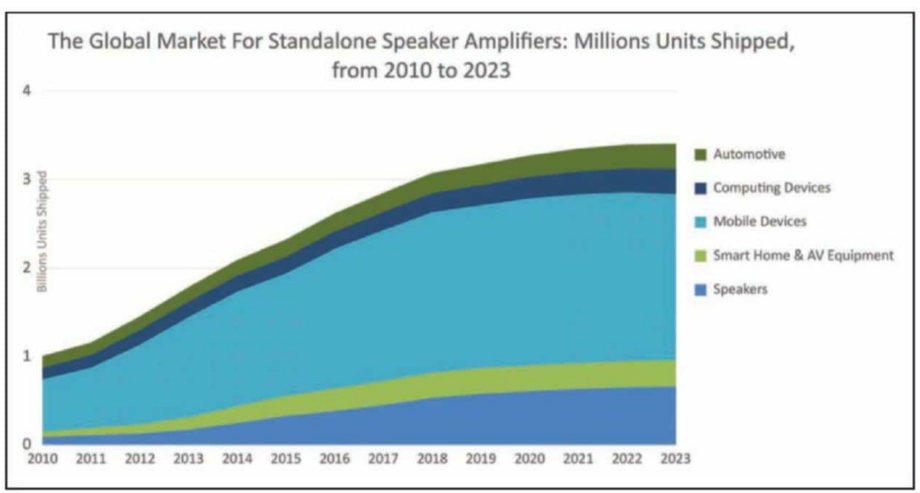


Figure 1: "The Global Market for Standalone Speaker Amplifiers" report indicates that market shipments for audio amplifiers have grown 3× from 2010 to 2018. (Image courtesy of Peter Cooney, SAR Insight & Consulting, 1Q 2019)

As users cut the cord on more traditional cable or satellite TV viewing and play more untethered content, smartphones and other mobile devices are fast becoming a primary platform for streaming and enjoying content. To match the clearer and brighter visual experience, more and more phones are moving away from single channel, or mono audio design, to two-channel stereo to offer a better listening experience. As this trend continues, the need for smart power amplifiers is becoming more critical. This means smartphone OEMs will not only look for space savings but also improved power efficiency and audio clarity, in both headphone and speakerphone modes, to differentiate their products.

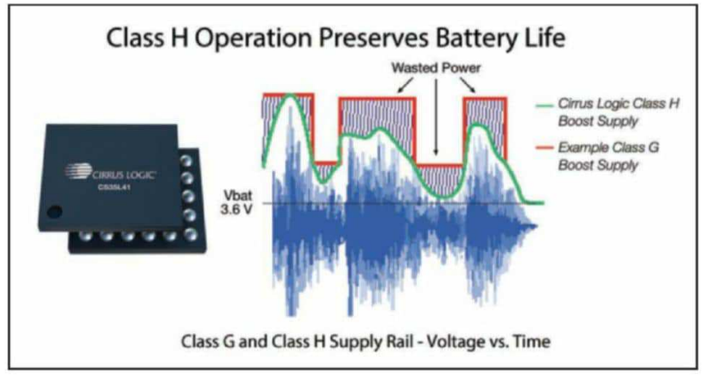
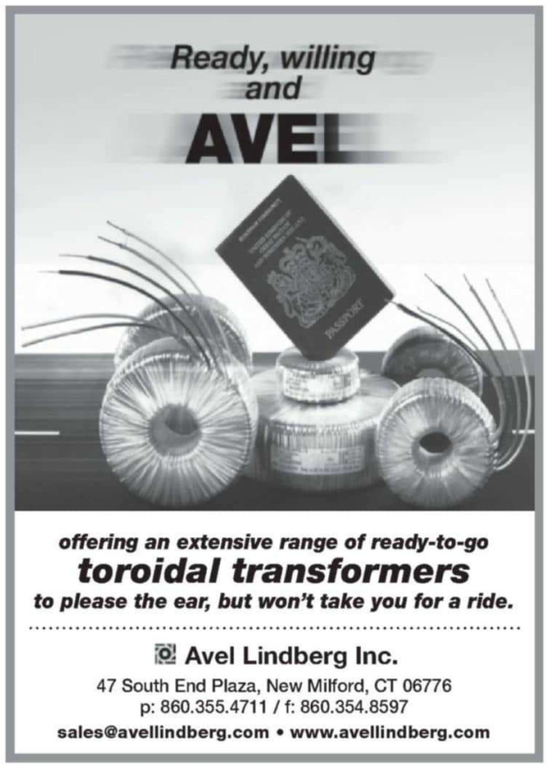


Figure 2: Class G and Class H Supply Rail Voltage vs. Time underscores the changing technology and "boosted" role of the boosted audio amplifier.

References

[1] Information taken from the SAR Insight & Consulting FOCUS on Audio Power Amplifiers: Market Growth, Channel Count, Amplifier Types, Input Type, Class, Speaker Protection; Principal Analyst Peter Cooney, published 4Q 2017 and updated data from SAR Insight & Consulting's Audio Signal Chain ICs Forecasts DATAbase in 1Q 2019.



Powersoft OEM Power Amplifier Modules

Flexible Power for Active Systems

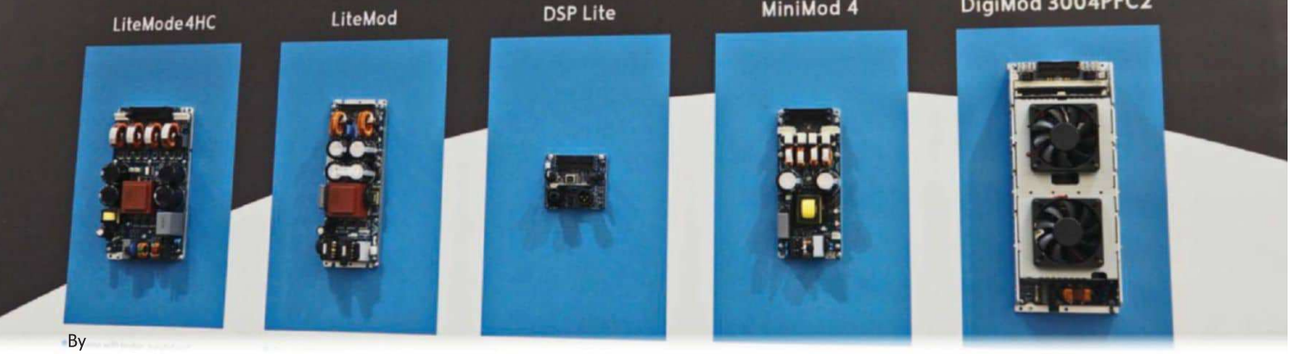


Photo 1: Powersoft offers a full range of high-powered, compact, and scalable solutions for loudspeaker manufacturers.

By
Ward Maas
(The Netherlands)

In this article series, *audioXpress* continues to explore the history, technological developments, and new generation products from some of the most innovative amplifier platform companies serving all segments of the audio industry. We continue this series with Italian company Powersoft and we will focus specifically on the brand's OEM amplifier modules.

Powersoft's story began in 1995 when the brothers Luca and Claudio Lastrucci, and their friend Antonio Peruch who, during the period that follows university studies, shared the two brothers love for music, technology, and innovation. They called their new company Powersoft, and in their kitchen, they began to modify amplifiers—and with their background in power electronics—design new ones from scratch. As switch mode power electronics started to become more common, they became interested in designing a Class-D amplifier. Not only did they succeed in making it work in a stable way, but also at high levels of power. Their first amplifier delivered 150 W. Ever since then, Powersoft has been focused on the high-power segment of the market. The company also believed in active speakers and it was the first to develop Class-D amplifier modules that could be integrated into professional audio speakers.

In the August 2014 issue of *audioXpress*, Stuart Yaniger looked in detail at the foundational amplifier technologies developed and perfected by Powersoft Audio and the unique company philosophy. In this highly recommended reading, the article details the company's Power Factor Correction (PFC) technology integrated into switching mode power supplies and its Class-D Pulse Width Modulation (PWM) high-efficiency output stage. We are not going to repeat the detailed description of those technologies, shared by the company's professional audio, installation, and touring amplifiers, and for this article, we will focus on the OEM modules.

Innovation

Powersoft products are entirely made in Italy. Today, Powersoft counts more than 90 employees at its headquarters in Scandicci (Florence, Tuscany), engaging in research and development, technical assistance, marketing, sales, and logistics. The R&D lab by itself involves more than 30 employees with interdisciplinary skills, which explains a lot about the company's pioneering innovations.

From its initial concept, every electronic project (which comprises PCB routing), mechanical project, firmware, signal processing algorithms, and software, everything is developed entirely within the head office premises, where all tests are also carried out—both for certification purposes and to assure proper operation of the products in the most demanding operating conditions. Powersoft, now a publicly traded company in Italy, also invests more than 10% of its turnover in the research and development of technologies applied to products and solutions dedicated to the pro-audio industry. And not only in amplification. In recent years, Powersoft has made significant inroads into innovative transducer technologies as well. As production also works closely with development, improvements deriving from market experience or from the results of research are easily implemented.

Three different production sites, focused on rack mount amplifiers and amp modules, are located respectively in S. Giovanni in Persiceto (Bologna), Ronchi dei Legionari (Gorizia), and Cotona in Italy, where around 100 people are employed. In line with the company values, these production sites make extensive use of solar energy. The entire production of Powersoft products is carried out using green energy—a trademark of its own technology efforts in amplification.

More than 20 years ago, Powersoft was the first company to introduce PWM in a way that enabled it to achieve an efficiency level that was unthinkable at that time. PFC was a natural product feature, allowing efficient energy savings and reducing heat generation due to cable losses.

In fact, the company's Smart Rails Management implemented into amplifiers and power supply and back electromotive force (EMF) energy recovery paved the path for Green Audio Power—Powersoft's registered trademark that identifies environmental friendly products.

"Saving energy has been an integral part of Powersoft, but fits perfectly with the demand for truly energy-efficient products based on global warming and increasing cost of petroleum-based products," Powersoft CEO Luca Lastrucci says. "These phenomena have produced a marketing trend that has been particularly well received in the entertainment industry where there is a special consciousness about what is environmentally friendly or green."

Powersoft has become a prime example of Italian excellence in audio and one of the leading companies in the global pro-audio industry with the most advanced products and technologies, reinforced by multiple associated patents. According to the company, there are currently more than 600 million watts of Powersoft amplification out there in the market.

The company is one of the world leaders in lightweight, high-power, rack-mounted, energy-efficient amplifiers for the professional audio market. Its amplifiers can be found in an array of markets, ranging from stadiums, sporting arenas, theme parks, performance venues, and airports to convention centers, churches, cinemas, and clubs and are used by the world's leading tour sound companies.

The subsidiary Powersoft Advanced Technologies Corp., located in New Jersey, is in charge of marketing and technical assistance for the US market.

OEM Approach

As mentioned, while Powersoft always had a focus on high-power rack-based products for the install and touring market, the company also introduced Class-D

	DSP-4	DSP-D	DSP-Lite	Small Heatsink	Medium Heatsink	Large Heatsink
DigiMod 3004PFC4	✓		✓			✓
DigiMod 3004PFC2	✓	✓	✓		✓	
DigiMod 3000PFC	✓	✓				✓
LiteMod 4HC	✓		✓		✓	
LiteMod HV	✓		✓	✓		
LiteMod	✓		✓	✓		
MiniMod 4	✓		✓			
IpalMod	✓					✓

Figure 1: The Powersoft OEM modules configuration table showcases the multitude of OEM options are available.

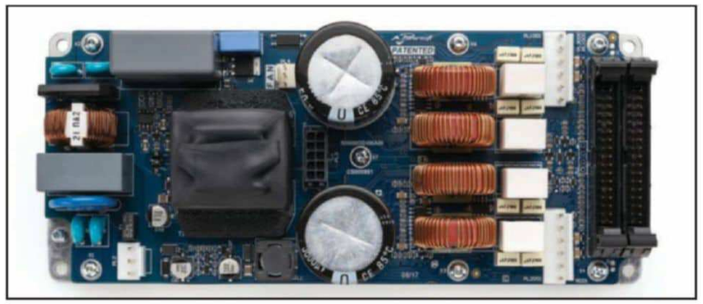


Photo 2: The Powersoft MiniMod 4 is an integrated module offering universal mains switching mode power supply with PFC and a total of 600 W of audio amplification, configurable as four, three, two channels, or single high power output channel.

amplifier modules for the OEM market. Initially with the DigiMod high-power modules, the company established a standard for active sound reinforcement systems in terms of quality, reliability, and robustness, complemented by flexible signal processing tools that allow tailoring the behavior of the module to perfectly fit any design. As with all Powersoft OEM products that came afterward, they include certifications, reports, and documentation to prevent designers from wasting time, energy, and resources.

Current OEM products range from the 150 W and more MiniMod series, the 600 W and more LiteMod series, to the 1000 W and more DigiMod series (see **Photo 1**). Special products like the IpalMod 8500W with 180 V output voltage and 95 A maximum output current are also available, allowing loudspeaker manufacturers to implement Integrated Powered Adaptive Loudspeaker (IPAL) technology, a functional solution to overcome the limitations of traditional low-frequency transducers.

The range now also includes special products such as the D-Cell504, suited for 12 V/24 V/48 V power supply operation. The D-Cell504 is an incredibly small and lightweight amplifier with 2× 350 W available at 4 Ω, bridgeable to have 700 W on a single 8 Ω load. It's proven to be a suitable solution for everything from small-mid power professional loudspeakers to studio monitors, home cinema systems, and even hi-fi applications. Including a PFC power supply and a two-channel output stage with easy mechanical integration, the D-Cell504 power supply can drive two amp units, one of which can be bridged to drive three-way systems, as well as two-way bi-amped or symmetrical loads with higher power even at higher impedance.

The D-Cell504 IS Series of modules also includes an integrated two input/four output DSP, which can be integrated in a single module with an aluminum heatsink panel, an interface panel with input volume potentiometer, controls, and LED indicators, which include limit and clipping functions signaling. As with all Powersoft amplification products, these solutions can all be remotely controlled through the company's powerful Armonía Pro Audio software suite.

All in all, a complete range of modules, and all an OEM loudspeaker manufacturer needs to design powered systems, including pre-configured cables. Having a look at these products, the first thing that comes to mind is the completeness of the range (see **Figure 1**), not only with integrated wide operating voltage PFC power supplies, but also with all the accessories needed to easily design a complete product.

A Closer Look

We looked in more detail at the MiniMod 4, a compact and flexible 4× 150 W amplification platform offering universal mains switching mode power supply with PFC and a total of 600 W of audio amplification (see **Photo 2**). This compact module has low standby power consumption with smart rails management that also helps to reduce heat. The MiniMod 4 boosts flexibility, which is illustrated by the fact that it can be configured with five different output types:

- 4 x Single Ended (SE): Four independent channels with 4× 150 W at 4 Ω SE
- 2 x Bridge Tied Load (BTL): Two couples of channels bridged with 2× 300 W at 8 Ω BTL
- 2 x SE + 1 × BTL: A couple of channel bridged while the remaining two channels are kept single ended with 2× 150 W at 4 Ω SE + 1 × 300 W at 8 Ω PBTL
- 2× Parallel Tied Load (PTL): Two couple of channels obtained by paralleling two channels of the module with 2× 300 W at 2 Ω PTL
- 1× Parallel Bridge Tied Load (PBTL): A single channel can be obtained by bridging the PTL configuration with 1× 600 W at 4 Ω PBTL
- These can be used in high channel count applications, three-way and two-way designs, subwoofer, and satellite systems.

Nowadays, every power amplifier module comes with a fault protection system designed to protect people from severe or fatal electric shocks and to avoid damage on the amplifier or the loudspeakers in case the electrical or thermal parameters run out of the design criteria limits or when radical changes in the environmental conditions occur. There is a clear separation between the power supply and the amplifier protection sections. Both are independently monitored.

For the universal input power supply there is a mains overcurrent protection in case of an over current situation due to internal failure (not customer replaceable or resettable). A mains overvoltage protection acts at 280 V_{RMS}, switching off the power supply and resetting at 275 V_{RMS}.

Furthermore, there is a three-stage power supply temperature protection starting at 80° when the fan starts. At 100°C, the power supply starts to reduce the amplifier output stage voltage and when the temperature exceeds 120°C the output stage is switched off.

The amplifier itself has a bit stricter thermal protection. The amplifier starts to reduce the output stage voltage at 85°C. When the temperature exceeds 100°, the output stage goes in safety thermal switch off mode.

In case of a too low load impedance or if the loudspeaker line has a short circuit (because of voice coil damage, wires that short circuit, and improper wiring, etc.), the amplifier output current protection trips. When the output current passes the threshold of 12 A peak, the amplifier section switches off instantly.

High-frequency signals tend to stress the amplifier section of the module as well as the loudspeaker's voice coils. If the input signal above 20 kHz is higher than 450 mVrms, the protection will activate, starting the fan first. With an increasing high-frequency content input, the amplifier module will start to reduce the output stage voltage until finally switching off the output stage completely.

All the protection monitoring signals are available on IDC connectors complete with auxiliary power supplies to enable customers to develop their own product implementations. To be able to evaluate the MiniMod 4,

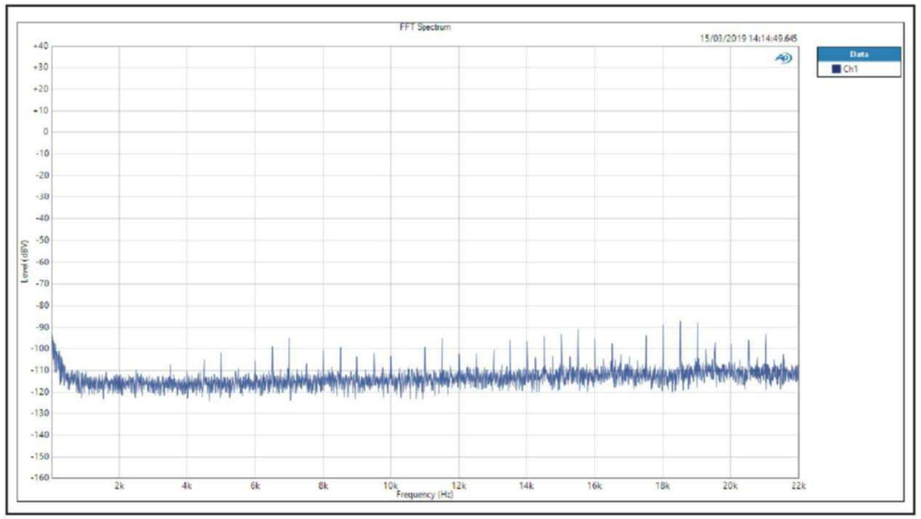


Figure 2: Looking at the MiniMod 4 amplifier module's specifications, we see that the single-end fast Fourier Transform (SE-FFT) at 8 Ω idle spectrum has a noise floor that stays below -110 dB on average with some spikes up to -90 dB.

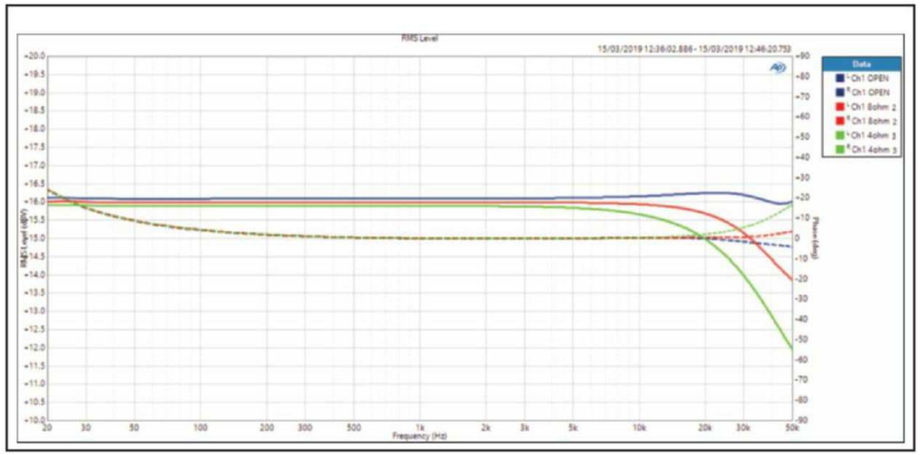


Figure 3: The MiniMod 4 amplifier module's single-end frequency response versus phase will vary depending on the load.

About the Author

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

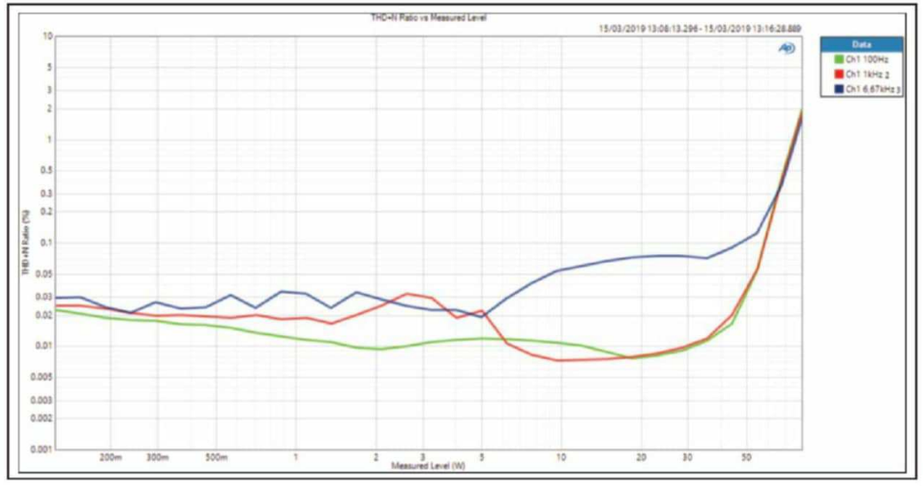


Figure 4: The total harmonic distortion plus noise (THD +N) versus power measurement at SE-8 Ω remains below 0.1% worst case (6.67 kHz) until the amplifier starts to clip.

there is the evaluation board KTP00477, which is an analog audio interface that includes a thermal limiter, a clip limiter, and other functions (mute, energy save enable) in order to evaluate the module without using a DSP. By correctly setting its limiters, it will be possible to run a long-term test without audio interruption, which would otherwise be triggered due to the module internal protection of over temperature or over power.

Certainly for Class-D amplifiers, apart from safety regulations, the EMC regulations are an area of particular interest due to the "switching" nature of these products. The MiniMod 4 comes with a comforting long list of relevant standards with which it is compliant. Clearly, Powersoft is living up not only to its own environmental vision but also safety and robustness.

Users working on the development of a project using this Powersoft module will be reassured with the remarkable quantity of information available to help them in the design. From thermal calculations, complete circuit descriptions to interface the Powersoft modules, to the already mentioned cabling and heatsinking solutions. Products designed around the hardware modules—the amplifiers, power supplies and DSP solutions—have the strong benefit of being fully configurable and controllable with the Powersoft Armonía software solutions, which the company continues to develop and expand.

When we examined the Powersoft MiniMod 4 amplifier module, we noticed it has a fixed frequency PWM modulator with multiple feedback. According to Powersoft R&D Director Claudio Lastrucci, this allows a high level of predictability and immunity from intermodulation artifacts. Without entering into the discussion regarding its qualities versus the self-oscillating variants, it is evident that Powersoft found a nice balance between practical product use and required quality.

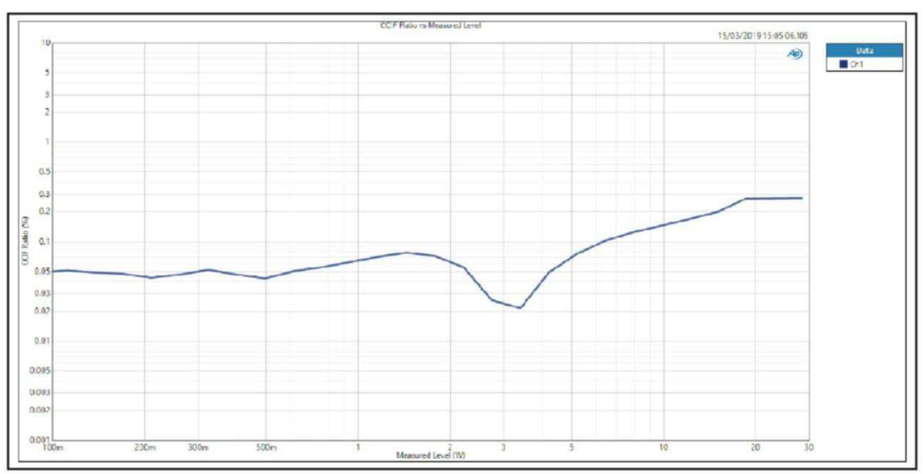


Figure 5: The MiniMod 4 amplifier module's single-end intermodulation distortion using the CCIF standard is 18.5 kHz+19.5 kHz at 8 Ω.

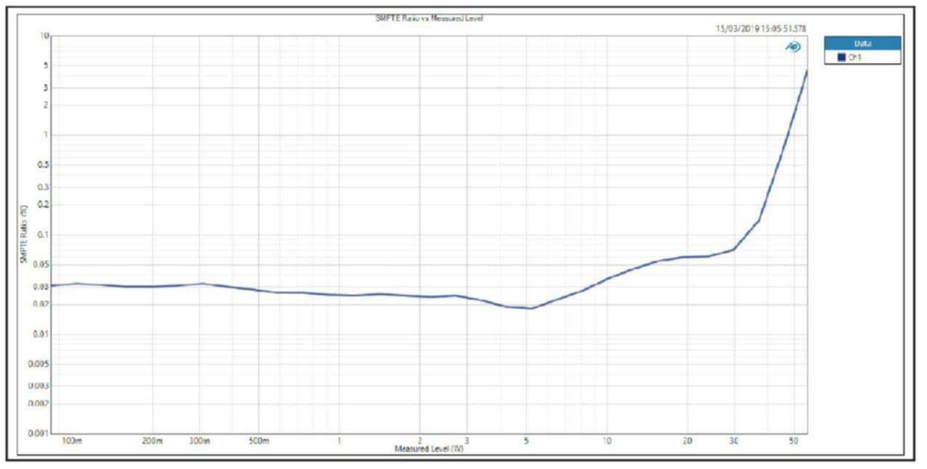
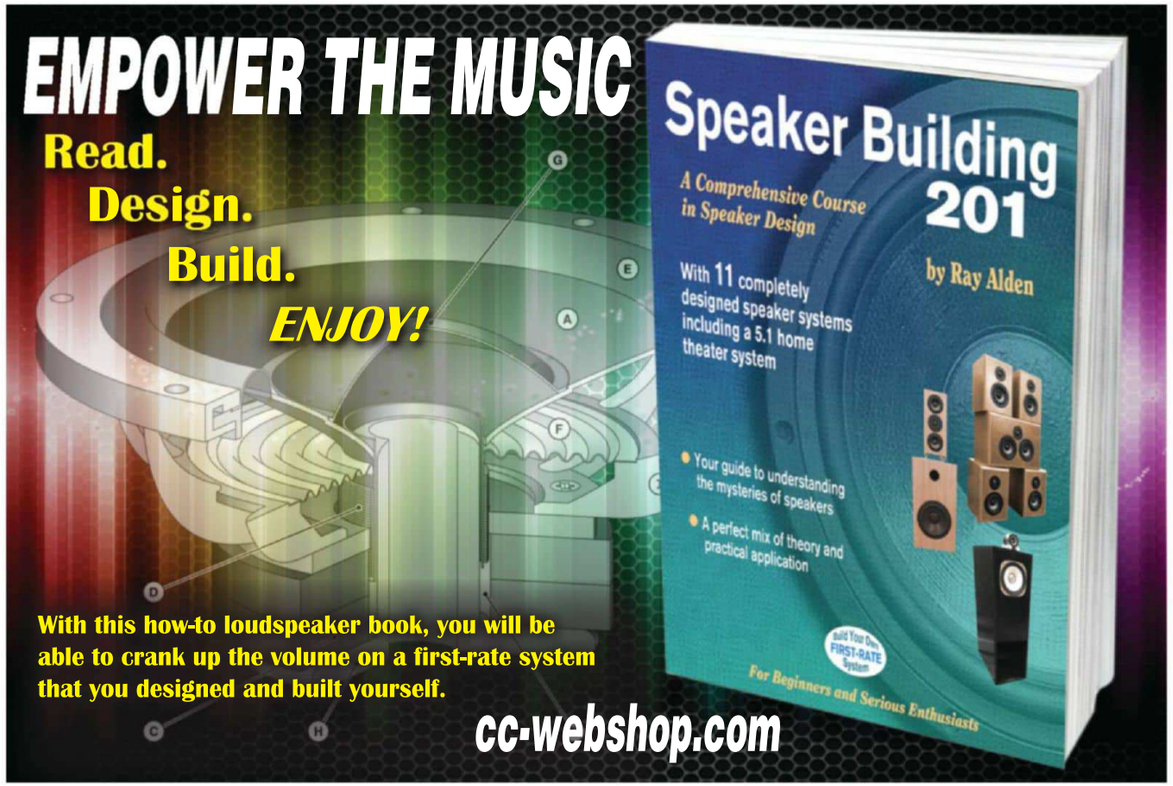


Figure 6: The MiniMod 4 amplifier module's single-end intermodulation distortion using the SMPTE standard is 60 Hz+7 kHz at 8 Ω.

Looking at the specifications of the MiniMod 4 amplifier module, the single-end fast Fourier Transform (SE-FFT) at 8 Ω idle spectrum has a noise floor that stays below -110 dB on average with some spikes up to -90 dB. (see **Figure 2**). When in SE at 8 Ω, the frequency response is flat with -0.25 dB at 20 kHz That will also vary depending on the load, with -1 dB at 20 kHz for a 4 Ω load (see **Figure 3**). The total harmonic distortion plus noise (THD +N) versus power measurement at SE-8 Ω remains below 0.1% worst case (6.67 kHz) until the amplifier starts to clip (see **Figure 4**). The CCIF and Society of Motion Picture and Television Engineers (SMPTE) intermodulation distortion measurements remain below 0.1% (see **Figure 5** and **Figure 6**).



[CCIF intermodulation distortion testing consists of using two equal-amplitude, high-frequency signals spaced closely in frequency. The SMPTE standard specifies a two-sinewave test signal consisting of a low-frequency, high-amplitude tone linearly combined with a high-frequency sinewave at one-quarter the amplitude of the low-frequency tone.]

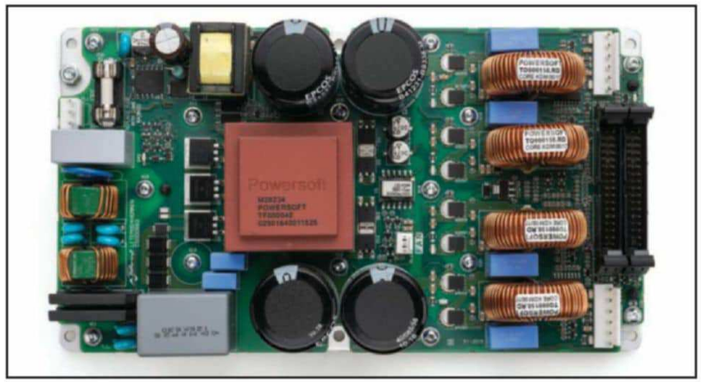


Photo 3: The Powersoft LiteMod 4HC amplifier module with PFC-equipped power supply for universal mains operation and four output stages delivers up to 2400 W total. It can be configured as four, three, two channels, or single high power output channel and support 2 Ω operation.

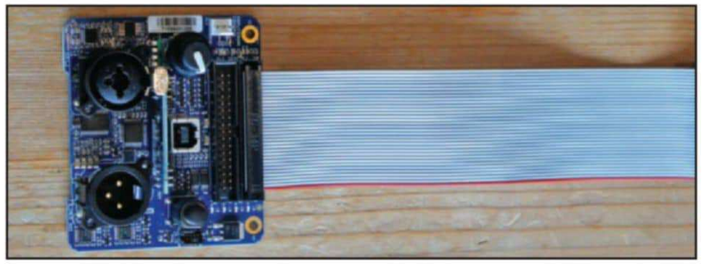


Photo 4: DSP Lite PCB, a complete DSP module for active speakers.

Photo 5: The assembled DSP Lite module with front plate

These values are good. Not extremely good, but considering these modules can be used in SE, BTL, PTL, or PBTL modes, the measurement values do vary slightly depending on the load and the modus operandi. It is very clear that this is a well-thought-out and practical amplifier module platform, capable for a wide range of applications.

In the professional audio industry, Powersoft has been known for its rugged, high-power, rack amplifiers for a long time. With its amplifier modules and the range's full complement of hardware and development tools, it becomes an attractive supplier for the OEM market and a powerful recipe for many active loudspeakers in multiple demanding application areas.

Working with a LiteMod 4HC Module

To demonstrate how easy and flexible the solutions that Powersoft offers are, we assembled a working system with a LiteMod 4HC module (see **Photo 3**). The LiteMod 4HC can be seen as MiniMod 4 on steroids, ideal to power sound reinforcement systems. It delivers 750 W at 4 Ω SE and can be bridged in BTL mode with 1400 W at 8 Ω, and 2400 W at 4 Ω PBTL. It is enough power to experiment with in very loud but efficient configurations.

But even a high-efficiency Class-D amplifier will generate heat. We can calculate an efficiency of 80% and 6 dB crest factor with 3400 W peak. The result is: 3400 (1-0.8)/4 = 170 W to be dissipated. With a thermal protection set at 75°C (167°F) and an ambient temp of 45 °C (113°F) it gives (75-45)/170 = 0.14°C/W or better requirement for the heatsink.

Powersoft has several heatsinks from which to choose, depending on the products you use. They all have a few things in common: they are predrilled, and they have an area for a brand's shield, PowerCON in/out connectors, and a place for DSP/connectors to the outside world. Of course, users can build in a DSP, connectors, and heatsink that matches with their products.

The next step is to choose the amplifier configuration. For instance, if a company wants to build a three-way line array element, it can choose a BTL setup for the bass, a SE in 4 Ω for midrange and a SE in 8 Ω for the highs. Given the flexible Powersoft approach, almost any configuration can be slected and combined with one of the brand's DSP solutions (see **Photo 4**). With that in mind, the appropriate cables can also be chosen from the bundle supplied in the sample system, and connected, including the DSP Lite unit (see **Photo 5** and **Photo 6**).

Having made all the right connections, it is time to make all the configuration settings with the Armonía software, which also allows saving product-specific presets. The ArmoníaPlus software is free to be used after registration and there's even a dedicated website (https://armonia.powersoft.it). After registration, developers can obtain a manufacturer's code, which allows to download and install the full Armonía Pro Audio Suite, Armonía ProManager, and activate the licenses.

Powersoft has worked extensively on the complete Armonía Pro Audio Suite, refining interfaces, updating platform support and delivering training tools, both for audio system users and developers. OEM manufacturers are able to use the software to display a system based on the chosen Powersoft amp modules, using the Armonía ProManager dedicated software tool to configure the system, and even initialize products with their own branding information.

Intriguing as it was, we did not fully pursue all of the options, as our goal was to evaluate the completeness of Powersoft's OEM solutions. But after downloading the right drivers for the USB connection, we could quickly find the correct hardware, after which we could configure delay input/output, input equalizer, output equalizer, crossovers, limiters, and lock the parameters. Where other manufacturers offer services for product development, Powersoft goes one step further by offering all the tools needed to develop products, guided by Powersoft's extensive expertise. Yes, very tempting!

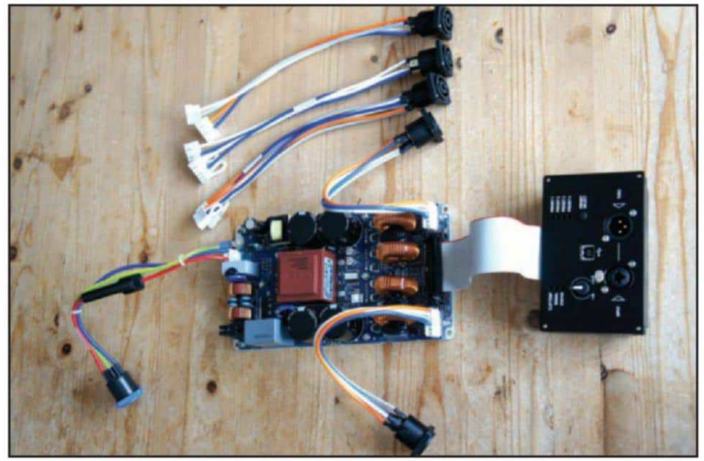


Photo 6: The Powersoft LiteMod 4HC module with all the cabling. Ready-made cabling sets are available for easy manufacturing of OEM products.

Resources

Armonía Plus System Manager, https://armonia.powersoft.it

Powersoft, SpA, www.powersoft-audio.com

S. Yaniger, "Future Shock. Powersoft Audio's Next Generation Green Amplifier Technologies," *audioXpress*, August 2014, www.audioxpress.com/article/r-d-stories-future-shock-powersoft-audio-s-amplifier-technologies

Dayton Audio HTA20BT Hybrid Stereo Tube Amplifier

Hybrid Power for All

What do a 10th century Danish king and two English engineers (one working at the dawn of the 20th century and the other prominent mid-century) all have to do with a hybrid stereo tube amplifier? Enter Dayton Audio's HTA20BT.

Photo 1: Top view of the Dayton Audio HTA20BT hybrid stereo tube amplifier

By
David Logvin
(United States)

Modern-day Bluetooth (BT) technology is named for 10th century Danish King Harald Blatland, famous for unifying Scandinavia (as well as sub-par dental health, with teeth that were notable for a black or blueish hue). Thermionic valve technology (aka vacuum tubes) was patented in 1904 by the English engineer John Fleming. Ubiquitous today, Class-D amplification was first made commercially available in the 1960s by Clive Sinclair of London-based Sinclair Radionics.

All three of these technologies (Bluetooth wireless, vacuum tubes, and digital amplification) are combined in Dayton Audio's HTA20BT, a new hybrid tube amplifier with built-in Bluetooth streaming (see **Photo 1**). This device provides an outstanding (and improbable) feature set that is unlike the offerings from older and more traditional consumer electronics brands.

This integrated amplifier incorporates Bluetooth streaming and is built around a pentode vacuum tube preamplifier with a Class D power amplifier section. Blue LED backlighting complements the soft amber glow of the vacuum tubes, so even the illumination sources are a hybrid. And if the feature set and more than 100 years of technological combinatorial artistry aren't quite enough for you, the Dayton Audio HTA20BT can also do double-duty as a vacuum tube headphone amplifier!

Mechanics and I/O

Physically, this amplifier is quite interesting. The housing is finished in a brushed and black anodized aluminum chassis. A large power transformer sits under an ABS cover. In addition, a pair of 6F1 pentode vacuum tubes (protected by acrylic finger guards) and a 2.4 GHz antenna mast quickly inform the casual observer that this is not your typical audio experience. Passive cooling is provided by ventilation slots on the bottom of the steel base.

Mains power is 120 V, 60 Hz via a non-detachable cord set and the rocker power switch is located on the back panel. The top panel features two well-proportioned knobs—one for gain control and the other for input selection. The knobs are plastic, but they have a nice molded texture and feel. A USB 2.0 port provides an onboard media player and direct local access to MP3, APE, WMA, WAV, and FLAC digital files via a personal USB thumb drive. There are also four small buttons for local digital file or Bluetooth Audio/Video Remote Control Profile (AVRCP) control. The buttons have a good tactile feel, though they are undersized for some hands.

Continuing the 360° "tour de amp," the inputs and outputs are impressive. There are two analog inputs via RCA connectors, as well as an external 2.4 GHz antenna for optimum Bluetooth 4.2 reception.

Bluetooth 4.2 standards specify a 10 m range indoors (and up to 50 m in a free-field environment). The inclusion of the external omni-directional antenna, instead of a compact chip antenna, provides outstanding wireless performance. Bluetooth testing in my home exceeded the 10-m range, going through plaster walls, piping, and more with zero dropouts throughout my testing of this product.

Outputs consist of two pairs of compact binding posts that accept up to 14 AWG wire. There is a subwoofer monophonic RCA connector allowing an elegant option to deploy this in a 2.1 system. Additionally, a 3.5 mm headphone output is also provided, permitting the use of vacuum tubes with your favorite set of headphones.

Media Player Functionality

The inclusion of an onboard USB media player provides an interesting option. By simply adding a USB 2.0 thumb drive, the HTA20BT can be used as a stand-alone audio source and integrated amplifier. This is ideal in areas where Bluetooth isn't consistently available or no analog sources are handy (e.g., in an attic bedroom, office, or small storefront).

The media player options are robust and it is capable of not only playing ubiquitous WAV and MP3 file formats, but also FLAC and OGG, which many budget codec chips don't typically support. While the format support is impressive, especially for those interested in efficient storage of higher quality digital files, the file and the folder management is lacking.

Once you install your USB drive into the HTA20BT, prepare to be "shuffled" back to 2008. This media player is very limited in navigating folder structures. Playback options are brutally simple: play the first track, and then play all other tracks in track order, before moving to the next folder(s). If you use the Pause button, you can restart that track with an additional push. Conversely, if you press Stop, and then Play, then you will be sent back to Track 1 in Folder 1. There is no option to randomize the tracks either, so your playlist is somewhat fossilized (or at least until you go in and manually adjust the tracks using your computer). One further negative, albeit a small one, is that there is no option to use a mobile device as a mass storage device, which would have allowed browsing of files. Since the HTA20BT already has two analog inputs and Bluetooth, this is not a sorely lacked feature.

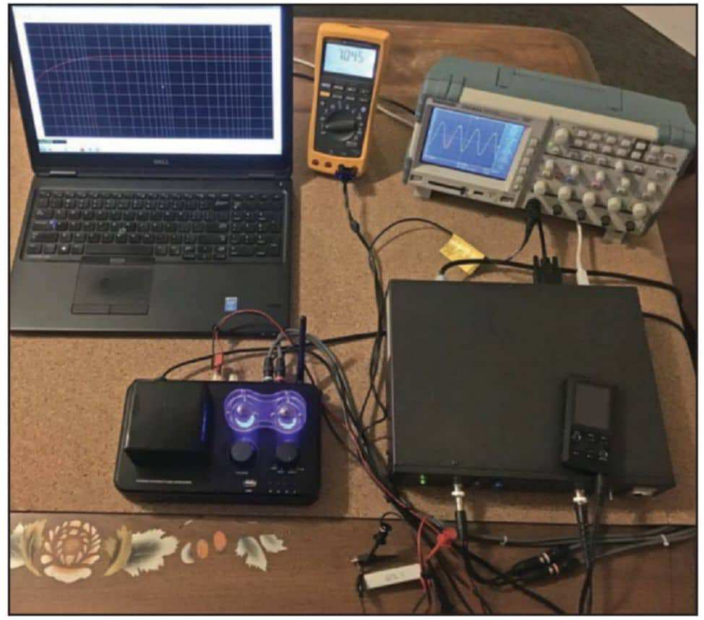


Photo 2: This is my DUT bench setup.

The biggest issue with the media player is the inclusion of the Stop button. It is of very limited use for the media player mode, and has no equivalent mapping when the input is set to Bluetooth. It is disappointing to have a button that doesn't do anything when the Bluetooth mode is selected. This is a small criticism, but one worth mentioning in a device that generally over-achieves in its feature set.

Audio Measurements

I used the following gear to bench test the HTA20BT: the DAAS4 USB (Digital Audio Analysis Software Windows 7 version), Tektronix DSO 2024, Fluke Model 77, and the FiiO X3 DAP (see **Photo 2**).

The preamp section uses a pair of Chinese 6F1

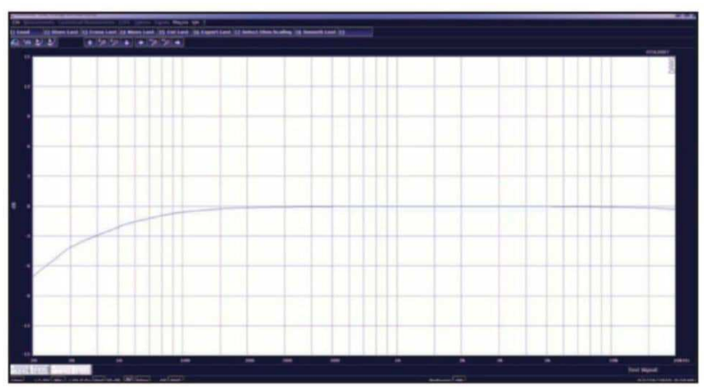


Figure 1: This is the frequency response of the amplifier transfer function.

pentode vacuum tubes, while the Class-D amplifier is built around the STMicroelectronics' amplifier dual BTL chip TDA7491HV.

Dayton Audio lists a frequency response of 20 Hz to 20 kHz, ±2 dB. The datasheet from STMicroelectronics lists a frequency response of 20 Hz to 20 kHz, with a maximum deviation of ±3 dB.

Figure 1 shows a measured frequency plot with a dummy 4 Ω load. Based on the measurement shown in Figure 1, the low-end response is -3 dB at 38 Hz while the high-frequency response exhibits a gentle roll-off of -0.2 dB at 20 kHz. Due to the available power budget of this amplifier, the low-frequency roll-off is perfectly reasonable, and shouldn't prove to be a serious issue when deciding on an appropriate loudspeaker mate. The slight roll-off on the top octave should prove pleasing to most ears during critical listening, while still remaining in the realm of neutrality.

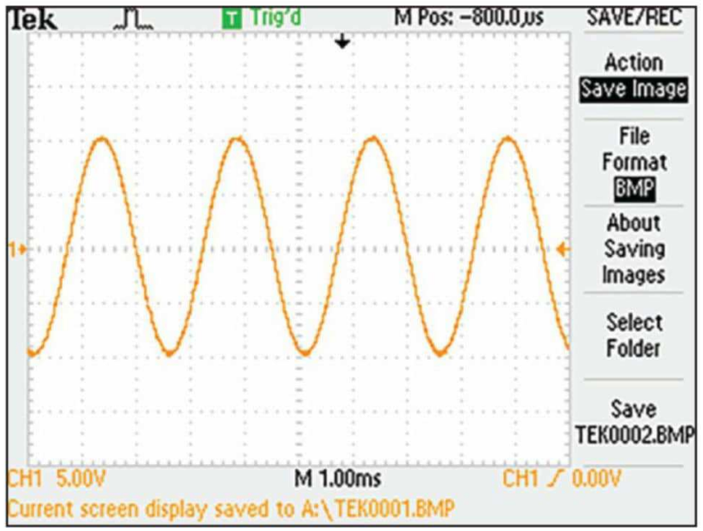


Figure 2: The oscilloscope capture displays a clean 12 W RMS output, with a 400 Hz sine wave driving a 4 Ω load, 25 W dummy load.

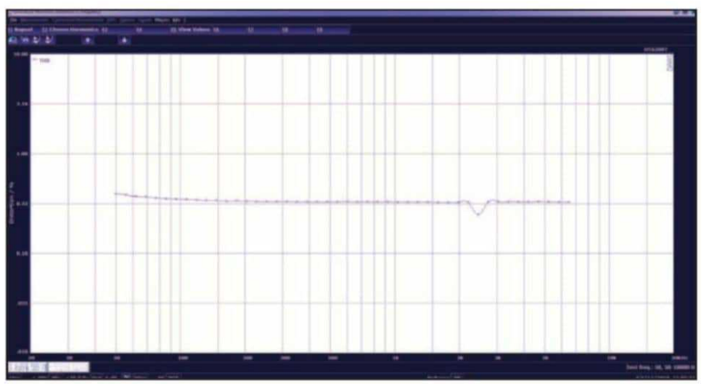


Figure 3: THD vs. Frequency is shown for 12 W per channel into 4 Ω.

Manufacturer data lists the output power as 12 W per channel at 4 Ω with THD of <0.1%. **Figure 2** shows a clean waveform, with a 7 V_{RMS} output, measured via my Tektronix oscilloscope. Each division is 5 V, providing 20.5 V_{PP}.

Given the available output power, I recommend matching this amplifier with speakers with sensitivities on the higher end of the spectrum, and lower impedances in the 4 Ω to 6 Ω range. For my tests, I used full-range 2.5-way towers, with a 4 Ω impedance and a 90 dB SPL sensitivity at 2.85 V_{RMS} at 1 m. This was a good way to take advantage of the strengths of this audio system.

Measurement data for THD vs. Frequency with 12 W per channel into 4 W shows a nominal value just over 0.32% (see **Figure 3**). Inspection of the spectral data (see **Figure 4**) with a 1 kHz fundamental, show that the typical vacuum tube influence on harmonic distortion is almost completely even order, and dominated by the second harmonic. In fact, the third, fourth, and fifth harmonic are of an order of magnitude smaller.

Dropping to a 1 W output, and THD vs Frequency is averaging around 0.17%. With a 1 kHz fundamental, THD is again is dominated by the second harmonic.

As mentioned in the initial inspection, the Dayton Audio HTA20BT is actually a tale of two amplifiers: a 12 W power amp for loudspeakers and a versatile headphone amplifier capable of driving headphones with impedances between 32 Ω and 600 Ω. The headphone amp output is specified at 7.8 mW at 32 Ω with a distortion limit of < 1% THD+N at 1 kHz.

Figure 5 provides the headphone amplifier distortion data for the maximum output with a 32 Ω load. With a 1 kHz fundamental, THD is measuring 0.78%, confirming the manufacturer's listed numbers. The only potential missing feature for this headphone amplifier is the lack of bass boost feature, which would have been a "nice to have." This appears to be an intentional design choice as the preamp stage is similarly missing any tone controls.

Audio Quality

For the wireless portion of my evaluation, I only used downloaded AAC files on my iOS device, to avoid any audio degradation issues due to network traffic. Bluetooth streaming of downloaded AAC files results in a 256 kbps bit rate.

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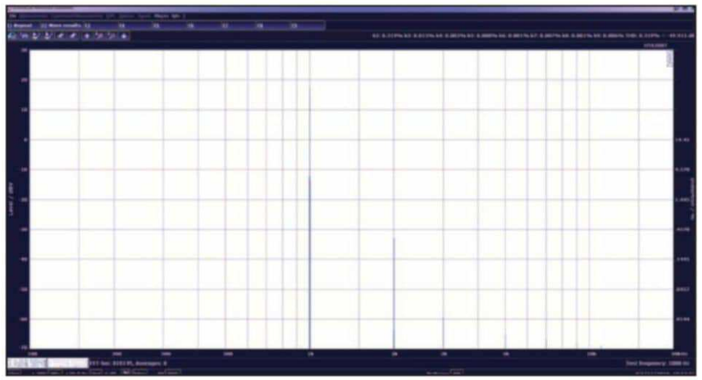

Figure 4: Harmonic distortion with 1 kHz input is dominated by the second harmonic for 12 W per channel into 4 W.

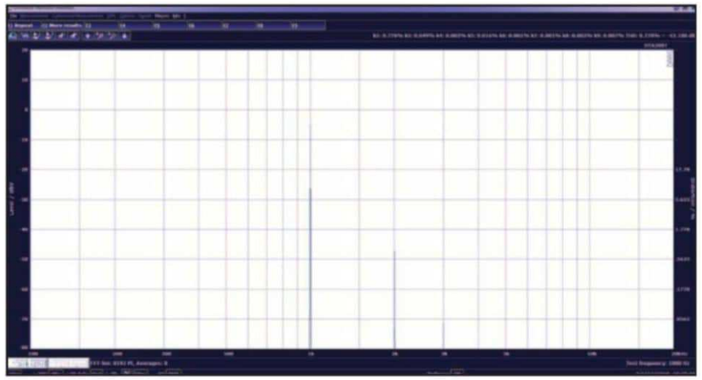

Figure 5: Harmonic distortion with 1 kHz input is 7.8 mW @ 32 W for the headphone amplifier.

Photo 3: The HTA20BT has a retail price of $181.84 USD and is packed with features such as Bluetooth 4.2, five-way binding posts, dual rear-mounted RCA inputs, a subwoofer, and headphone audio outputs, as well as a built-in USB media player.

On Kate Bush's legendary 1978 recording of "Wuthering Heights," her vocals were brilliantly reproduced, with no issues as her voice soared into the stratosphere. But while her vocals retained the purity one would demand, there was a slight smearing that was noted on the keyboard's upper register that slightly detracted from the expected harmonic content.

Next, I streamed a 2014 cover of the David Bowie classic "Ziggy Stardust," by Margot & the Nuclear So & So's. The natural presence and warm sound stage that this system produced was outstanding. This was a demonstration of the alignment where the speakers, source, and electronics all worked in harmony to deliver an engaging experience. Dynamics and output all felt natural, and without any signs of strain or artifacts.

To investigate the analog inputs, I moved on to a more recent recording, Sara Bareilles' "Love Song" (96 kHz FLAC), from the Live at the Variety Playhouse. The warmth of the preamp paid dividends in providing an engaging and pleasing experience. I did find myself wishing the amplifier section had a larger power output, as greater headroom would have enhanced the wide dynamic range that this recording demands for maximum enjoyment. When I listen to live recordings, my optimum experience is always listening at the sound pressure levels (SPL) that the audience would experience during that performance. This is my preference not only when I listen to orchestral music, but with any high-resolution recording such as this one.

Speaking of orchestral tracks, my next test was the San Francisco Symphony's 96 kHz FLAC recording of Bernstein's "America," from its 2014 *West Side Story* album. Dynamics, both micro and macro, were spectacular here. There were no issues dealing with high crest factor events and this too proved to be a good match between all the moving parts involved in reproducing a detailed musical experience.

The HTA20BT handled the many high crest factor events perfectly that are so prevalent on Santana's "Smooth" (256 kbps MP3). The natural character of the preamp stage was the perfect complement to Carlos Santana's guitar riffs.

In order to gain a better understanding of how the HTA20BT digital amplifier section responded, as well as how the tube preamplifier performed in different configurations, I also experimented with several other loudspeakers, both sealed and vented with nominal impedance loads between 4 Ω and 8 Ω. While the degree of influence did vary from speaker to speaker, the sonic signature of this integrated amplifier was consistent and in line with the expectations for vacuum tube electronics in the signal chain.

Presence on vocals was enhanced and sonically pleasing. Midbass and lower frequencies picked up an added warmth that made fundamentals sound more prominent, if not exactly precise. The added character did result in some smearing and loss of clarity in the top octave. This was most apparent on more exposed and open musical passages, particularly on chimes where harmonics and ringing overtones are critical.

I spent considerable time testing the headphone output as well, with my 32 W Grado SR60s as the headphone of choice for this study.

On Paul Simon's 25th anniversary 96 kHz FLAC remix of "Graceland," a solid soundstage was created, with depth and clarity to impress. This is a favorite track of mine due its complex and layered vision, and the experience lived up to expectations. Going further back in time, I also played Gram Parson's 1974 "$1000 Wedding," and on this cut, the positive delta was in the micro details in projecting the lower register of Parson's voice. Since the headphone output had plenty of headroom, there was not a need to push the gain higher and risk any signs of strain to the audio signal.

Final Thoughts

In aggregate, the Dayton Audio HTA20BT offers an intriguing set of features, as well as some thoughtful touches. It has no shortage of inputs, and the inclusion of an external omni-directional antenna ensures an outstanding Bluetooth experience, even in a noisy RF environment. This integrated device is handsomely packaged, with smooth lines and a good tactile feel on the knobs, as well the prominently featured pentode tubes for the preamplifier on the top panel (see **Photo 3**).

This device supports a myriad of varied use cases, with intriguing options for the technically minded users to whom it is squarely targeted. Whether it forms the basis of a headphone setup, 2.0, or 2.1 speaker system, it provides a variety of I/O options and an excellent sound quality offering. The Dayton Audio HTA20BT hybrid stereo tube amplifier is a standout in the world of consumer audio, aligning the most unlikely of product features under one housing.

About the Author

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

Sonarworks True-Fi Software

The Simple EQ to Fix Your Headphones

True-Fi by Sonarworks aims to improve headphone listening on the desktop and smartphone at a price consumers can afford. We test drove the software to discover its strengths and limitations, and we got a peek at what's coming next for Sonarworks.

By
Luke McCready
(United States)

There are a lot of comforts to listening over a good pair of headphones. Room acoustics don't enter the equation. There's no sweet-spot to balance yourself in. Good headphones are certainly cheaper than good speakers. There's no noise pollution for your partner to put up with, and if you're dealing with noise pollution of your own, sealed headphones are always an option.

Photo 1: Sonarworks True-Fi is available on mobile and desktop electronics.

However, headphones can bring a few drawbacks of their own. One of which, certainly, can be the frequency response. As difficult as it is to find a pair of near-flat speakers, it's even harder to find flat headphones. If you're a headphone listener, you probably have a pair or two of cans that generally reflect your listening preferences. But undoubtedly, they still produce more or less bass than you'd prefer; more or less brilliance; and vocals and instruments aren't cast in the same pristine light as from excellent speakers in a well-treated room.

The Software Solution

I've spent the last few weeks with True-Fi by Sonarworks, a software application working to counteract the frequency aberrations in your headphones (see **Photo 1**). And if you're a headphone listener that's yet to test drive True-Fi, I'm writing this for you.

Sonarworks is a Latvian company that specializes in tuning up recording studios, purportedly 30,000 worldwide. The concept is simple: Use a measurement microphone with the Sonarworks Reference software to play test signals through your studio system, identifying frequency aberrations that it can fix with equalization. Your studio's new, custom equalization allows you to track, mix, and master truer to the signal than without. Sonarworks evolved to also correct for studio headphones, and now the company is targeting consumers as well. I interviewed Helmuts Bems, CEO and co-founder of Sonarworks, who filled me in on the story and direction of the company.

"Sonarworks started off in 2012 when two music lovers—Helmuts Bems and Martins Popelis—met scientist Kaspars Sprogis: the perfect storm of acoustic know-how and passion for good sound became the software that's currently transforming sound monitoring. The problem back then: all studios sounded different. The idea was to get free from unwanted coloration and to be consistent across rooms, devices, and users." Bems summarized that Sonarworks has its roots in the studio, but today is correcting and personalizing sound for consumer headphone listeners as well. And in the future, Sonarworks intends to be the "ultimate sound experience on any device."

True-Fi is Sonarworks' consumer- and audiophile-oriented software. In 2017, Sonarworks launched True-Fi on the desktop for Windows and MacOS. The road to mobile support was inevitable, and Sonarworks began Android and iOS development early in 2018. As of this writing, mobile True-Fi is in early-access, expected for a full release in the second half of 2019. A permanent desktop/mobile True-Fi license can be purchased for $79 and can be activated on three devices, and Sonarworks intends to sell True-Fi in the Google and Apple app stores at $99 for lifetime access, or $3.99/month.

The Functionality

True-Fi has a petite interface with few functions. The primary purpose is to select your headphones and correct for them. Adding headphones is easy with its search bar and list of models sorted by brand (see **Photo 2**).

When your headphone is loaded, the frequency response of your headphone is displayed in a graph. Clicking the power button at the bottom of the interface activates True-Fi, applying the frequency correction curve to the sound you hear. It's immediate and can be toggled on and off to hear the difference (see **Photo 3**).

The interface also includes a Personalization section useful for counteracting age-related hearing loss and dialing in the amount of low-end to your taste (see **Photo 4**). Low-end is controlled by a slider from -8 to +8 dB. The high-frequency boost is applied a little differently. You select your age and gender, and the software applies a series of high-frequency shelves, which can be tamed with a slider from 100% down to 0%, with a default at 25%. Raising your selected age increases the effect. Set to female, the software boosts the treble to a lesser degree and beginning at higher frequencies relative to males.

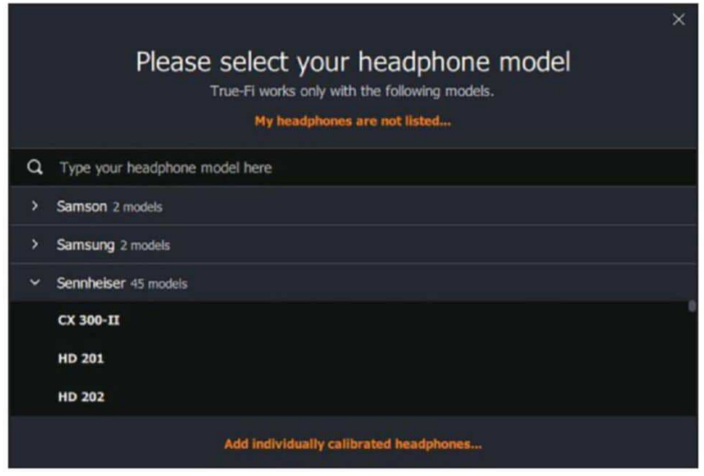

Photo 2: At present, 253 frequency maps of headphones across 71 brands are built into True-Fi.

Support for your headphone model means that Sonarworks tested several different pairs of that same model and averaged the resulting curves to be able to correct your headphones with reasonable accuracy without ever measuring them directly. But if that's not precise enough, you can buy headphones that have been directly measured from the Sonarworks store, or pay $149 plus shipping to have Sonarworks directly measure the headphones you already own. Bems clarified, "No two headphones of the same model are made identical. Usually, the more expensive ones tend to be more consistent, but that's not always the case. We try to cover as [many different headphone models] as possible and still maintain our quality standards. [But] there are

Photo 3: True-Fi can be toggled on and off.

some models that we don't support due to [the] inconsistency being too great."

OS Support

True-Fi makes for a breezy experience on the desktop. The installation is quick, the minimal interface is easy to understand, and system-wide support means that all sounds from all applications are corrected for. The only trick is remembering to disable True-Fi when switching from headphones back to speakers.

Sonarworks was right to see the limitations of a desktop app as most consumers also listen on-the-go from a smartphone. However, smartphone app development brings new challenges. For some devices, sample rates remain limited, but Sonarworks is on the case. "We have plans on supporting all sample rates relevant to music, from 44.1 kHz all the way up to 192 kHz. Platform support of these sample rates varies, and we are constantly working on improving support," said Bems.

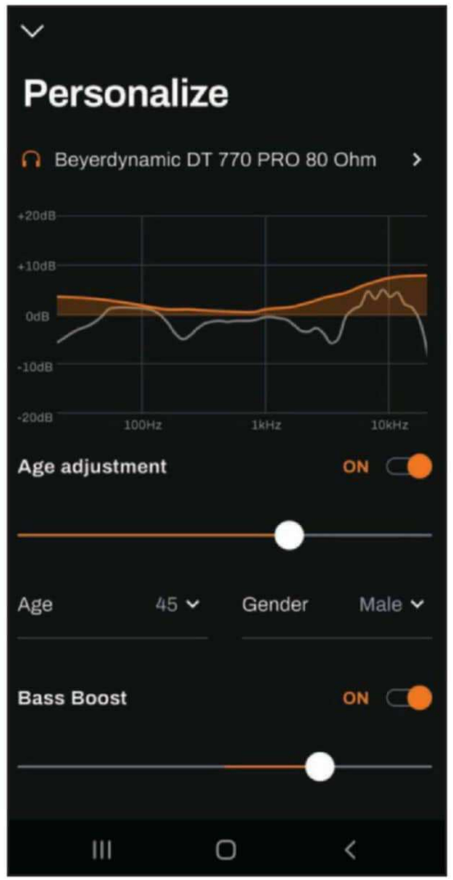

Photo 4: Personalization options are simple yet flexible.

Another challenge is that, at the time of this writing, True-Fi on Android is limited to local files played directly through True-Fi as a media player, and Spotify streaming, not system-wide audio (see **Photo 5**). Bems informed me that "working with the limitations of creating a mobile application, we are planning on supporting additional streaming services and applications. While we keep building the application itself, we are also interested in offering deeper integrations with mobile phone and chipset manufacturers so that our technology becomes available on a system-wide level. This, in contrast, takes more time, so in the near future, we hope that additional streaming services will be welcomed by the community."

Hearing the Difference

I listened on four pairs of over-the-ear headphones, and in all cases, the sound is better with True-Fi enabled. The correction is immediate and generally satisfying. Of course, the difference you hear will depend on the headphones you use, since every model of every brand of headphone will have a different frequency response, requiring different correction (see **Photo 6**).

However, across the headphones I tested, three pairs came with a natural upper-bass bloom around 150 Hz without much low-end to speak of below 80 Hz. Applying the correction gives kick drums and low bass notes clean emphasis and power that's reminiscent of a solid system with good subs.

In all cases, voices sound more centered and

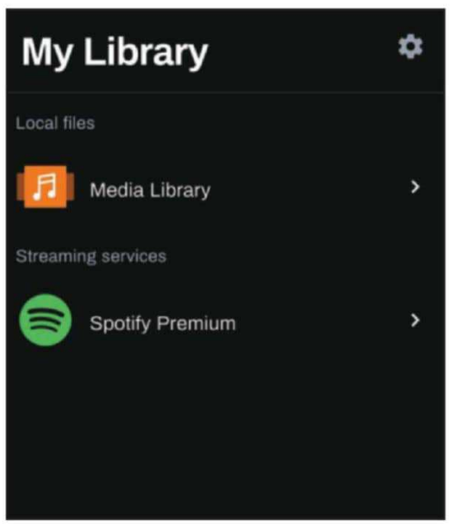

Photo 5: At present, there are limited audio sources on Android.

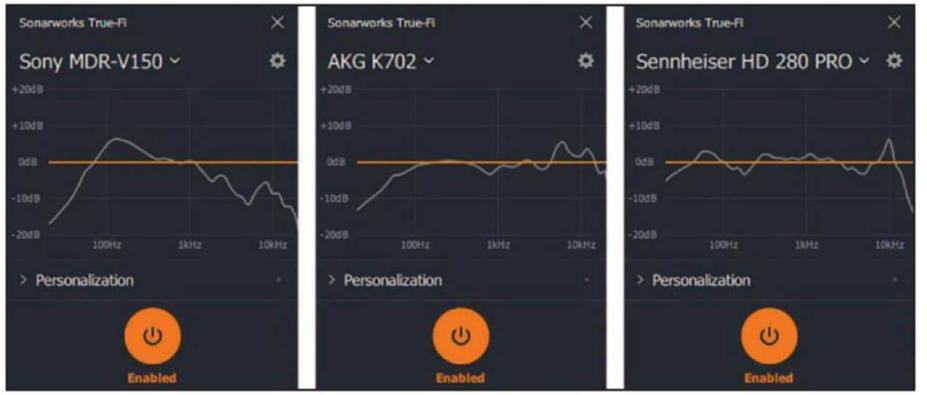

Photo 6: Every pair of headphones requires unique correction.

real, as heard from high-end speakers. Instruments sound truer and more hi-fi with the midrange brought into alignment. And invariably, cymbals just sound different, as though a different engineer mixed them. Looking at the frequency curves of my pairs of headphones in True-Fi, I can see how flawed the high-frequency response is in all four cases: correcting these flaws changes the frequency of sparkle in a song's cymbals, which I find interesting to key into while toggling True-Fi on and off.

How much better does it sound? Sonarworks shipped me the Sony MDR-V150, a small pair of headphones currently $24 on Amazon. Enabling True-Fi with these creates a blinding difference of clarity, warmth, and depth. It's truly startling, even for non-sophisticated listeners.

My favorite cans at home are the AKG K702s, now a relatively affordable $200 on Amazon. The difference here is more subtle, as the headphones already sound sweet and quite nice. But their brilliant clarity isn't reduced as the mids are brought into balance and sub-bass is enhanced enough to be heard. I certainly appreciate True-Fi enabled versus without.

I listened with the personalization features turned

Celebrating 40 Years!
ACOustics begins with ACO™
Measurement Microphones
Polarized and Electret
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off for most of my analysis, to hear the frequency correction for what it is without unfairly skewing the differences with additional low- and high-frequency boosts. Fair comparisons aside, I personally enjoy the age-related high-frequency boost around 15% to 20% engaged, and I find a 2 dB boost to the lows to sound quite pleasing.

Consumer vs. Pro

Above the $79 desktop/mobile price for True-Fi, Sonarworks' desktop-only Reference 4 is priced at $99 for the Headphone Edition, $249 for the Studio Edition that can also correct for speakers with the help of a measurement mic, $299 for the Studio Edition to include a measurement mic, and $699 for the Studio Edition with a mic and also a pair of directly-measured Sennheiser HD 650 headphones. All flavors of Reference come with additional information and customization beyond True-Fi, and a defining feature for professionals is the ability to load Reference into a digital audio workstation in order to hear the correction while mixing or mastering.

"An ongoing challenge has been, for one, to deliver a subset of capabilities of Reference in a much more user-friendly way," Bems said. "This required changing the approach we took: while being used to professional users that need to have detailed feedback about what is happening to their sound, the focus of True-Fi was to hide those details and to make it easy to adjust the most impactful settings."

And the True-Fi development team certainly succeeded (see **Photo 7**). Sonarworks Reference contains a lot more graphed detail to take in, including Correction (the inverse of your headphones' frequency response), Limits (designed to protect drivers from low-frequency excursion), Simulated After (an estimation of frequency response after equalization), Accuracy (margin of error when using averaged headphone results), and Phase (with large differences between the Linear Phase and Zero Latency modes).

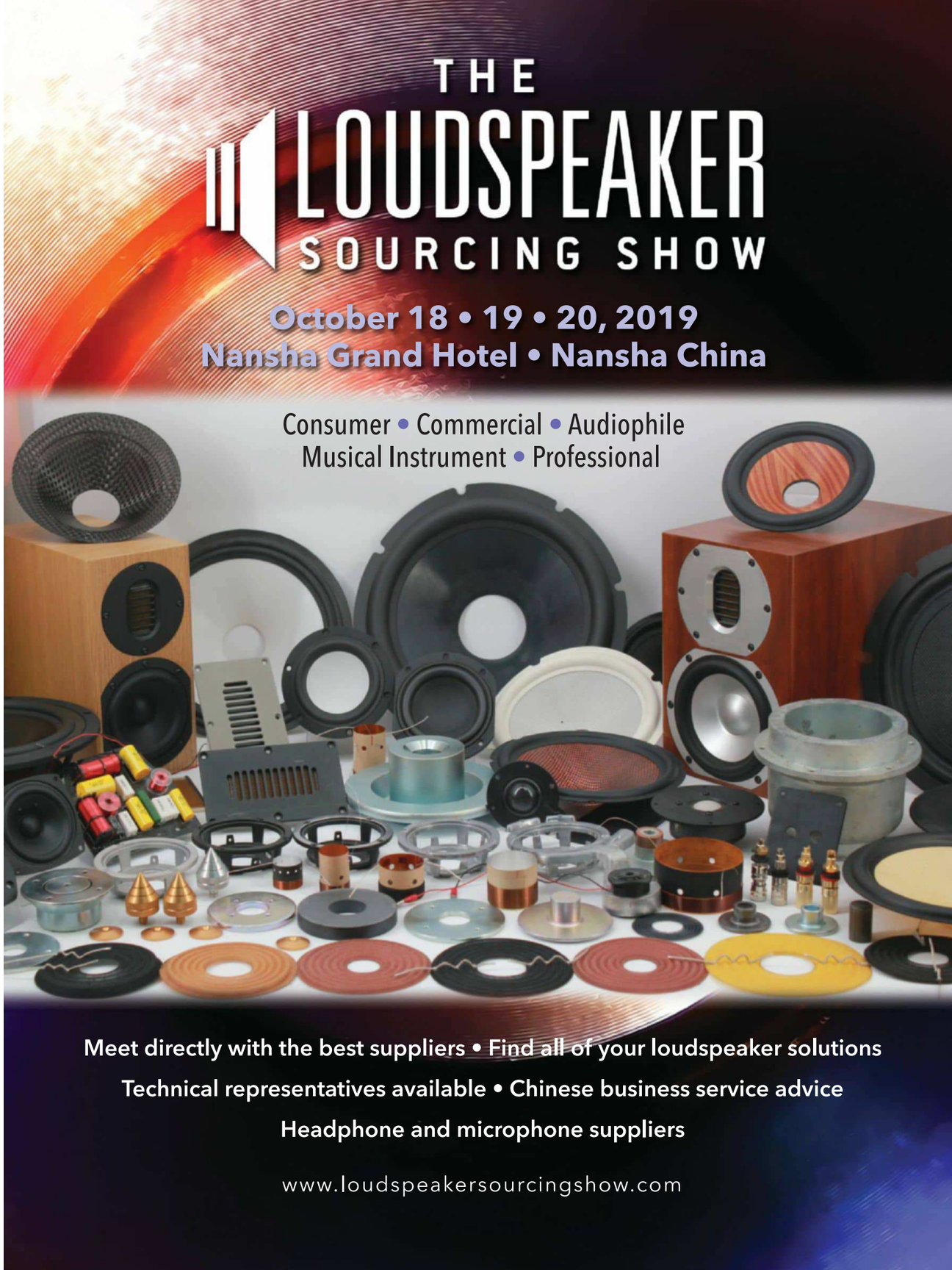
Reference 4 also has customization features beyond True-Fi, including Tilt, a function that slants the target frequency response up to +3 dB at 20 Hz and -3 dB at 20 kHz, or in reverse, brightening the sound as it leans out the bass. The interface also includes the Bruel & Kjaer 1974 hi-fi frequency curve, and the X-Curve, a recommended frequency response for movie theaters. Also present are pre- and post-loudness meters, automatic and manual input volume attenuation to avoid clipping, a toggle for mono listening, and a Dry/Wet knob to dial in the strength of the correction.

I like seeing the many graphs of Reference and tinkering with the different curves, but truthfully, one can get very close to the desired sound with True-Fi's frequency correction and its controls to adjust for bass preference and age.

Shifting Toward Consumers

As pleasing as it is to have 80% of my over-ear

Photo 7: There are many graphs visible in Sonarworks Reference 4.



headphones accounted for with True-Fi correction, only 20% of my in-ear headphones are currently supported by True-Fi: both of these pairs were stock Samsung earbuds that came bundled with my various smartphones over the years (see **Photo 8**). In both of these cases, True-Fi radically alters the sound, but it isn't so much improved as it is differently—yet equally-bad.

There are two immediate problems here: the first is that, at present, consumers are unlikely to find the headphones they already own to be supported by True-Fi. Bems acknowledged this gap: "Up until now, we have been geared toward serving music studios and engineers, and as such, providing profiles for the headphones they typically use. These are radically different than the headphones most listeners will be using, so we're [now] paying more attention to the other side of the equation and have a roadmap of headphones and in-ears in wider use among most music listeners."

Fortunately, Sonarworks has a history of continuing to add new headphone models to its database, and the models added are chosen with intention. "We take into account particular model requests from our existing and potential clients, headphone bestsellers in the biggest online stores, [and] popular headphones among different audiences," Bems added. Sonarworks has a complete list of supported headphones on its website, which I recommend investigating before purchasing True-Fi.

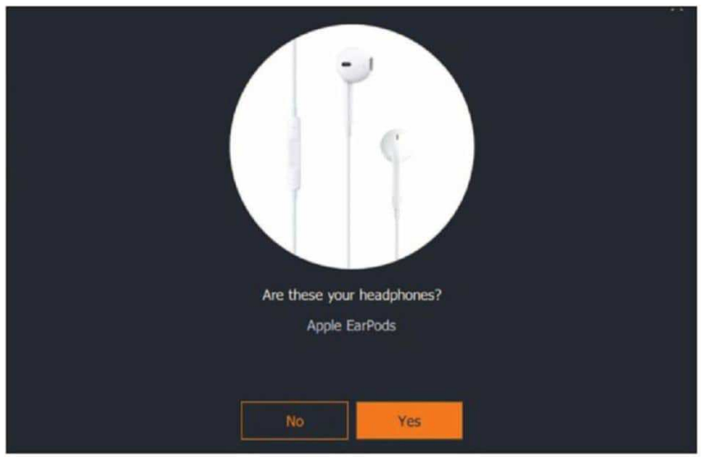


Photo 8: Inexpensive, bundled in-ears come with additional challenges.

The second problem is the ineffectiveness of True-Fi on my two models of Samsung in-ears. I shared my experience with Bems, who said, "We've seen substantial differences in frequency response between the same model in-ear headphones that come bundled with a phone. These kinds of headphones make up a significant part of the whole market, so we try to cover as much as possible and still maintain our quality standards."

But there's more to the story than unit-to-unit variance and invisible design refreshes across the cheapest and highest shipping in-ear headphones. Bems added, "The calibration effect usually is more drastic for in-ear headphones than over-ear headphones, as the curves tend to be more extreme: therefore, [True-Fi] sometimes makes massive cuts or boosts. In-ear headphone frequency response is dependent on the listener's physical properties, and [a poor] seal can lead to diminished calibration accuracy. Also, we've encountered knock-offs of popular in-ear headphones that are visually indistinguishable from the real thing but have very different frequency response: obviously, our calibration isn't accurate in such cases."

The Technical Side

Astute readers might wonder what happens when you boost EQ to signals potentially as loud as 0 dB full-scale. Upon installing the desktop version of True-Fi and enabling it, the software displayed a message that the audio was clipping and gave instructions to pad the volume inside True-Fi, providing headroom for EQ boosts. The problem was easily identified and easily fixed.

The Future of Sonarworks

Beyond adding calibration curves for many more headphone models, Bems offered me a glimpse into the future of Sonarworks: I asked if Sonarworks will ever support phase correction, and Bems said that his team is considering it for the future. He also shared, "We have some exciting new features in the works that should greatly improve users' listening experiences. Some will assist people in finding a target curve they like better without delving into details of audio or equalization themselves. Others are focused on hearing correction tailored for each specific listener."

And perhaps more exciting for those who enjoy a good set of speakers, Bems suggested that Sonarworks is coming to the living room. Whether this would come in the form of integration into receivers or a stand-alone device, he didn't specify. He said that we could expect this "most probably [in] 2020, but we can't promise exact timing."

Conclusion

When it comes to desktop listening, I've rapidly become a contented True-Fi user: it simply makes my headphones sound better. I wouldn't go so far to say that it makes all headphones sound the same—a claim I have heard tossed about. I can still hear more detail and liveliness on my best headphones, and my worst headphones are still a bit fatiguing due to their higher distortion. But even so, each pair sounds better with True-Fi than without.

For mildly enthusiastic headphone listeners, $79 may be a steep price, but such users are likely to listen through headphones that could benefit tremendously. And for those listening on nicer pricier cans, the value proposition makes even more sense relative to the cost of other upgrades.

As we've discussed, Sonarworks is shifting its focus to include audiophiles and consumers, and of course, that means many additional headphones for which to develop calibration curves. This will be the first of Sonarworks' growing pains.

And there's still a lot of room for development for mobile listeners: The tech is solid and sounds great, but the support just isn't there yet. I look forward to Sonarworks integrating with more of my favorite apps and streaming services, but for true ease-of-use, Sonarworks needs to bring system-wide True-Fi to the smartphone. I'm glad to hear that Sonarworks is connecting with smartphone makers and chipset makers for this to happen.

Bems said, "We're targeting a very broad audience: everyone who loves music. We are trying to tackle a very general problem that underpins all music today. Our technology should be a game changer in how casual music listeners, artists' fans, and audiophiles all listen to music, and can do so in the palm of their hands without spending a fortune on expensive equipment."

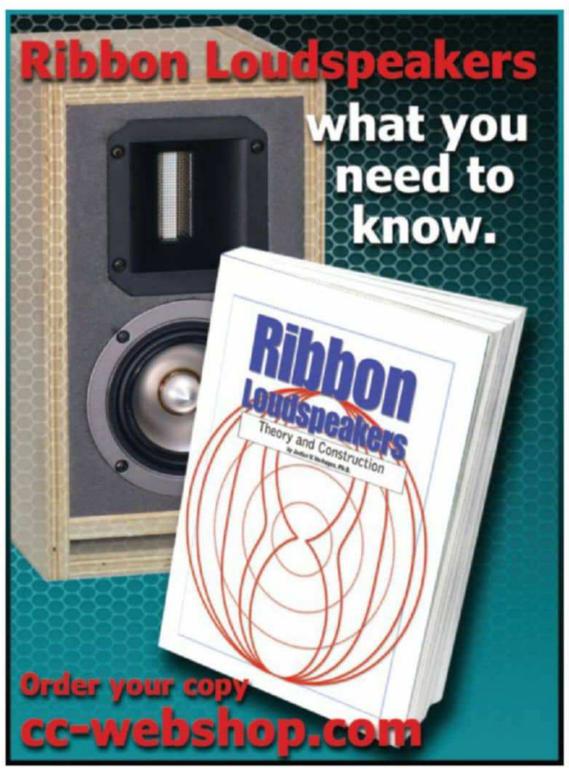
For now, I recommend True-Fi to anyone who takes satisfaction in listening to headphones fed by a computer. Support from the right streaming service can likely make or break a mobile customer. And the future will be bright indeed when every phone and every living room has the potential to sound its best using Sonarworks equalization.

Resources

Sonarworks, www.sonarworks.com

About the Author

Luke McCready is an audio engineer, a technical writer, and a music lover based out of Boulder, CO. He graduated with a degree in audio engineering from North Central University with a passion for audio and a fascination for speaker setup and acoustic treatment.



AoIP Delivers on Flexibility, Is Controllability Next?

The transportation of audio over IP is profoundly flexible, yet the ability to retain controllability over all these IP connected devices remains a frontier for discussion. This article considers approaches to matching the flexibility of AoIP network structure with new technologies and standards for control and monitoring user interfaces.

Photo 1: Markus Schmidt, DeusO GmbH co-founder, discusses the Microsoft Production Studios Cabasa prototype. The diagram outlines device control descriptor concepts and how web-based unified interfaces offer promising gains for controllability in flexible AoIP networks.

By
Matt Hardy
(DeusO GmbH)

Audio over IP (AoIP) has proven to be a transformative technology paradigm. As studios and broadcasters replace previous audio transport interfaces with IP-based solutions, they benefit from the IP network topology. The flexibility introduced through these network structures has inspired countless audio and broadcast professionals worldwide, with some characterizing AoIP as "profound" and "revolutionary."[1]

Got Control? Field Trip to Redmond, WA

Recently, this topic was explored at Microsoft Production Studios. Late in 2016, lead engineer John L. Ball had completed the transition to AoIP utilizing equipment enabled by Audinate's Dante solution. Ball was looking for ways to more effectively wield the flexibility that the Dante solution had delivered. So he approached a likely bunch of companies, members of the Open Control Architecture (OCA) Alliance, asking, "You guys are working on interoperability, how can I get better interoperable control of our AoIP devices?"

Our initial introduction to the studio complex in Redmond, WA made us think about alternative ways to approach control of large networked AoIP systems and devices, utilizing the web browser as a platform to deliver unified control interfaces. In conjunction with Yamaha Corp. and with the Manager of Audio Product Development Division at the time, Hiroshi Hamamatsu, our team formulated a project plan for a fully functional prototype to prove our newly fueled R&D concepts.

This project was code-named Cabasa and a fully functional prototype was eventually delivered over a three-day workshop at the Microsoft campus in Redmond late 2017 (see **Photo 1**).

A number of the underlying features introduced through Cabasa are explored in the following sections. Academic ideas concerning complexity, flexibility and controllability, the importance of standards based technologies as well as UX and user-centered design all help formulate strategies for enhancing interoperability. These aspects are important for commercial imperatives intent on finding additional efficiencies in organized studios or broadcast operations.

Exploring Complexity Theory, Flexibility, and Controllability

There are a number of perspectives on the topic of organizational complexity and flexibility, both with academic and practical interpretation. We start with exploring some academic perspectives, which provide some degree of parallel to the practical applications observed in the field.

This section is not a technical instruction and any concepts referred to here are not concerned with analysis of device performance or best architectural design processes, per se. It is more about how conclusions from an academic perspective support arguments for a new software design paradigm for user interfaces in workflows. The intent, therefore, relates to possible directions in design thinking, in conjunction with AoIP, where specific commercial goals exist to increase workflow flexibility in complex systems without disrupting controllability.

In the 2014 MIT paper, "Flexibility, Complexity, and Control of Large Scale Systems," David Broniatowski and Joel Moses discuss approaches to complexity theories, examining hierarchical tree structures and networks.

Moses outlines how hierarchical control is important to systems architecture because higher elements in a hierarchy have the capacity to control lower elements. He provides a definition of flexibility, where "a system can change its configuration in response to a change in the environment."[2]

Consider the tree structure as illustrated by Broniatowski and Moses in **Figure 1**. Pure tree structures are easy to control due to each node having exactly one parent. Layered hierarchies are defined by their nodes having multiple parents and the possibility of horizontal links. Networks are non-hierarchical and are highly flexible.

A pure tree structure provides clear paths and can be easily broken into sub-trees for purposeful functionality. An example might be an Access Control Model for a sub-branch (as proposed by Alban Gabillon, et al).[3] Where a system demands fine grain access control, there is a degree of attention required on what can be controlled and by which mechanism and through which method. So in this sense, fundamental concepts underlying control depend on hierarchical structures in which elements are ordered according to a relation of superiority and subordination.[2]

When examining the structures and general difficulties described by Moses, parallels appear with the similarities in how current audio device control software is organized into complex AoIP systems. For example, after witnessing the process Ball followed to configure various parts of his equipment, we discovered it involved multiple top level single entry point workflows (see **Figure 2**). The workflows resembled pure tree structures requiring systematic steps into each sub-organization of parts for each device connected to the larger system. This workflow highlighted an inherent focus on control by the existing installed software, leading to inflexibility, a quality of a pure tree structure Moses also describes.

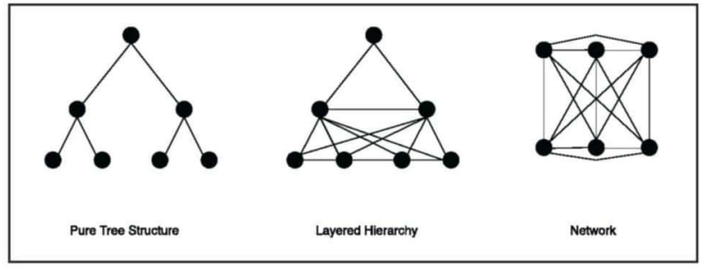


Figure 1: Organizational tree structure as illustrated by David Broniatowski and Joel Moses

It should be noted that to describe installed software as inflexible is not a negative criticism of that software, as it is an inherent design feature for that software's correct and proper operation. An apt heuristic of this dilemma is encapsulated by: "One person's architecture is another person's detail. One person's system is another's component."[4]

As with the larger system, each software had its own architect and each device its own set of components with finer details limiting a specific software's necessary awareness or ability to be interoperable with its peers. Consider an expansion of the problem Ball presented—how can the user control multiple parameters in multiple devices with a single point of entry?

One advantage of networked devices is they are already IP addressable. The potential then exists to expose each device as a tree of functionality (referred to as objects). One possibility of addressing devices, with a suitable control protocol, is that each object in a device could also be addressed and abstracted into an additional user interface, bypassing installed control software altogether. However, once all devices and functionality is openly available, the risk of losing controllability is increased due to the resulting networked topology of fully connected device objects. A network structure quickly becomes so complex, controllability in this paradigm will be reduced.

Moses established that hierarchy is essential to controllability, so a highly flexible but pure network approach will not help. In reference to layered hierarchy structures, Moses says "as the number

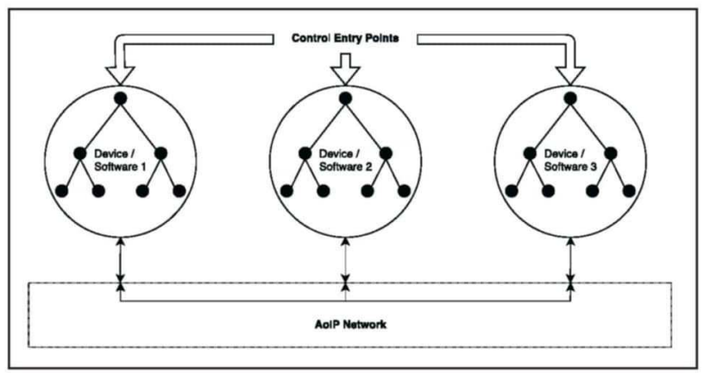


Figure 2: When examining current audio device control software organized into complex AoIP systems, we discovered multiple top-level single-entry-point workflows.

of horizontal links and multi-parent vertical links grow, either in human organizations or engineering systems, the flexibility usually tends to develop while keeping the increase in complexity in check." Key to this statement for practical interpretation is not that a user will necessarily want lateral connections or single points of entry for every node or branch to give them more "flexibility" (see **Note 1**). The requirement here is to understand what it is that a user additionally wants to achieve from a workflow perspective, and whether a system is flexible enough to achieve the stated goals without costing controllability.

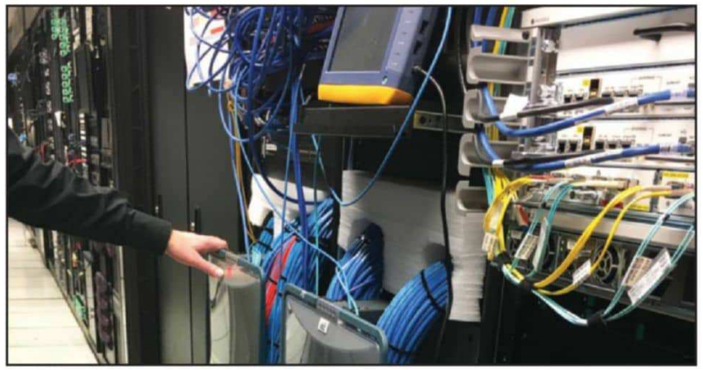


Photo 2: Cisco routers at Microsoft Production Studios power the audio-over-IP installation. In the approach taken for the Cabasa project, it was possible to detach the user's need to think about a network at all. The UI incorporated a filtered list of channels able to be connected for each audio channel available, using a select/connect widget.

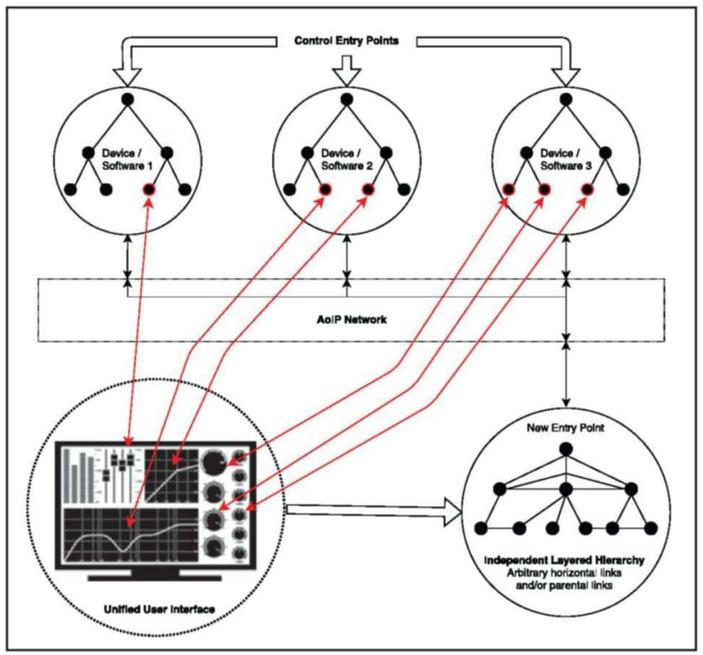


Figure 3: The Cabasa prototype can create a new structure, in parallel to existing control system structures, with the possibility for multiple parents or lateral connections

Workflow Demands Drive Change

Ball provided our team with many critical perspectives and his concerns were not just about saving on operational time. Ball noted that, "at Production Studios, because multiple people access our Audio over IP devices, being able to secure this access to specific functionality from a single workflow UI [web browser] would be a huge help for us."

Case studies exist for why workflow drives the pursuit of flexibility as observed at Redmond, and at other installations. All revolve around the reality of multiple users with differentiation in roles. The differences involve automation options, differing equipment requirements, space or location, security clearance, and it's always possible requirements are found as new projects unfold.

Blair Liikala, media engineer at the North Texas College of Music, revealed in our discussions on this topic that a "problem with web-based UI has been around presets, and managing multiple devices along with keeping things in sync." This observation focuses on how roles and workflows are influenced by user demand and existing control software realities. Blair suggests that choosing a web browser for control is not the central issue, rather it is about how roles and workflows define system boundaries. The web browser platform can support the industry in moving toward a user-centered control software design, but it is how underlying system structures are implemented that remains of higher importance.

Flexibility in a system is key to fulfilling the types of demands encountered (see **Photo 2**). Liikala explained, "For all our production computers I simply have the browsers logged into an account that syncs bookmarks and that bookmark shows up on every machine. When an update for the web-UI comes out, I only have to update the device, not every single computer." With a contrasting perspective, Ball described "regularly receiving calls from talent asking [John] to turn down one of the producers mic gain due to issues with level/voice power."

The common theme for both Ball and Liikala is they want the ability to more easily access a given workflow within their respective network topologies, without the risk of disruption. Also, at any point, a user may need the system to function in ways that the installed control software may not have been designed to do. These points of view are reinforced by simply contrasting user roles (i.e., Admin vs. User or Producer vs. Artist).

Photo 3: This collection of user interfaces illustrates custom UIs of similar collections of functionality. These functional widgets can be arbitrarily combined and recombined, with various web-based user interface examples for plug-in DSPs, instruments, and control surfaces. Arrow links indicate which widgets are unified in alternate UIs, utilizing layered hierarchial structures.

The situations described by Ball and Liikala relate to the same question: Can system control workflows reflect the needs of the user, or is the user forced to adapt to the needs of the system control software?

Building Workflow Hierarchies in Network Structures

Independent layered hierarchies offer a way to meet requirements in flexible ways, and not disrupt controllability, but also do not necessarily require any changes to existing system control software.

The Cabasa prototype approached the workflow problem by building independent layered hierarchies where lateral connections and multiple parents could be bound directly to the objects throughout the network of devices (see **Photo 3**). A key strength of this approach is its ability to maintain hierarchical controllability using hierarchical structures, while delivering great flexibility through the possibility of unlimited lateral connections.

Figure 3 illustrates how the Cabasa prototype creates a new structure, in parallel to existing control system structures, with the possibility for multiple parents or lateral connections. This approach enabled Cabasa to re-establish the primacy of user centered design for control user interfaces and workflow simplification.

In addition, building on a web browser platform resulted in no disruption to the existing control software paradigm, as IP native web browsers exist happily on any modern desktop, laptop, tablet, or phone.

Cabasa demonstrated unified user interfaces. They involved gains, level knobs, faders, connecting Dante transmitters and receiver by channel, device metering, phantom power and EQs all on web-based user interfaces (see **Photo 4**). These interfaces controlled multiple parts of multiple devices, cross-vendor and even over different studios located throughout the 65,000 ft^2 complex (see **Photo 5**).

Road Maps Involved with Interoperable Control

With workflow as a driver of change, the user role forms a central focus, as it is the user (producer, staff, or performer) who commands significant economic value in the context of live productions or other studio activities.

Workflow efficiencies are, therefore, tightly bound to a system's flexibility when the demand on system configurations change. With these various considerations let's look at technical strategies to deliver both flexible control and functionality, which adapt to systems prone to complex configuration changes.

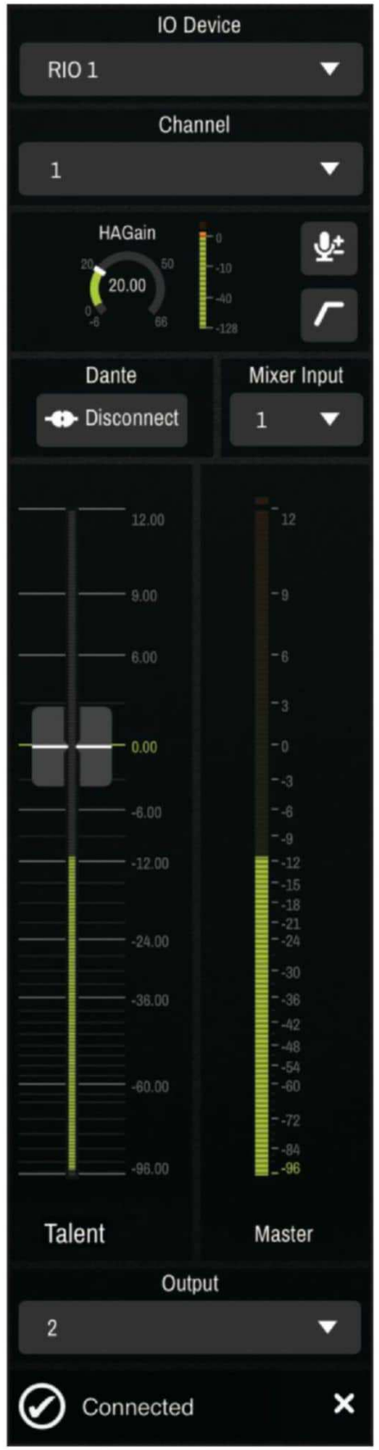


Photo 4: An example of a Yamaha CL-3 console channel UI incorporating a Rio Series gain with a 48v toggle, connecting pre-filtered available Dante channels.

Photo 5: A third-party web browser UI controls all the AES70 objects in a Focusrite MP8R (an eight-channel remote-controlled mic preamp and A-D interface for Dante AoIP networks). It would be possible to unify each strip with additional parameters (e.g., a console channel) and utilize both over Dante/AES67 + AES70.

Web Browsers, Ready When You Are

Web browsers have typically not been considered high-performance software, and perhaps even relegated to a "poor cousin" of the natively compiled C++ and Objective-C applications that users have relied on over the last decade. However, the game has changed.

The web platform is now critical in many sectors, from banking to Citizen-Government communications. Critical to this was better performance, robustness, and continuity. The web browser now offers a reliable platform for future-proofed web technology strategies.

Arguably, standards are what provides the web browser platform with the continuity commercial operations need. Leaders in the standards-based web browser development use open-source code, rarely break backward compatibility and have driven performance to rival natively compiled applications (e.g., Chrome delivered 20× performance increases between 2008 and 2018, see 10 years of speed[5]).

Standards-Based Control Protocol

IP network control protocols are fundamental to building audio device control software applications. From a practical controllability perspective, while devices commonly have some form of control protocol onboard, it's common that not all share one standards-based protocol. A standards-based control protocol delivers continuity and commonality, enabling interoperability to become a natural expectation for workflow architects.

OCA is a standards-based control protocol specifically for media networks. OCA or AES70, as it is also known from the AES standards ratification, offers a modular approach to enabling control for many different types of network devices. Because devices may require user interfaces across multiple control software or vendor installations, device interoperability is seen as an important part of the OCA vision.

Consider the concept of a layered hierarchy described in the Cabasa prototype, each device has an object tree of controllable parameters, OCA/AES70 can be used to expose these as controllable objects over a network topology (see **Photo 5**). Each individual OCA object contains type information and all its functional parameters. For example, a gain might have minimum, maximum, and step information, an EQ an even greater number of controllable parameters. Cabasa could take any device with AES70 support, discover its object tree, then generate a user interface matching each object's set of functionality, thereby providing a user unlimited parameter unification possibilities. High levels of controllability can be retained by the independent layered hierarchy.

The business case of a standards control protocol exists. First, the value of continuity from a standards-based protocol is retained after the initial implementation investment. Second, more interoperable network design possibilities can be realized. Should all devices naturally expose controllable parameters with AES70, integrators and workflow designers can choose OEM workflow, control, and monitoring technologies that do not restrict them to only the specific vendor's software or software platforms.

UX Design

The importance of user experience (UX) in designing workflows is relevant in the vision of layered hierarchy structures. When following a user-centered UX design methodology, users tend to provide qualitative evidence for how they specifically find operational efficiencies. It follows that the flexibility in a control software comes from avoiding rigid structures. How to transition from pure structured control softwares to solutions that resemble layered hierarchies is therefore a value consideration driven by user demands for flexibility and efficiency gains. The value proposition of utilizing both a standards-based control protocol and standards-based user interface platform is a specific ROI in which a UX Design process can assist in developing.

Complexity is Required, Allow Simplicity to Reign

Having considered how complexity theories help underpin the practical problems facing system integrators seeking more flexibility, the primacy of user experience from a control perspective is reinforced. This discussion involved various problems faced by network engineers, differing paradigms, practical examples and possible technological road maps.

Through combining modern web technologies, communication standards and more flexible hierarchy structures, future users of control interfaces may not even realize the scale of complexities underpinning the potential of much simpler customized workflows.

In the case of independent layered hierarchies, it is possible to make workflows appear to the user as a collected simplification of essential functionality, they simply get on with more efficient workflows. Having a user not realize how complex a system is or understand these complexities, is indeed part of the point. Encapsulate the approach to unified user interfaces with another commonly used heuristic, "Simply, simplify, simplify!"

Author Acknowledgements: I would like to thank the following contributors. John L. Ball, Microsoft Production Studios; Blair Liikala, North Texas College of Music; and the support I received from Hiroshi Hamamatsu, Markus Schmidt, Arne Gödeke, Tobias Josefowitz, Jeff Berryman, and Douglas Owens.

Notes

1. David Broniatowski and Joel Moses produce evidence that supports the statement that "flexibility of layered hierarchies comes both from having multiple parents and having several horizontal links; however, adding horizontal links tends to increase flexibility much more quickly than does adding multiple parents. In contrast, only horizontal links greatly disrupt controllability because they necessarily involve the addition of loops."[2] It is the lateral connections which need to be carefully considered when designing an augmentation or transition of an existing pure tree to a layered hierarchy.

About the Author

Matt Hardy is Business Development Manager at DeusO GmbH. He has 20 years experience with web-based technologies and organization development using the web platform. Matt holds a bachelor degree in Philosophy & Management and is currently focused on organizational development and strategy.

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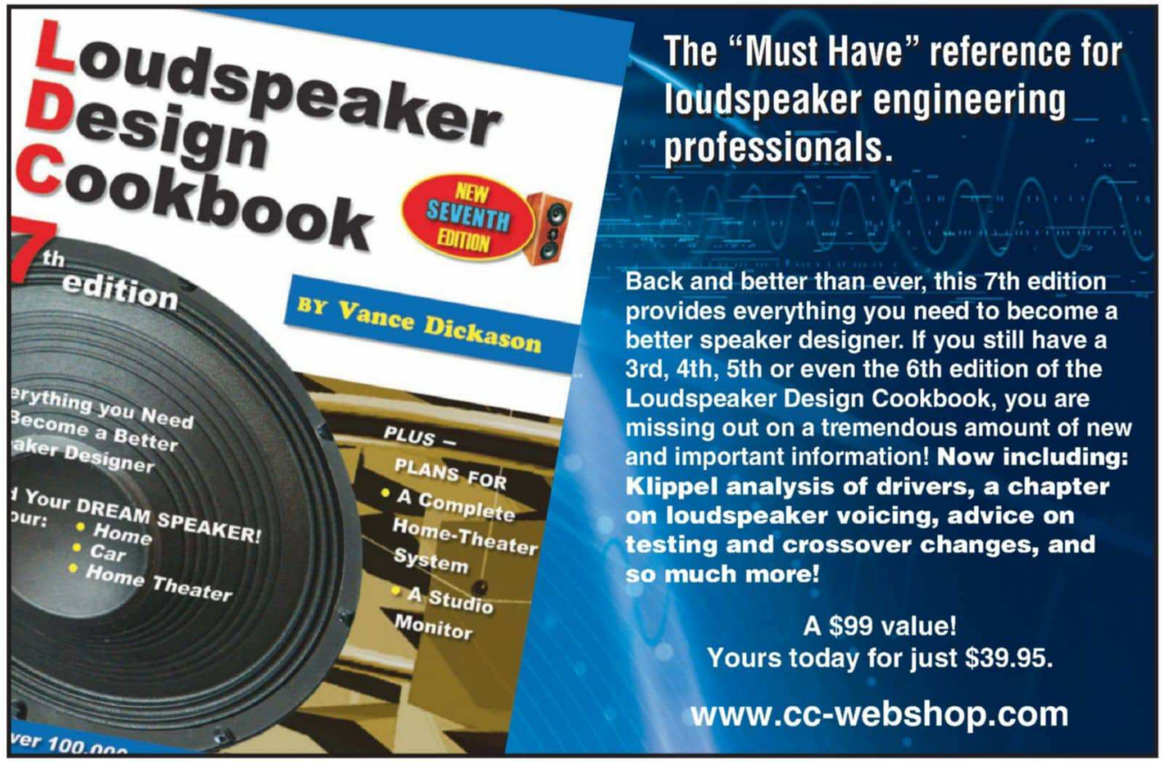
[5] A. Osmani, "10 Years of Speed in Chrome," Chrome Blog, September 2018, https://blog.chromium.org/2018/09/10-years-of-speed-in-chrome_11.html



Buzz Abatement in Audio Amplifiers

This article provides several details about inductive coupling effects in audio amplifiers and the simple way to minimize them.

By
Chris Lewis and
Dimitri Danyuk

System design considerations for audio power amplifiers are well documented. The topology and the distribution of the wiring, partitioning between low power and high power circuits, grounding scheme, types of cables, protection techniques, and safety are important factors that must be considered to minimize the interference effects. While most manufacturers take due precaution with their products to minimize interference, complete immunity is hard to attain. Industrial (mechanical) design, thermal limitations can be named as a possible obstacles to ideal layout, wiring, grounding, and shielding techniques implementation.

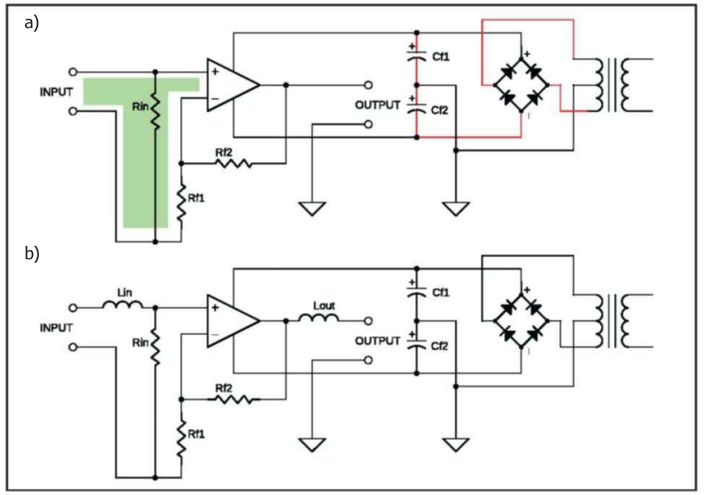


Figure 1a: This is a block diagram of conventional audio amplifier, with the power supply capacitors charging path marked with red and the input loop area marked with green. b: The audio amplifier is shown with compensation coils in input (Lin) or output (Lout) network.

In conventional audio amplifiers, significant alternating current flows through the secondary winding of the power transformer, the rectifier (bridge), and the main reservoir capacitors Cf1 and Cf2. This current path is marked with red in **Figure 1a**. The typical waveform of the charging current is shown in **Figure 2**. The charging current can easily reach several dozen Amperes. It generates a large magnetic field, which couple with other amplifier circuits. The magnetic field is proportional to the current and the cross-sectional area of the loop. Amplifier designers try to minimize the area by twisting wires, keeping the current path short, moving high current conductors away from the amplifier's front end, shielding, and so forth. The current with the wave form shown in Figure 2 also travels through the transformer primary winding and mains wiring (scaled by the transformer turns ratio).

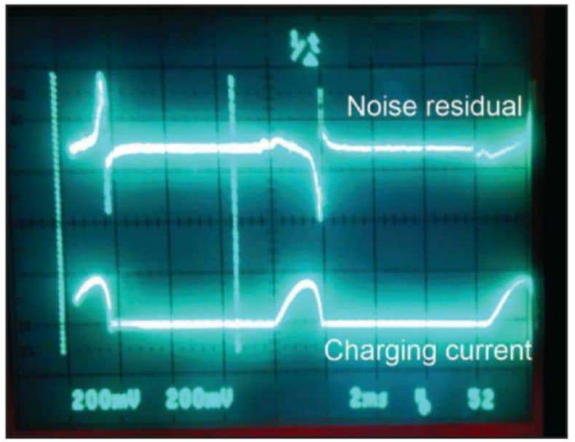


Figure 2: Here is the main capacitors charging current (bottom trace) and the voltage on the amplifier output terminals (input terminals are shorted).

Photo 1: A Mark Levinson amplifier is pictured with an external output compensation coil (yellow wire).

In audio amplifiers, the input circuit is particularly vulnerable to magnetic fields. This area is marked with green in Figure 1a. The front stage, input terminals (signal source), the feedback resistor, and more form a single turn inductor. The magnetic field from the charging current intersects this single turn inductor and causes a current to flow. This loop current produces a voltage drop, which appears in series with the input signal. While the magnetic field from charging current is minimized by design, there still can be some unwanted residual, induced by this current.

A typical waveform of the unwanted signal on the loudspeaker terminals is shown in Figure 2. Inductive reactance is directly proportional to the frequency and the noise residual waveform has the shape of the rate of change of charging current with time. While the residual level is often less than 1 mV, it is rich with high-order harmonics of mains frequency and can be heard from the midrange driver as an annoying buzz. This is especially true for older amplifiers built with through-hole parts, as it is physically impossible to keep the input loop area small.

The practical solution to reduce buzz in that case is to use a compensation coil, which will inject a nearly equal interference signal with opposite polarity in the signal path. This will cause some degree of buzz compensation in the output signal. Two possible places for the compensation coil are shown in Figure 1b. The compensation coil can be inserted in series with the loudspeaker or in series with signal source.

Experimenting with Potential Solutions

The first experiment was done with the output coil, shown in **Photo 1**. The yellow coiled wire is inserted between the speaker wire and the output binding

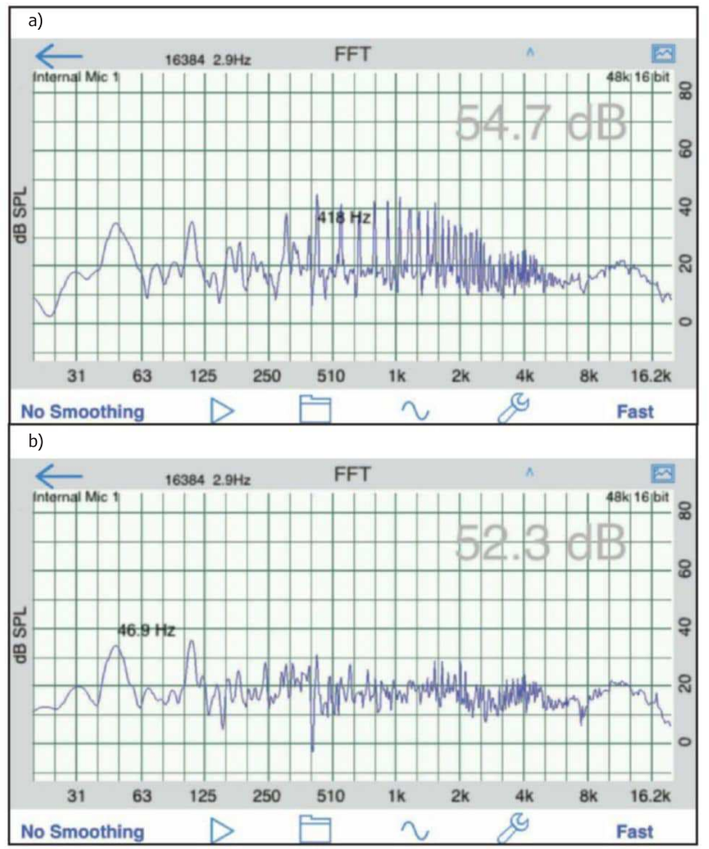


Figure 3a: Frequency spectrum of the noise floor, without coil; and b: the frequency spectrum of the noise floor, with external output compensation coil, shown in Photo 1 (more than 15 dB improvement in the midrange band).

Photo 2: The Mark Levinson amplifier is shown with an internal input compensation coil (magnet wire).

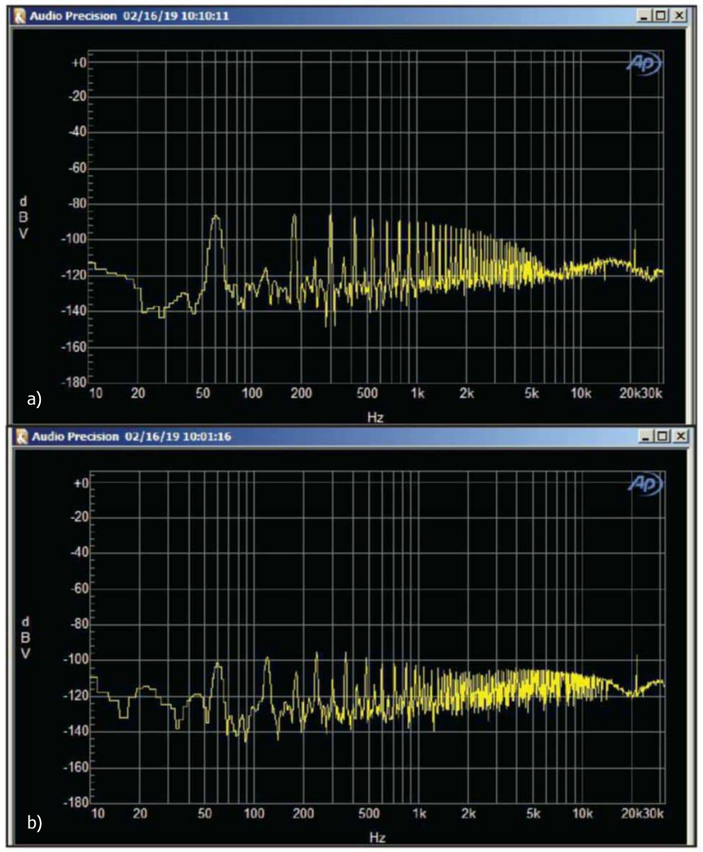


Figure 4a: The frequency spectrum of the amplifier noise floor, without coil; and b: the frequency spectrum of the noise floor, with internal input compensation coil (note more than 12 dB improvement in the midrange band).

post. The number of turns and orientation of the coil was adjusted for best sonic results. Spectrums of the residual without output compensation coil and with it are shown in **Figure 3**. The result is a very audible reduction in the buzz. Most of the harmonics are down 10 dB to 20 dB. As expected turning the coil over increases the buzz two times. Ten turns of wire is a good starting point.

The second experiment was done with the input coil, shown in **Photo 2**. The number of turns and orientation of the coil was adjusted until the best results were achieved. The final coil has 15 turns of magnet wire, wound on a 0.5" former. Coil turns are bonded together with hot melt adhesive. The coil is shunted with a 49.9 Ω CMF55 resistor. The coil was rotated and bent in three dimensions until the best results were achieved. The buzz spectrums with and without an input compensation coil are shown in **Figure 4**. Again, there is solid improvement. It looks like while it attenuates the odd harmonics, the even harmonics are boosted, maybe because the input coil is physically located between several conductors with currents.

When implementing this technique, it is recommended to monitor the noise residual on the speaker terminals with the sensitive scope or by using a spectrum analyzer. It will provide help to find the right orientation of a compensation coil, the coil diameter, and the number of turns.

Resource

H. W. Ott, *Noise Reduction Techniques in Electronic Systems*, 2nd Editon, Wiley-Interscience, New York, 1988.

About the Authors

Chris Lewis started Pyramid Audio Service in 1980 after receiving his BSEE from Texas A&M University in 1978. Located in Austin since 1989, Pyramid Audio is the regional warranty station for McIntosh, B&O, Denon, Marantz, Hegel, Yamaha, and Harman luxury audio. Chris has published more than 130 reviews and articles for various magazines including *AutoSound & Security*, *Autosound 2000*, and *Car Audio & Electronics*. He is a home and car audio fanatic and a specialist in 1970 and newer vintage home audio.

Dimitri Danyuk has been the principal hardware engineer at Harman luxury audio since 2013. He was 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. Dimitri also provides consulting services for various businesses.

You Can DIY!

Sigma 100 (Part 2)

Driving the Satellite/Subwoofer Loudspeaker System

Photo 1: The Sigma 100 loudspeaker system

In the second part of this article series, George Ntanavaras describes two cases for the active crossover that he used to drive the satellites and the subwoofer—one using a mini-DSP 4x10 HD versatile Digital Signal Processor (DSP) and a second one using a pure analog active crossover based on op-amps.

By
George Ntanavaras
(Greece)

The Sigma 100 System is a stereo loudspeaker system consisting of two satellite speakers for the left and the right channel and a subwoofer common for both channels (see **Photo 1**). After constructing the system, I wanted to use an active crossover to drive the satellites and the subwoofer, both digital and analog.

Crossover Design with the miniDSP

The mini-DSP 4x10HD is a versatile multi-channel Digital Signal Processor (DSP) that can function as a digital active crossover, a parametric equalizer for loudspeaker and room correction, a remote controlled volume, and a digital to analog converter (DAC). A block diagram of the system based on the miniDSP 4x10 HD as the active crossover is shown in **Figure 1.** For the setup of the miniDSP, a plug-in running on a PC (or Mac) is needed. After the setup, the PC is no longer needed and can be disconnected. The plug-in is operating at 96 kHz while it's running the algorithms with 56 bits resolution.

I began with the equalization of the subwoofer at the lower frequencies. For the frequencies above the crossover frequency no equalization is needed.

The Linkwitz transform circuit is an extremely useful tool to equalize the low-frequency response of a closed-box loudspeaker. It places a pair of complex zeroes on top of the poles pair of the loudspeaker response (F_o, Q_o) to compensate for their effect. A new pair of poles (F_p, Q_p) can be placed at a lower frequency to obtain a different, more desirable frequency response.

I measured the subwoofer's near-field response and computed the diffraction loss of the subwoofer box with the EDGE program. Combining both, the total response of the subwoofer was similar to a second-order high-pass filter with Q = 0.5 and f = 46 Hz. I used these parameters for the Linkwitz transform, setting the target parameters at Q_P = 0.5 and f_P = 24 Hz. The spreadsheet (see Resources) is a biquad programming spreadsheet containing a module that can calculate the bi-quad coefficients a1, a2, b0, b1, and b2 needed for the Linkwitz transform filter. In the miniDSP plug-in software, I set a filter of the parametric equalizer to the Advanced mode and I entered the five coefficients previously mentioned. The software then calculates the Linkwitz transform and is ready to run.

Additionally, in the crossover setup for the subwoofer, I included a Butterworth first-order high-pass filter at 15 Hz to protect the driver from excessive excursions at the very low frequencies that will increase its distortion. I measured the subwoofer's near-field response, which was -6 dB at 26 Hz as expected. Finally, in the crossover parameters of the miniDSP, I used a low-pass fourth-order Linkwitz-Riley filter at 100 Hz.

I continued with the equalization of the satellite following a similar procedure. The satellite woofer's near-field response, after combining it with the diffraction loss of the box was similar to a second-order high-pass filter with Q_O = 1.25 and f_O = 115 Hz. I used these parameters for the Linkwitz transform of the satellite setting the target parameters at Q_P = 0.71 and f_P = 100 Hz. Additionally, I used in the crossover a second-order high-pass Butterworth filter at 100 Hz so that the total response was identical to a fourth-order high-pass Linkwitz-Riley filter. **Figure 2** shows the voltage as measured at the terminals of the subwoofer and the satellite.

The miniDSP crossover is a convenient solution when a digital source is used. I connect via coaxial S/PDIF the player to the miniDSP and I use its volume control to adjust the level of the reproduction. With this setup, only one DAC is performed at the output of the miniDSP.

With an analog source, the analog inputs of the miniDSP are used. The signal is converted internally to digital so the audio algorithms can run and it is converted back to analog again at the outputs of the miniDSP. I was concerned with this double conversion from analog to digital and vice versa but this can only be avoided if an analog crossover is used. Would it sound better? I decided to find out by designing one that will have an identical frequency response to that of the digital crossover.

The Analog Active Crossover

The block diagram of the Analog Active Crossover is shown in **Figure 3.** From the input buffer, the signal goes to the high-pass and low-pass sections and to a buffered output. The high-pass section includes an equalizer, a Linkwitz transform circuit with a second-order high-pass filter to form a fourth-order Linkwitz-Riley filter. The low-pass section accepts the signal from the other channel, adds both of them, adjusts the level and drives it to a fourth-order Linkwitz-Riley

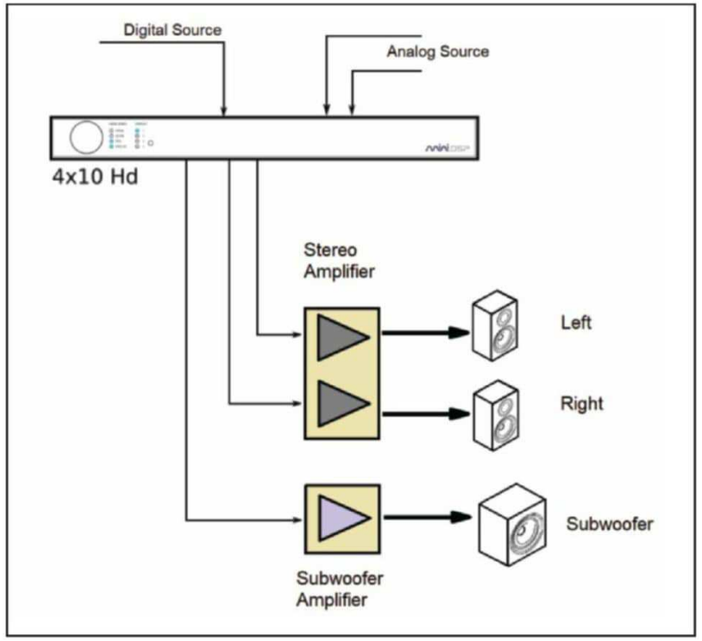


Figure 1: Block diagram of the miniDSP system

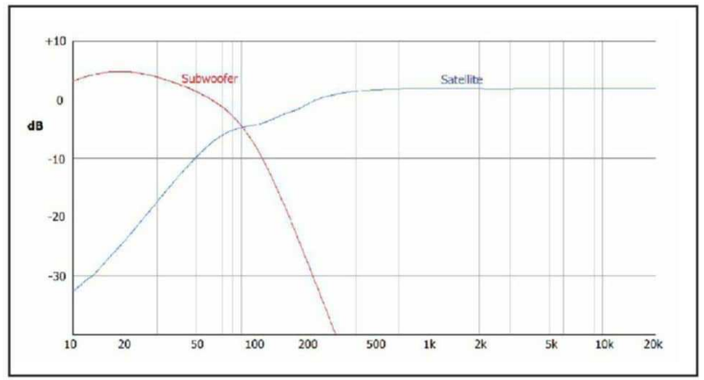


Figure 2: miniDSP crossover response for the satellite and the subwoofer

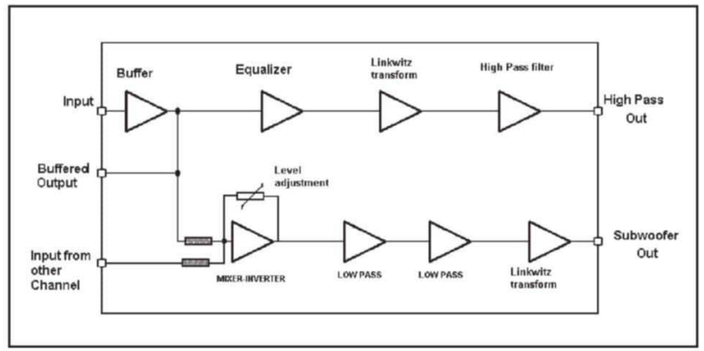


Figure 3: Block diagram of the analog crossover

Resources

ARTA Audio Measurement and Analysis Software: www.artalabs.hr

EDGE program, www.tolvan.com/index.php?page=/edge/edge.php

miniDSP biquad programing spreadsheet, www.minidsp.com/images/fbfiles/files/All_digital_coefs_v1-20101026.zip

Sources

EAGLE PCB Design
Cadsoft | www.cadsoftusa.com

Front Panel Designer
Schaeffer AG | www.schaeffer-ag.de/en/downloads/front_panel_designer

low-pass filter realized with two second-order stages. A Linkwitz transform circuit follows to equalize the subwoofer's low-frequency response.

The complete electronic diagram of the Analog Active Crossover is shown in **Figure 4**. Capacitor C1 blocks any DC voltage from the previous equipment and forms with R5 a high-pass filter to reject the very low frequencies that will force the subwoofer cone to make large excursions.

IC1B buffers the input signal and drives the high-pass section, the low-pass section and the buffered output through components R1 and C40. This output is used when only one subwoofer is used and the Left and the Right channels are added to produce the common low frequencies signal. In this case, R1 and C40 are placed only in one channel. The components around IC1A form an equalizer to attenuate the frequencies below 200 Hz while the next stage around IC3A forms the Linkwitz transform circuit.

The components around IC3B form the second-order high-pass filter, which in addition to the Linkwitz transform circuit form a fourth-order Linkwitz-Riley filter. Resistor R42 buffers the output from any external capacitive load and capacitor C35 blocks any output offset.

The low-pass section drives the signal from the IC1B buffer to the mixer amplifier IC2B, which has two different ways to adjust the gain—by the trimmer RW placed on the PCB or by an external resistor selected by a switch placed on the front panel of the crossover that will be described later.

The output of the mixer drives the first stage of the fourth-order low-pass Linkwitz- Riley filter, which is formed with the components around IC2A, while the second stage is realized with the components around IC4A. The next stage around IC4B forms the Linkwitz transform circuit for the equalization of the subwoofer. Resistor R43 buffers the output and the bipolar capacitor C41 blocks any offset from the signal.

I used a 18 W desk-top switching power supply for both channels of the crossover. It has a 24 VDC output with 5% total regulation and a minimum switching frequency at 24 kHz, outside the audio bandwidth. Through a connector on the back panel of the crossover, the voltage goes to pins V+ and V- of the circuit. F1 is a slow-blow fuse, mounted on the PCB to protect the external power supply from short

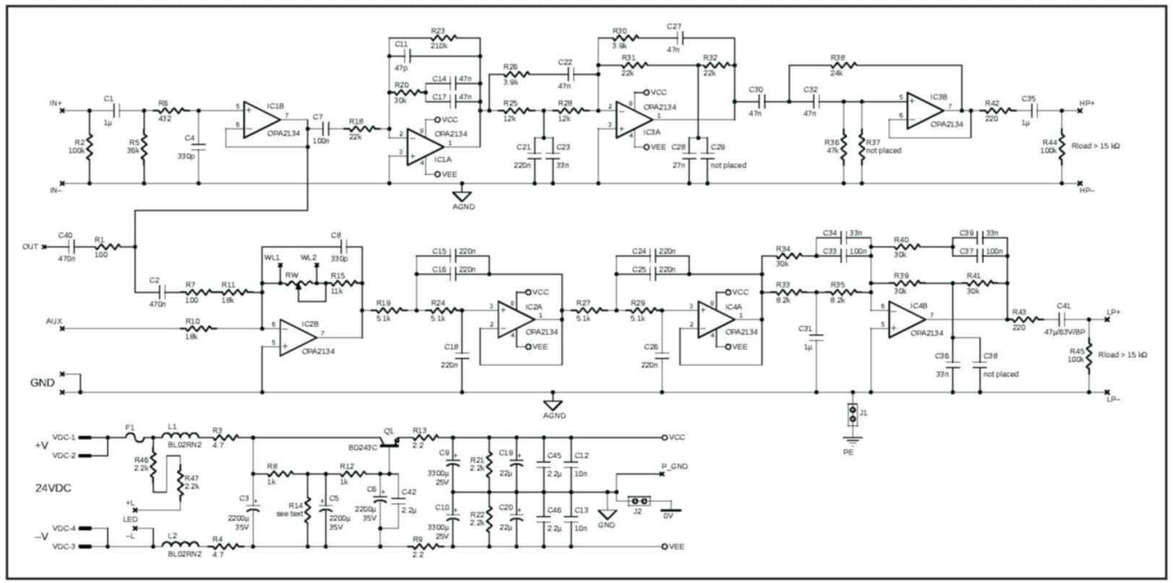


Figure 4: Electronic diagram of the analog active crossover

circuits within the device. L1 and L2 are ferrite beads inductors offering electromagnetic induction (EMI) protection from several megahertz up to gigahertz by suppressing any noise coming from the power supply. Resistors R3 and R4 with capacitor C3 form the first low-pass filter, transistor Q1 with resistors R8, R12 and capacitors C5, C6 form a capacitor multiplier with a very low noise output and R9, R13, C9, and C10 form the second low-pass filter. Resistor R14 is used to set the voltage drop between the collector and the emitter of Q1 at 2 V for the proper operation of the capacitor multiplier. With all four op-amps used in my prototype, R14 was not needed and it was not placed. In the second PCB—when only the two op-amps were placed—R14 was 43 kΩ.

Resistors R21 and R22 are equal and their common point creates the virtual ground of the circuit. Capacitors C19, C20, C12, C13, C45, and C46 decouple the supply voltages improving the stability of the op-amps.

Level Adjustment for the Subwoofer

Trimmer RW is mounted on the PCB and should range from 360 Ω to 30 kΩ to adjust the level of the subwoofer from -5 dB to +6 dB. Usually, for better adaptation of the subwoofer to the room that it is placed, it is preferable to adjust its level with an easier way. For this reason, I designed an accurate level adjustment circuit that can be placed on the front panel of the crossover, which is shown in **Figure 5**.

A 12-position rotary switch selects the proper resistance and connects it to the pins WL1 and WL2 of the PCB. For the 0 dB position, the total value of the resistance that adjusts the total gain of the op-amp IC2B should be 19.2 kΩ. Subtracting the R15 (11 kΩ) which is in series, the resistance should have a value of 8.2 kΩ. All other resistances are calculated so that the gain of the op-amp increases or decreases by 1 dB ranging from -5 dB up to +6 dB. All the resistors were soldered on the contacts of the rotary switch.

A good quality make-before-break (shorting) switch should be used to avoid large excursions of the subwoofer cone during switching that will produce very loud noises. For the initial adjustment with the switch to the 0 dB position, the voltage level at the frequency of 91 Hz should be equal at the subwoofer and satellite terminals.

The Construction of the Analog Crossover

I designed the PCB using the Eagle Layout editor Demo version, which can be downloaded for free. It is a fully operational version, with the only limitation being the maximum dimensions of the PCB. Each PCB contains only one channel, so two PCBs are required for a stereo system.

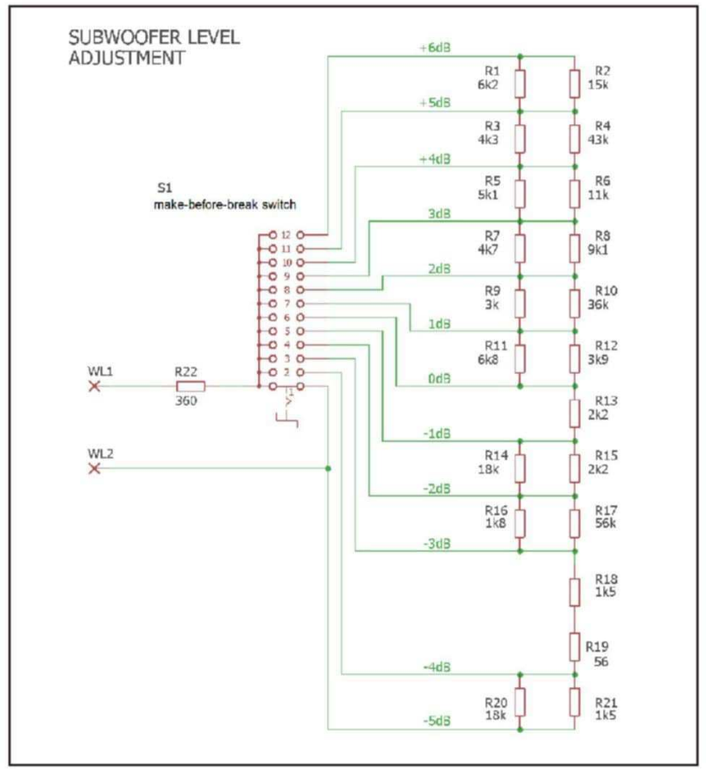


Figure 5: Subwoofer level adjustment circuit

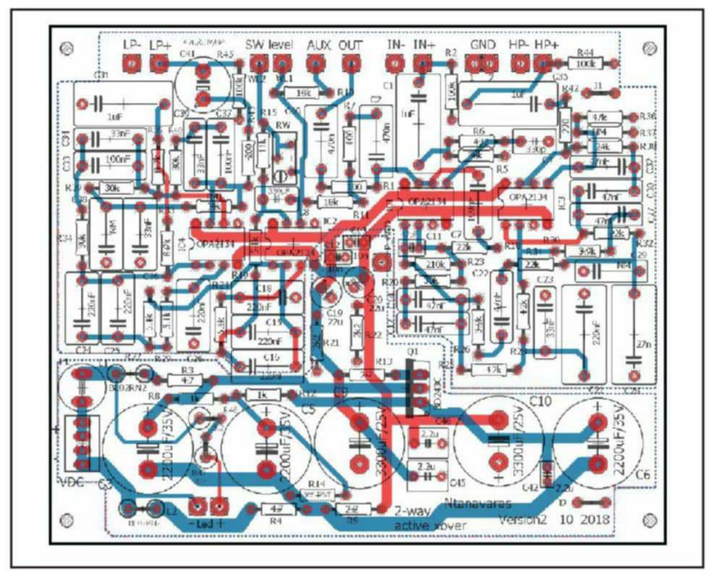


Figure 6: The PCB assembly of the Analog Active crossover

I ordered a high-quality PCB manufactured from FR4 composite material with a 1.6 mm thickness and 35 µm copper solder on both sides and silkscreen on the top side. The placement of the components on the PCB is shown in **Figure 6**. As in every filter construction, I measured all the resistors and all the capacitors. A tolerance of 1% for the components will guarantee that the filter is very close to its theoretical response, which is very important. For the resistors, it is easily succeeded. For the capacitors, I bought a larger quantity from what I needed and I measured the value of each capacitor choosing the ones with a value closer to their nominal. This is a good method to construct higher-accuracy filters without increasing the cost too much.

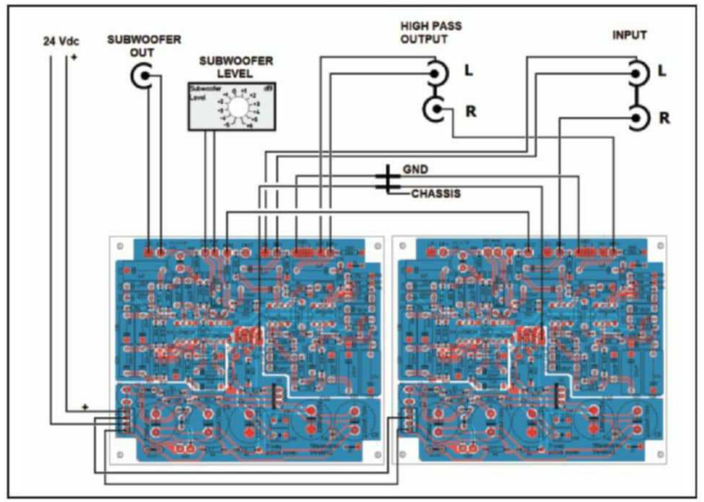


Figure 7: Internal cabling

Photo 2: The front of the analog crossover

Photo 3: The back side of the analog crossover

I separated the signal ground and the power supply ground plane to minimize the noise. They are connected together to a single point placed between the two PCBs and then to the chassis as shown in **Figure 7.** For all the op-amps, I used high-quality IC sockets with gold-plated pins to facilitate the testing or any repairs if necessary.

I housed the crossover in a metal enclosure with aluminum front panel and external dimensions 330 mm wide × 280 mm deep × 50 mm high. **Photo 2** shows the front panel that I created with the Front Panel Designer program. I used my ink jet printer to transfer the file to a transparent self-adhesive sheet.

Two red LEDs at the left side of the panel indicate the operation for each channel. In the middle is the rotary switch, which adjusts the volume level of the subwoofer.

Photo 3 shows the back panel. On the left side five female gold plated RCA sockets, are used for the connection of the input and output signals. **Photo 4** shows an internal view of the crossover while the internal cabling is shown in Figure 7.

On the right PCB, the components required for the low-pass section are not placed, since only one subwoofer is used. The buffered OUT of the PCB is connected to the AUX input of the other PCB.

The Testing of the Analog Crossover

First, without the op-amps placed, I tested that the power supply voltages were close to ±11 V. With the op-amps inserted, I measured +10.71 V and -10.74 V in my prototype. The voltage drop between the collector and the emitter of Q1 should be 2 V otherwise R14 is used as described.

The frequency response of the low-pass and high-pass filters was checked, according to the following values at high-pass output:

1 kHz	0 dB
200 Hz	-3.1 dB
100 Hz	-6.4 dB
75 Hz	-7.5 dB
33 Hz	-17.7 dB

And at the low-pass output:

200 Hz	-30.4 dB
106 Hz	-12.2 dB
75 Hz	-6.7 dB
44 Hz	-2.6 dB
20 Hz	0 dB
10 Hz	-1.6 dB

The crossover should be turned on prior to the power amplifier because of a small thump that is produced at its outputs. Also a time period of about 30 s is needed to be fully operational due to the large capacitors and the resistors in the power supply's capacitor multiplier.

It can drive all its outputs with a maximum voltage of 6.6 V_{RMS} (±9.3 V peak), which corresponds to a maximum 2 V_{RMS} at its inputs when both channels are driving the low-pass output.

Conclusion

The Sigma-100 is a compact system easily integrated into any room. It is a good sounding system with a well-balanced frequency response and excellent bass extension. I have listened to both crossovers for a long time and I am very satisfied with their performance. The digital crossover is more versatile and can be used to equalize the room response which offers a big improvement to the reproduction. The analog crossover is cheaper but any changes are more difficult to be implemented. And many of audio enthusiasts believe that the analog sound is superior.

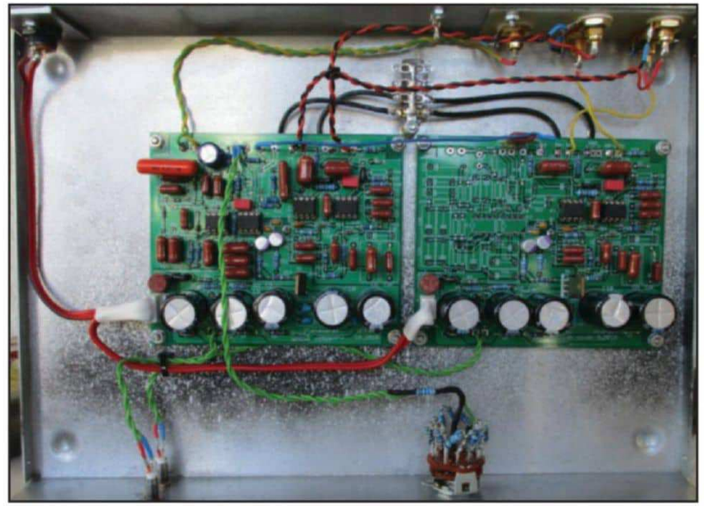


Photo 4: The internal of the analog crossover

About the Author

George Ntanavaras graduated from the National Technical University in Athens, Greece, in 1986 with a degree in Electronic Engineering. He currently works as a Technical Consultant for Products Development and Manufacturing. He is interested in the design of preamplifiers, active crossovers, power amplifiers, and most loudspeakers. He also enjoys listening to classical music.

Analog Crossover For One channel

Part	Value	Notes
C1, C31, C35	1 µF, 100 V, 10%	Polyester
C2, C40	470 nF, 250 V, 5%	Polyester
C3, C5, C6	2200 µF, 35 V	Electrolytic
C4, C8	330 pF	Polypropylene
C7, C33, C37	100 nF, 250 V, 10%	Polyester
C9, C10	3300 µF, 25 V	Electrolytic
C11	47 pF	Ceramic, COG
C12, C13	10 nF, 50 V	Multi-layer Ceramic
C14, C17, C27,C30, C32, C22	47 nF, 250 V, 10%	Polyester
C15, C16, C18, C21, C24, C25, C26	220 nF, 250 V, 10%	Polyester
C19, C20	22 µF, 35 V	Electrolytic
C23, C34, C36, C39	33 nF, 250 V, 5%	Polyester
C28	27 nF, 250 V, 10%	Polyester
C29, C38	NOT PLACED	
C41	47 µF, 50 V	Bipolar
C42, C45, C46	2.2 µF, 50 V	MLCC Ceramic
F1	100 mA Fuse with Time delay	For PCB mounting
IC1, IC2, IC3, IC4	OPA2134	Op-amp
L1, L2	BL02RN2	Ferrite Bead
Q1	BD243C	transistor
R1, R7	100 Ω	1%, metal film, 0.25 W
R2, R44, R45	100 kΩ	1%, metal film, 0.25 W
R3, R4	4,7 Ω	1%, metal film, 0.25 W
R5	36 kΩ	1%, metal film, 0.25 W
R6	432 Ω	1%, metal film, 0.25 W
R8, R12	1 kΩ	1%, metal film, 0.25 W
R9, R13	2.2 Ω	1%, metal film, 0.25 W
R10, R11	18 kΩ	1%, metal film, 0.25 W
R14	see text	1%, metal film, 0.25 W
R15	11 k Ω	1%, metal film, 0.25 W
R18, R31, R32	22 kΩ	1%, metal film, 0.25 W
R25, R28	12 kΩ	1%, metal film, 0.25 W
R19, R24, R27, R29	5.1 kΩ	1%, metal film, 0.25 W
R20, R34, R39, R40, R41	30 kΩ	1%, metal film, 0.25 W
R21, R22, R46, R47	2.2 kΩ	1%, metal film, 0.25 W
R23	210 kΩ	1%, metal film, 0.25 W
R26, R30	3.9 kΩ	1%, metal film, 0.25 W
R33, R35	8.2 kΩ	1%, metal film, 0.25 W
R36	47 kΩ	1%, metal film, 0.25 W
R37	NOT PLACED	
R38	24 kΩ	1%, metal film, 0.25 W
R42, R43	221 Ω	1%, metal film, 0.25 W
RW	see text	Subwoofer Level

Subwoofer Level Adjustment circuit

Part	Value	Notes
R1	6.2 kΩ	1%, metal film, 0.25 W
R2	15 kΩ	1%, metal film, 0.25 W
R3	4.3 kΩ	1%, metal film, 0.25 W
R4	43 kΩ	1%, metal film, 0.25 W
R5	5.1 kΩ	1%, metal film, 0.25 W
R6	11 kΩ	1%, metal film, 0.25 W
R7	4.7 kΩ	1%, metal film, 0.25 W
R8	9.1 kΩ	1%, metal film, 0.25 W
R9	3 kΩ	1%, metal film, 0.25 W
R10	36 kΩ	1%, metal film, 0.25 W
R11	6.8 kΩ	1%, metal film, 0.25 W
R12	3.9 kΩ	1%, metal film, 0.25 W
R13, R15	2.2 kΩ	1%, metal film, 0.25 W
R14, R20	18 kΩ	1%, metal film, 0.25 W
R16	1.8 kΩ	1%, metal film, 0.25 W
R17	56 kΩ	1%, metal film, 0.25 W
R18, R21	1.5 kΩ	1%, metal film, 0.25 W
R19	56 Ω	1%, metal film, 0.25 W
R22	360 Ω	1%, metal film, 0.25 W
S1	Rotary Switch	1 pole, 12 position, Make-before-break

Industry Calendar

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

July 18–20, 2019
Summer NAMM
Music City Center, Nashville, TN
www.namm.org/summer/2019

Every year, the National Association of Music Merchants (NAMM) promotes its Summer NAMM event, which is traditionally very strong for the music instrument communities. With the recent expansion of the Winter NAMM also toward the professional audio market and stage technologies, more companies from live sound and recording segments will attend as well. At Summer NAMM, attendees can develop relationships with the more than 500 exhibitors; dive into product demonstrations from more than 1,600 brands; find unique products; experience the new Retail Training Summit with six tracks for all areas of a company's retail team; learn best practices during NAMM U sessions, TEC Tracks, and the Top 100 Dealer Awards; and discover the future of audio with A3E and Dante Certification Training. Visitors can also network with industry leaders and peers during receptions, concerts, and explore Nashville.

July 26-28, 2019
California Audio Show
Hilton Oakland Airport (Building 5), Oakland, CA
www.caaudioshow.com

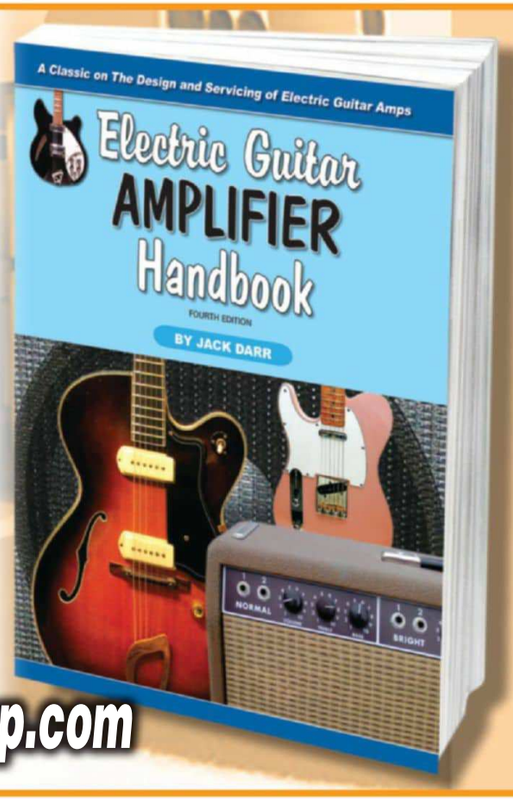
The California Audio Show (CAS) is a HiFi show for music lovers, this year happening during three full days at the Hilton Oakland Airport (Building 5) in Oakland, CA. The promoters have arranged for up to 44 specially furnished Sound Rooms for auditioning high-end audio systems and there will be a Headmasters section, a mini show of high-end headphones and accessories inside the CAS. Record vendors will also attend to offer a one-stop experience in vinyl shopping, and the promoters have arranged a series of seminars from industry leaders, manufacturers, and reviewers discussing all things audio. CAS is founded and produced by an audiophile who shares the passion of stereophonic playback with other music lovers in the San Francisco Bay Area. Response from exhibitors and attendees in previous editions was overwhelmingly positive with very good attendance.

July 27–28, 2019
CanJam London
Park Plaza Westminster Bridge, London, UK
https://canjamglobal.com/shows/london2019

CanJam returns to the heart of Central London at the ideally located Park Plaza Westminster Bridge Hotel. Of all the CanJam events promoted by Head-fi, the London edition is one of the fastest growing, even if it's not the largest one. As the only European location for CanJam, the show has proven to be extremely dynamic and interesting, featuring more than 100 leading brands in more than 20,000 ft² of exhibition space. The Park Plaza Westminster Bridge's location in London's South Bank is a short walk to some of the city's most iconic attractions. The Waterloo and Westminster tube stations are a five-minute walk away.

Are you interested in learning how guitar amps work, as well as how to fix and service an amp?

Then this book is for you.



Jack Darr's *Electric Guitar Amplifier Handbook* details the following:

- How guitar amplifiers work
- How to properly and safely make amp repairs
- How to troubleshoot tube and transistor amps of all sizes
- Details on "typical" amp circuits

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with Dual Passive Radiators
and Built-in DSP

Epique CBT24K
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