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Things Getting Better

I just returned from three very rewarding days at the High End show in Munich. Promoted by the High End Society, it is currently the best consumer audio show in the world. The 2017 edition was also the strongest and it enabled me to see and listen to some very interesting new products. One of the most interesting ones, as highlighted in our online coverage, was undoubtedly the LiveBox fully-integrated Crosstalk Cancellation speaker, designed by Swiss companies Weiss Engineering and PSI Audio. Designed to be like "a piece of furniture," the LiveBox allows both convincing stereo and binaural experiences from a "single source" speaker. I recommend you read my report at audioXpress.com, together with other High End 2017 highlights (http://www.audioxpress.com/tags/high-end-2017).

I must also mention this product, first because it is a perfect segue to my previous editorial in *audioXpress* June, discussing binaural experiences. Second, because it is a good example of how new audio products can leverage all the possibilities of modern digital processing technology and electro-acoustic design to create surprisingly good experiences.

I must confess that I have been finding many other examples of products that truly have impressed me, and not necessarily in the price point of the LiveBox (which will be selling upward of USD $20,000). Yes, high-end shows have allowed me to find fine examples of pricey headphones, speakers, amplifiers, and audio source components that provide convincing experiences. But I have been looking particularly for examples of affordable products that impress me just as much, that are technologically new, and that I could (want to) take home. And, I must confess that some of the most memorable experiences I've been having from trade shows involve wireless speakers and soundbars.

Wireless speakers by RIVA Audio (Wand Series), Sonos (PLAY:5), Bluesound, and more recently the new Electrocompaniet EC Living Tana are examples of "affordable" new-generation wireless speakers that deserve a mention and serious consideration. And most of those companies also design soundbars, although soundbars are frequently scorned by some, they just keep getting better and more impressive. All is derived from a combination of better drivers and speaker design, cleverly tuned advanced digital processing, better amplifiers and converter stages, and smart and convenient connectivity. Soundbars also show a perfect understanding of the average consumer's needs.

Another example of a product that impressed me is reviewed in this edition, and is the new Benchmark DAC3 HGC D/A converter, which not surprisingly features the new ESS Technology ES9028PRO converter chip, another of the reasons why many solutions are sounding better. I have been traveling with the latest OPPO Digital HA-2SE portable headphone amplifier and DAC, also updated with the ES9028Q2M—the top-of-the-line converter for portable designs from ESS Technology—and I've been enjoying listening to headphones even more.

Of note, amplifiers – and, in particular, the Class-D kind—are also improving every year. Companies are pushing the envelope on Class-D and we can find examples of superb products now reaching the market that are effectively improving audio experiences for everyone, in all sorts of applications, from portable and personal audio to professional sound reinforcement. And the more we look into the topic—which we continue to explore in our Audio Amplifier series—we discover new companies and examples of products that prove things just are getting better.

No doubt, the more we look at new high-quality amplifier designs, the more we have to admire what Benchmark Media Systems did with its AHB2 amplifier, using the THX-patented AAA technology, which Gary Galo reviewed back in 2015 (www.audioxpress.com/article/Fresh-from-the-Bench-Benchmark-AHB2-Stereo-Power-Amplifier). The AHB2 continues to be a true benchmark!

João Martins
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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.

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.

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.

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.

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957, he started Testronic Development Laboratory, which became TDL Technology, to develop audio electronics. All product sales and services were terminated on December 31, 2015, but the TDL website is still online with a variety of audio information and downloads.



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Loudness Revisited (Part 2)

Varying Methods of Loudness Measurement

Now that media streaming has become a preferred way of listening to music, we all want to avoid the need to constantly adjust the volume control. Loudness normalization—the attempt to avoid having to adjust the volume—has become desirable, if not a true necessity. But it isn't that easy a problem to solve.

By
Ron Tipton
(United States)

There are several free and non-free loudness measuring programs for Windows and Mac. Most of them compute and display the True Peak value. All those I have looked at display the absolute loudness numbers and some also display the relative numbers. Let's begin by defining true peak and absolute vs. relative loudness.

True Peak

Figure 1 shows a portion of an analog waveform as the smooth blue line with digital sampling points represented by the large dots. The digital peak levels are the two green lines but they don't indicate the true peak, which extends to the red lines. It's important to know the true peak because clipping distortion might result when the digital samples are converted back to analog to drive our amplifiers and loudspeakers.

International law requires true peak compliance if the content is to be broadcast by radio or TV. Streaming services are slowly being asked to comply. More information on this subject is included in the

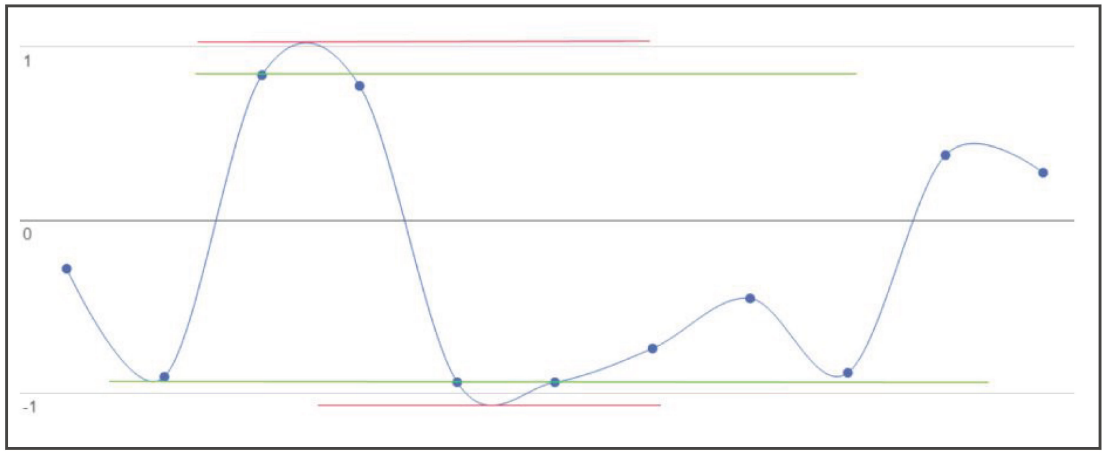


Figure 1: An analog waveform (blue line) with dots indicating the digital sampling points. The digital peak values are shown by the green lines with red lines for the true peaks. (Image courtesy of NUGEN Audio)

Supplementary Material as "The Truth about True Peak Limiting for Audio.pdf." The Supplementary Material can be found on the audioXpress website or by following the link listed under Project Files at the end of the article.

True peak is measured in loudness analyzers by over sampling the original analog signal. That is, a higher sampling rate (often four times the original rate) is used to provide a good probability of finding the true peak. Some of the analyzer programs display the original digital peak and the true peak.

Absolute vs. Relative Loudness

The three measurements in absolute loudness are the integrated loudness over the entire duration of the selected audio, which may be a track, song, symphony, and so forth. This is the "I" value in LUFS, loudness units relative to full scale. LUFS is sometimes written as LKFS, which means K-weighting has been applied. More information about this can be found in the first article in this series, "Loudness Revisited: Part 1" (*audioXpress*, June 2017). Also, a loudness unit is usually taken as 1 dB, so an "I" reading of -10.0 LUFS is 10 dB below 0 dB full scale.

The second measurement is the "S" value, which is made with a sliding 3-second integration over the selected audio. It, too, is in LUFS or LKFS. The third measurement, "M," is the momentary loudness. Its primary use is calculating the loudness range, usually abbreviated LRA. Loudness range is not the same as dynamic range, although they are related.

Relative loudness has the same three measurements, but they are relative to a "target" loudness. There are several targets in common use (see **Table 1**). The target values are used to normalize the audio's loudness so having a variety of targets can be confusing. For example, targets of -23 and -16 LUFS are suggested for home server music. To see what this means, let's say our "I" value is -20 LUFS. Normalizing value (dB) = Target – "I" (or maximum "S," depending on your choice).

Normalizing to -23 LUFS requires 3 dB attenuation applied to selected audio. The -16 LUFS needs an amplification of 4 dB. But, we also have to keep our true peak in the -2 to -1 dB range. We will have no problem with the -23 LUFS target because of its attenuation, but we may have a problem meeting the -16 LUFS target. The 4 dB amplification may push the true peak over the limit so the only way to meet both the target and the true peak is to allow some compression. This is all about realistic sound reproduction—we probably don't want to add compression.

Fortunately, the loudness analyzer will show the true peak of the original selection. If it's -5 dB or lower, the 4 dB gain will be acceptable.

Service	Standard	Normalization	True Peak
Suggested levels for home servers	ITU BS.1770/3/4	-16.0 or -23.0 LUFS	-2 to -1 dB
Deezer	No information	No information	No information
Apple Music	No information	-16.0 LUFS*	No information
Spotify	No information	-12.0 LUFS*	No information
Tidal	ITU BS.1770/4	-16.0 LUFS	-2 to -1 dB
YouTube	No information	-13.0 LUFS*	No information
* These do not use ITU BS.1770/3/4 measurement so numbers are approximate.			

Table 1: This table shows the available loudness normalization data as of November 2016. In the US, the CALM Act has set a -23.0 LUFS target for the broadcast industry, but the streaming services have been slow to catch up.

The Real World

I have several friends and acquaintances who have digitized all their collected music and put it on a home server. The sources include 78 RPM records, reel-to-real magnetic tape, audio and video cassettes, 33-1/3 and 45 RPM records, purchased CDs, purchased DVD movie sound tracks, and streaming services downloads. It would be helpful to have all the play lists play at a consistent loudness.

Thus, we enter the world of loudness measurement and normalization.

BS1170GAIN and MP3TAG

I will mention these two Windows programs first because they are free. Their use is limited but

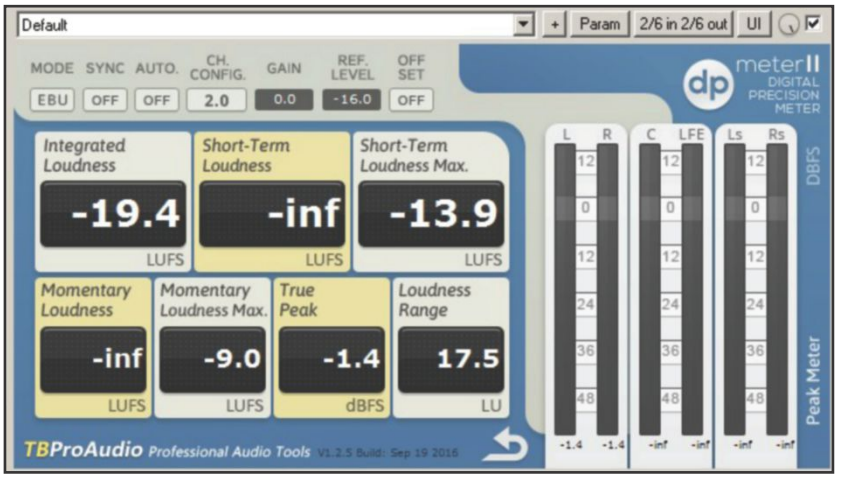


Photo 1: This screenshot of dpMeter II has a reference level of -16.0 LUFS. The loudness values and true peak are not adjusted to the reference. The 3.4 dB gain needed to match the Integrated Loudness to the reference would increase the true peak into the clipping range unless compression is allowed.

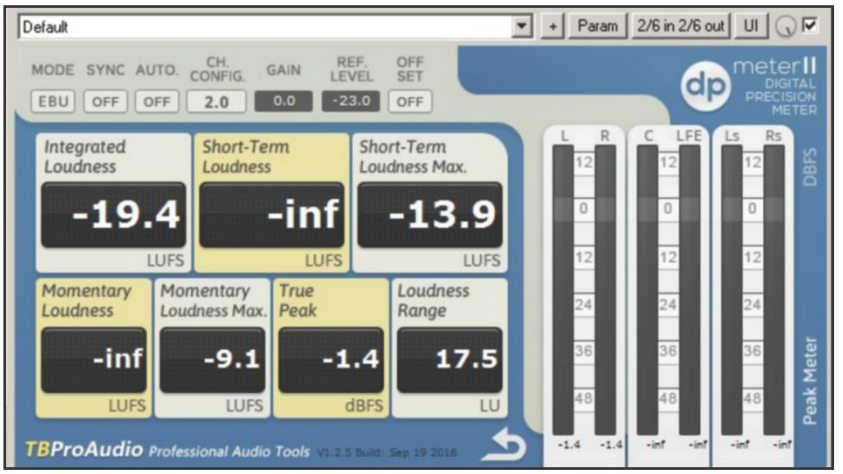


Photo 2: Another dpMeter II screenshot with the reference set to -23.0 LUFS. Now, 3.6 dB of attenuation will match the Integrated Loudness to the target so this is a good solution. Of course, the normalization would change if the Short Term Loudness Maximum were used rather then the Integrated Loudness.

they will work fine for loudness measurement and normalization if you are using a Sonos-Connect unit between your music server and stereo system. This would also work for any hardware unit that recognizes metadata tagging.

The BS1770GAIN program (which has recently been replaced by an updated version, R128GAIN) measures "I," "S," and True Peak. MP3TAG lets you define the REPLAYGAIN tag in the audio file's metadata. The "MP3" part is just a name, the tagging works fine for FLAC files. You will have to pick a consistent target for your music files. Then, using your choice of "I" or "S," calculate the gain or attenuation needed for the tag. Also, keep in mind the effect gain may have on the true peak. Documentation for both programs is included in the Supplementary Material.

dpMeter II

This free loudness measurement DAW plug-in is for Windows XP SP3 and newer, and Mac OSX 10.5 and newer. I used the Reaper DAW for all my measurement, but most all DAWs should work

In **Photo 1**, the reference level is set to -16.0 LUFS. With an "I" of -19.4, the added gain needed to match the reference is 3.4 dB. Adding this to the true peak of -1.4 dB bumps it well into clipping—not a good solution. If instead, you choose to match to the maximum "S," the attenuation of 2.1 dB would also lower the true peak. But you would have to be consistent and use maximum "S" for all the normalization.

I used a reference of -23.0 LUFS for the measurement shown in **Photo 2**. Matching "I" to the reference shows a needed attenuation of 3.6 dB and matching to maximum "S" requires 9.1 dB attenuation. Both would work.

Now that we have the normalization numbers, how can we use them? It's easily done with any audio editor. Open the music file and amplify its level

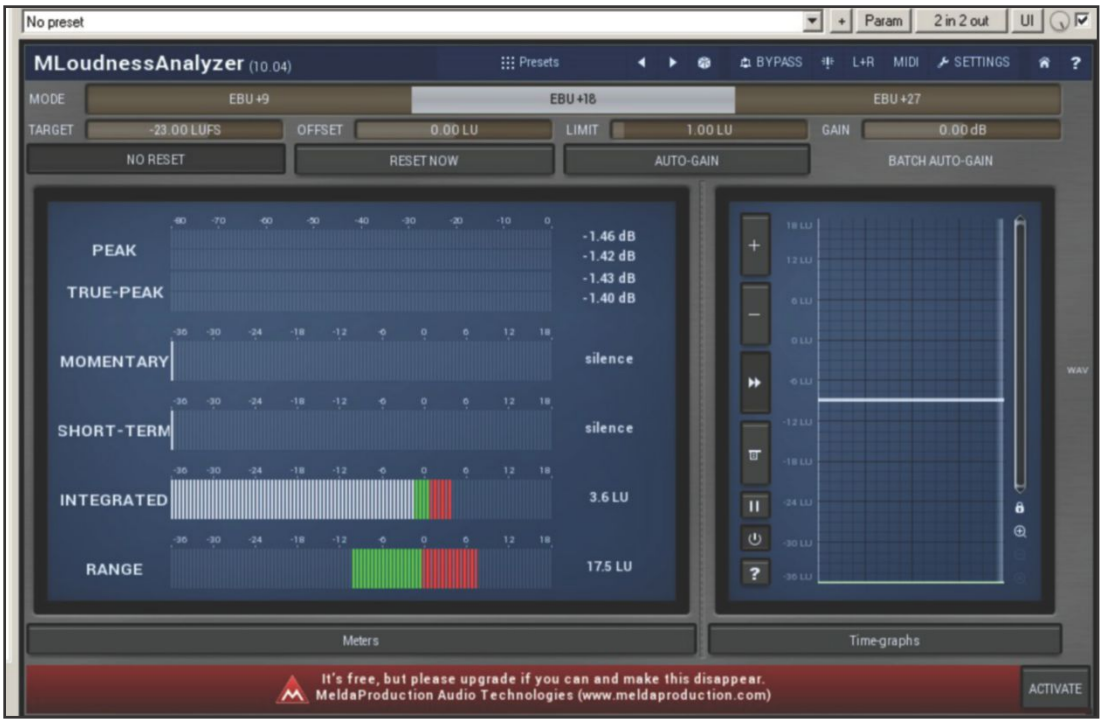


Photo 3: A MLoudness Analyzer screenshot with a -23.0 LUFS target. This analyzer displays the 3.6 dB attenuation to match the Integrated to the target. The Maximum Short Term is not held so it must be read from the screen just before playback ends. Both free analyzers display the same measured true peak.

Sound thinking in coaxial design



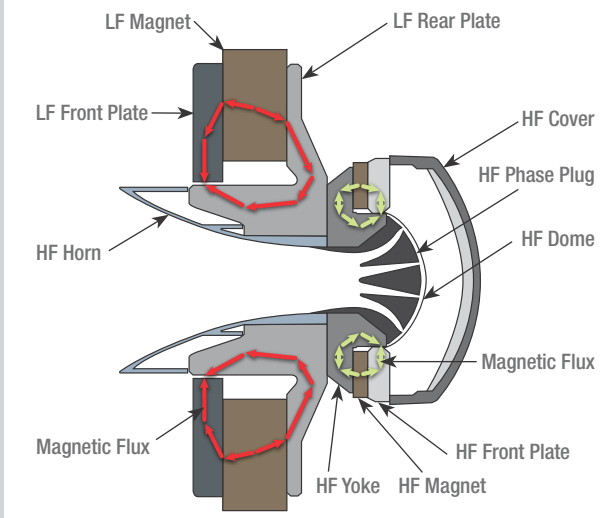
How a common magnet motor delivers a better sounding, lightweight, full-range speaker

Coaxial drivers have long been chosen as an alternative to two-way speaker systems, delivering benefits in size, weight and off-axis response. By concentrically aligning the low and high frequency components, coaxial loudspeakers act as a single source.

Moving the concept forward, Celestion's FTX coaxial range features fully combined LF and HF components that are powered by a Common Magnet Motor Assembly - i.e. using the same magnet for both elements. This enables the voice coils and hence the acoustic centres of the two drivers to be brought much closer together. The end result is a significant improvement in signal coherence and time alignment, and the most natural sounding audio reproduction.

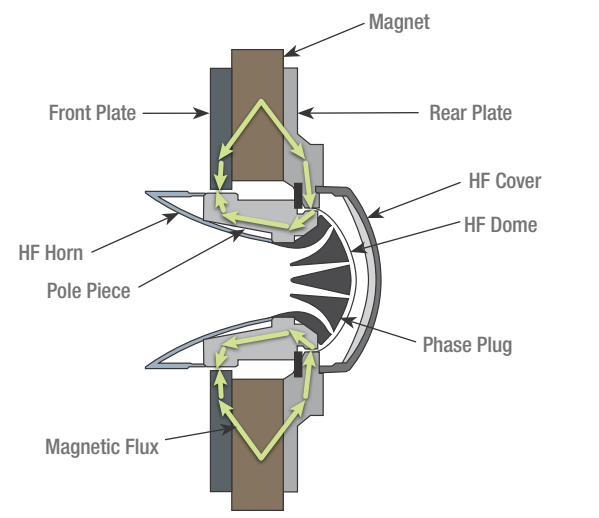
The magnetic flux is optimally distributed between the two elements (calculated using FEA), to achieve the best balance between HF and LF response (see diagrams below). Additionally, the use of a single magnet assembly also means lighter weight and a more compact profile, compared to more conventional dual motor designs.

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Traditional Coaxial Driver

With a magnet for each driver, the traditional coaxial design is heavier and the voice coils are further apart.



Celestion Common Magnet Motor Design

Celestion's single magnet design is a more compact, lighter-weight alternative with significant improvements in signal coherence and time alignment.

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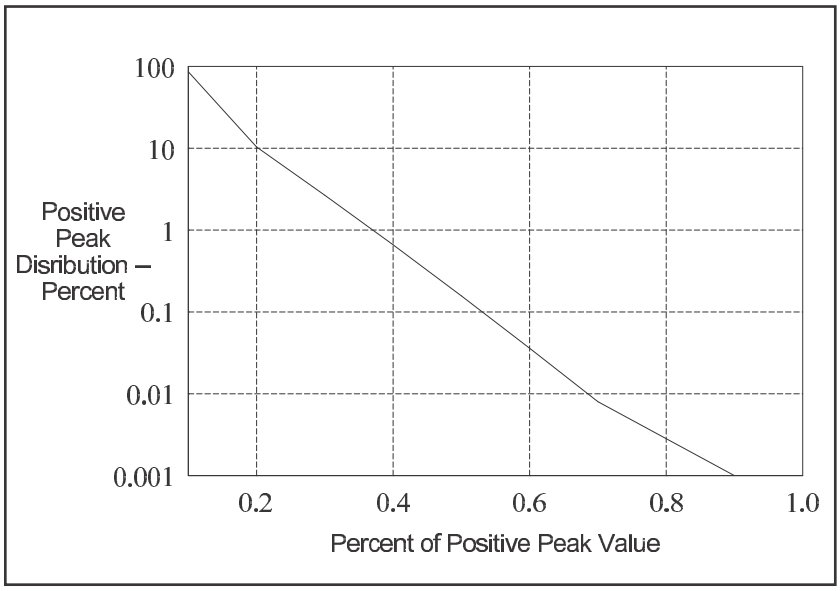


Figure 2: I tested the dpMeter and the MLoudness Analyzer with the Chopin Nocturne mentioned in the text. This rendition by Arthur Rubinstein has an excellent dynamic range, which should remain uncompressed during normalization.

by the needed positive or negative decibel value. It seems a target of -23.0 LUFS would always be safe, but perhaps some experimentation is needed to determine what is best for your music collection.

MLoudness Analyzer

Another free DAW plug-in from Melda Production is for Windows and Mac and it's available in 32- and 64-bit versions.

In the screenshot shown in **Photo 3**, I set the reference to -23.0 LUFS. Unlike the dpMeter, this analyzer shows that 3.6 dB attenuation is needed to match the reference without having to do the math. It's not too difficult to remember the true peak will also decrease by the added attenuation. Unfortunately, the short-term peak is not saved when the track ends. But you can write it down just before playback stops.

For both these measurements, I used a Deezer streaming capture of Frédéric Chopin's "Nocturne No. 1" in B-flat minor from the Arthur Rubinstein album: *Chopin Nocturnes and Mazurkas*. This track has excellent dynamics (see **Figure 2**). I think it would be un-musical to enable normalization compression to mar this beautiful rendition.

There have been some blog comments on the web about -23 LUFS being too quiet for mobile devices because of their often noisier environments. This may be true. I have normalized the Chopin track to -23 LUFS and it's fine in my listening room. A 90-second clip of this stereo wav file, 44.1 kbps and 16-bit depth, is included in the Supplementary Material. Listen and decide for yourself. Adobe pdf User Manuals for these two loudness analyzers are also included in the Supplementary Material.

To be continued...

Project Files

To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html.

Resource

"Loud Commercials," CALM Act, Federal Communications Commission (FCC), www.fcc.gov/media/policy/loud-commercials.

Sources

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Deezer | www.deezer.com

MLoudness Analyzer
MeldaProduction | www.meldaproduction.com/MLoudnessAnalyzer

MP3TAG
MP3TAG | www.mp3tag.de/en

BS1770GAIN (now replaced by an updated version, R128GAIN)
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dpMeter II
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Hypex Electronics

Class-D High-Fidelity

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. Dutch company Hypex Electronics is one of the fastest rising stars on the global amplifier technology scene. Known mainly for its Ncore power amplifiers modules, Hypex is taking Class-D technology to an entirely new level.



Photo 1: The Hypex Nc400 module has been on the market for some time now and its popularity in the DIY market is an indication of the nCore technology's success.

By
Ward Maas
(Netherlands)

Hypex Electronics, B.V., founded in 1996 by Jan-Peter van Amerongen, started as a supplier of plate amplifiers for the pro market. The first products were mainly used for active subwoofers for live sound. Shortly after that, multi-channel products with active crossovers followed. Those plate amplifiers also solved a few production problems both for manufacturers of studio monitors and active hi-fi loudspeakers.

The demand for higher power led van Amerongen on a search for a high-efficiency amplifier that combined both power and quality. At that time, the available Class-D solutions did not appeal to him as those products were not that good and not simple at all.

Philips in the Netherlands, however, had a solution that might work. During a meeting with designer Bruno Putzeys, van Amerongen became convinced that this was the solution for which he was waiting. This meeting turned out the pivotal event in Hypex's history. It was the start of a long-term cooperation between van Amerongen and Putzeys, which continues today.

In 2003, Hypex completed a migration to "Universal class-D" (UcD) and, in 2004, the company moved to its current headquarters in Kattegat 8, Groningen (The Netherlands). In 2005, the first Class-D modules were available for the OEM market and Hypex began delivering amplifier platforms to many big name audio brands.

Meanwhile, Putzeys left Philips and started to work further on the development of Class-D amplifiers. Hypex made the strategic decision to move from being a technology user to being a technology source and hired Putzeys (UcD's inventor) to be its chief of R&D. As Putzeys explained in an Audio Engineering Society (AES) meeting in the Netherlands, it kept bothering him that his calculations for UcD were "always almost correct." That "almost" was a driving force that one day led to a "eureka moment." His explanation to the Hypex R&D department on a whiteboard, on a Friday afternoon, was enough to convince the team to open a bottle of champagne. Their confidence was such that building a working prototype was not seen as a problem. Ncore was born.

In 2008, Hypex started incorporating its new world-class NCore technology in its end-products. The Nc 400 amplifier module, the first of the Ncore products, has a remarkable appearance. It has a round shape. Rumor has it that this is a practical joke translation of the back-of-a-beer mat calculation that was at the base of Ncore (see **Photo 1**).

From its origins, Hypex also became popular with DIYers in the need of an amplifier or a complete plate solution. Its popularity remains unchanged. Recognizing that it has strong following, in 2016, Hypex remodeled its website and gave its DIY section its own domain at www.diyclassd.com.

The story of Hypex typically illustrates the recent events in the world of audio power amplification design. The typical linear audio amplifier designer is, in most cases, not the same as a Class-D amplifier designer. Input and output of their designs are the same. In between, there is a world of difference. Where an analog amplifier is a work of careful balancing of analog circuits, a Class-D amplifier is a mathematical challenge.

Since the launch of its nCore amplifiers, Hypex has been continuously putting out new designs and solutions. Today, Hypex Electronics designs high-performance Class-D power amplifiers, power supplies, and DSP solutions—based on its NCore and UcD technologies—for the professional, high-end, and DIY audio industry. The company's catalog also includes a range of switch mode power supplies (SMPS), active speaker plate amplifiers and an extensive range of high-end audio solutions, including analog circuit and DSP designs.

All Hypex production is mainly done in Malaysia—a production location chosen by most



Photo 2a: Well-established in the OEM market, Hypex is reputed for many projects developed in close cooperation with known brands, often providing complete solutions that include loudspeaker and system control. Hypex' recent focus seems to be in general-purpose power amplifier/supply module development, expanding its portfolio extensively. The company is updating its complete range of OEM DSP modules, planned to be released in the second quarter of 2017. 2b: Hypex has been expanding its nCore based NC*xxx*MP amplifier module range, as we can see in the picture. On top we can see the new Fusion Amp plate amplifiers, displayed at Prolight+Sound 2017. 2c: The Hypex nCore range of Class-D amplifiers combines the stability of UcD with improved load-independence, lower distortion and lower output impedance. In its Ncore NCxxxMP amplifier module series, Hypex incorporates a low power standby power supply that meets 2013 ERP Lot 6 0.5 W requirements, a highly efficient switch mode power supply, and the Class-D amplifier circuit in one compact and easily applicable solution.

Class-D manufacturers. At its Groningen facilities, Hypex keeps its R&D, planning, sales, support, and a small warehouse. The company's largest warehouse is located in Hong Kong.

Ncore Optimization

Ncore technology combines the stability of UcD with improved load-independence, lower distortion and lower output impedance. There is a technology white paper regarding Ncore available on the Hypex website, which describes the progress made by Ncore. (The white paper is also available in the Supplementary Material located on the audioXpress website, see Project Files for the link.) In that paper, Putzeys explains how Ncore builds on the stong heritage of UcD, "Ncore takes the things UcD does well and does them 10 times better. Literally."

Basically, Putzeys characterizes the technology in four parts:

- A mathematically exact understanding of self-oscillation. This allows optimization of large signal performance.
- Improved 10 transistor comparator circuitry, insuring that actual behavior matches the theoretical model as closely as possible.
- New gate drive circuitry improving open-loop distortion at moderate signal levels while significantly reducing idle losses.
- A new control loop enlarging loop gain by 20 dB across the full audio range without sacrificing stability.

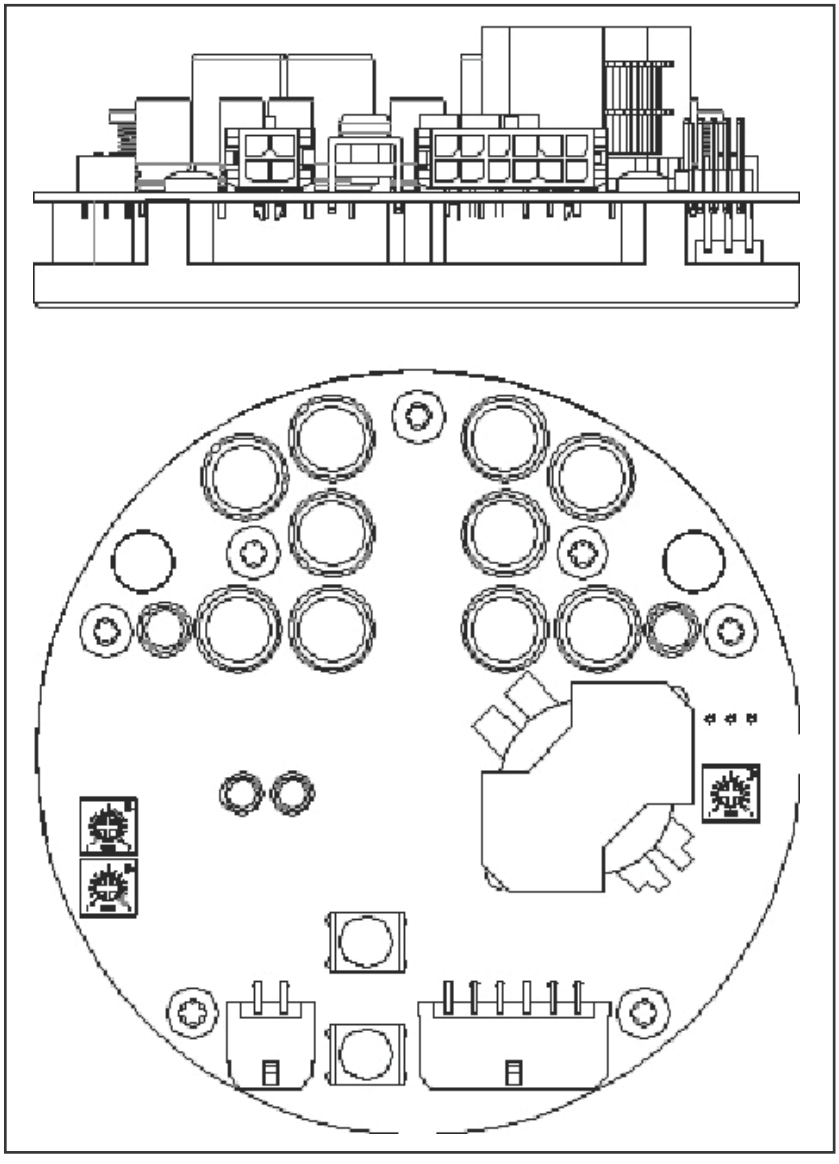


Figure 1: This is a sketch of the Hypex Nc 400's outline.

"Amplifiers using all four parts will be marketed under the name Ncore. Amplifiers using only the first three will still be sold under the UcD brand, even though their internals no longer resemble that of the well-known 2001 circuit and their performance is already a clear step up," Putzeys adds.

Ncore Specs and Graphs

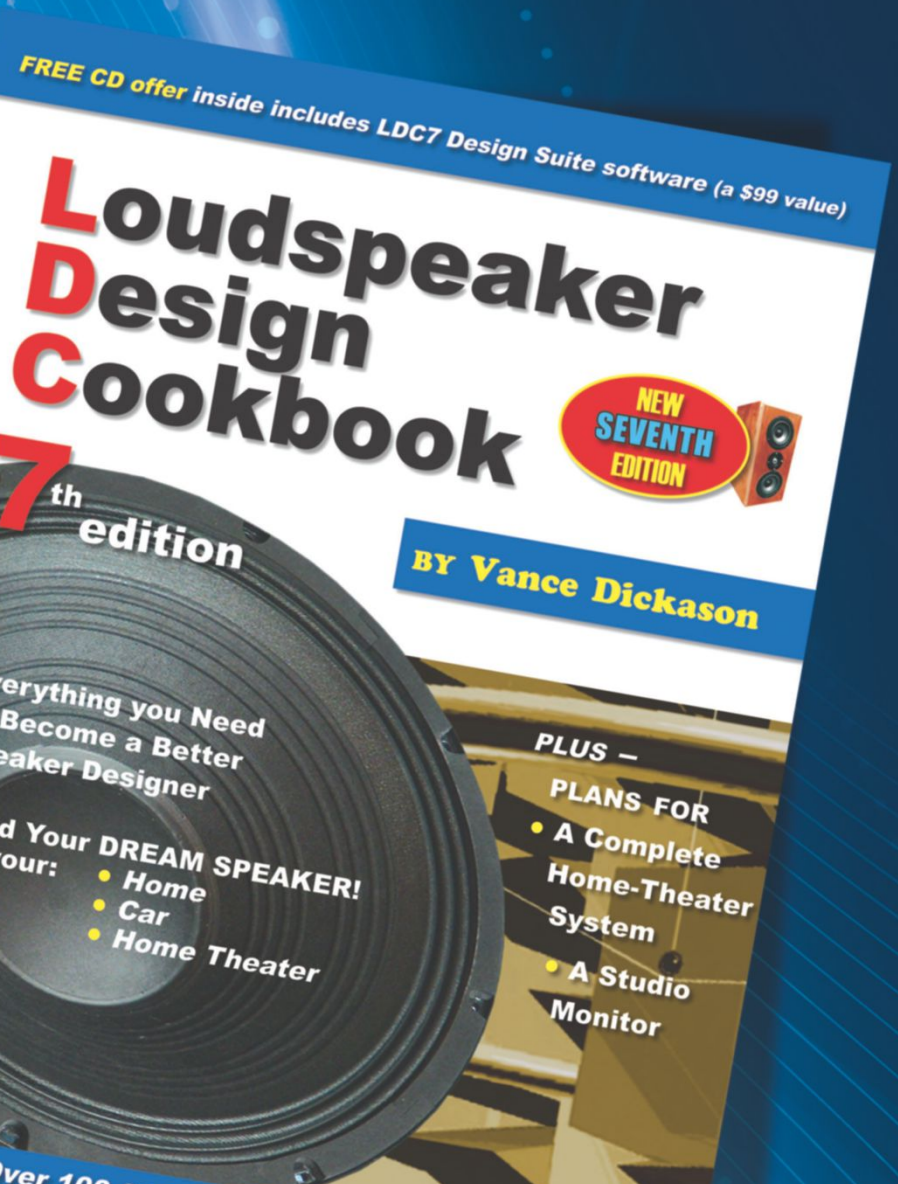
On the Hypex OEM website, we can find approximately 16 different amplifier modules ranging from a stereo 30 W to a 2500 W UcD module, including seven different products with Ncore technology. The company has several more in the works, which have already been displayed at recent trade shows (see **Photo 2**). Most of them come with a matching power supply. They have remarkable specs in common. The original Nc 400, as described later in the article, is one example. The specs do slightly differ depending on the product. That is due to the PCB layout, the intended use of the product, and the target customer.

Knowing this is an established reference on the DIY market, we looked closer at the Nc 400, as a first example of a Hypex Ncore amplifier. The Nc 400 has been on the market for some time now, so why not focusing on a newer product? Obviously, the availability of a complete unit for testing comes in handy, but mainly because the same nCore calculation is the foundation for all the first generation nCore amp modules. The slight deviations in specs are product choices. Please do not think analog amps here.

The first noticeable thing is the size. The device is only 88 mm in diameter, with a height of 37 mm, and yet it is capable of delivering 400 W in 4 Ω, and reaches an impressive peak power of 580 W in 2 Ω. The efficiency is 93% (see **Figure 1**).

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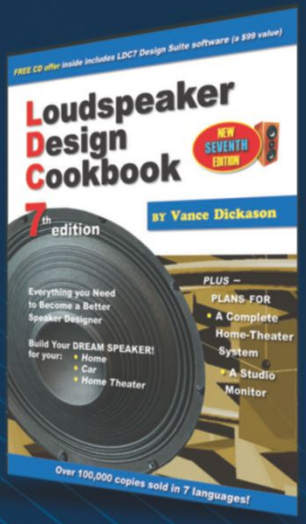
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The high-efficiency of Class-D amplifiers often leads to an underestimation of the cooling required. The amount of cooling needed varies with usage. Music has a large peak-to-average ratio, so most of the available amplifier power is used as headroom. Audio enthusiasts typically listen to music at a few watts at most with only the occasional full-scale peak. On the other hand, during a party where hyper-compressed modern pop music might be played, the average level could reach one-third of its rated output. The Nc 400, without additional cooling, has a thermal resistance of around 6 kΩ/W. When targeting intensive use, design for 133 W, translating to Ploss = 4.5 + 0.06 × 133 = 12.5 W. Required heatsinking is 45 kΩ/12.5 W = 3.6 kΩ/W. This indicates that any normal metal enclosure will provide enough additional cooling even for heavy use.

The main Nc 400 specifications show, apart from the impressive power and efficiency, an extreme low distortion of 0.0007% and a signal-to-noise ratio (unweighted) of 102 dB. In case of a Class-D amplifier it simply means that you know what you are doing. It means a very fine control of switching on and off of those power field-effect transistors (FETs). Anything not under control simply means a fast deterioration of those figures.

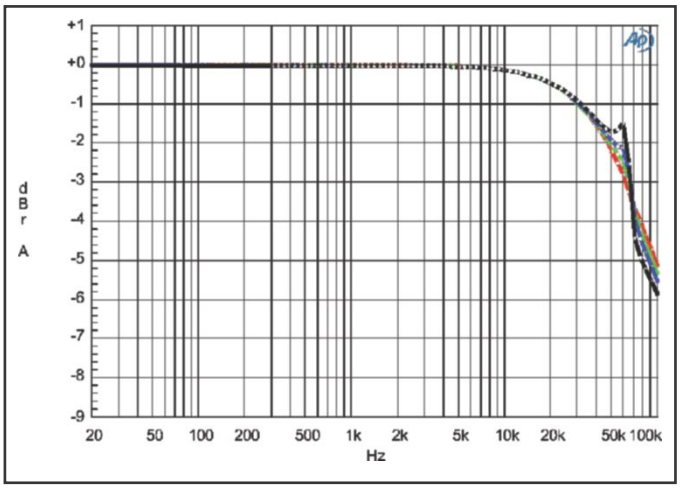


Figure 2: Hypex Nc400's frequency response

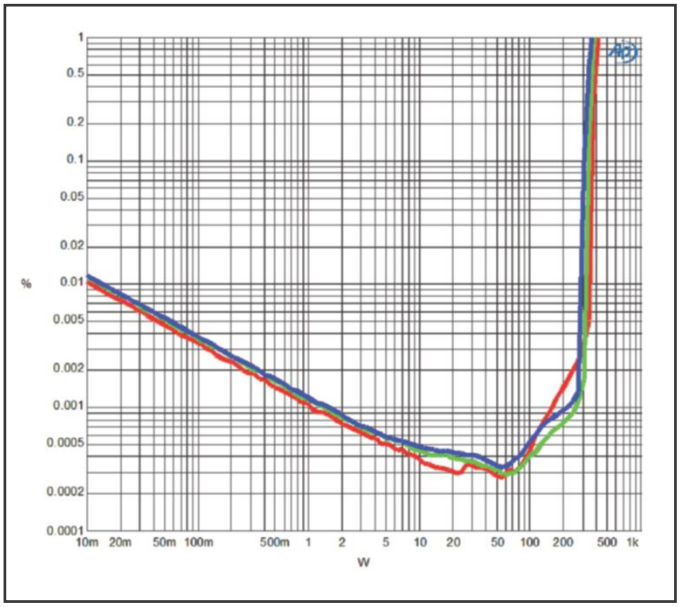


Figure 3: Hypex Nc 400's THD + N vs. Power

The graphs shown in **Figures 2–4** speak for themselves. Or so you would think.... The fact is that you will find these kind of graphs on very good analog amplifiers. Until a few years ago, they were not to be found for Class-D amplifiers. True, the frequency graph shows a -1 dB point at 30 kHz, which means at 20 kHz there is already a 0.5 dB drop in amplitude. However, I believe that to be a wise design decision and not a design flaw.

The Power Supply

Until recently, a powerful amplifier was recognized by its massive heatsinks and massive power supply. As the efficiency of Class-D is about 90%, the massive heatsinks are gone. Wouldn't it also be tempting to get rid of that massive power supply as well? A switch mode power supply would be a desirable choice, of course, but will it be comparable with a good old large toroidal and a mighty capacitor bank?

During an AES lecture in the Netherlands, held at the Hypex

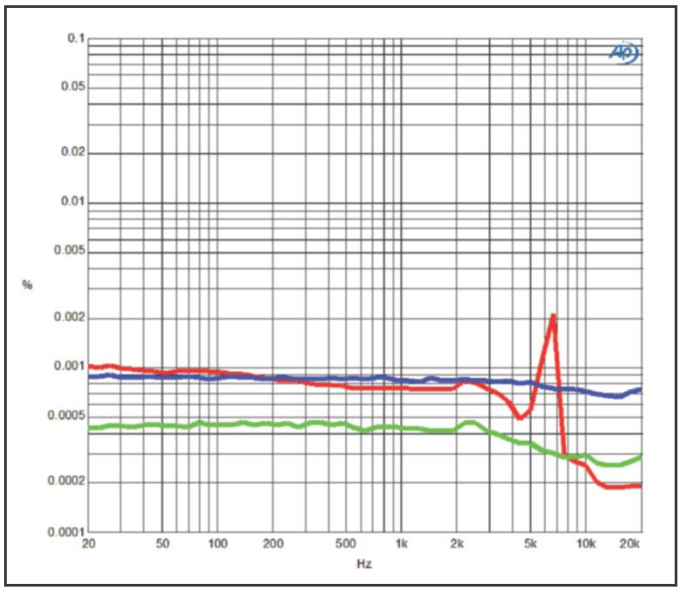


Figure 4: Hypex Nc 400's THD vs. Frequency

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.

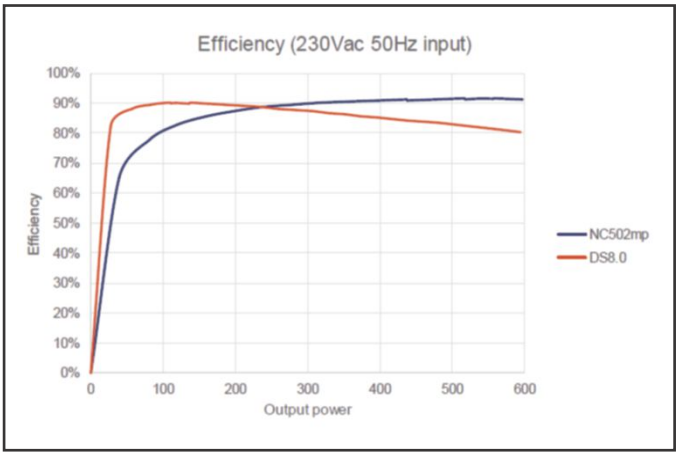


Figure 5: Power supply efficiency

premises in Groningen at the end of last year, the power supply development group of Hypex showed some measurement results that were very revealing. Indeed, not every switch mode power supply is a good candidate to replace the linear power supply as used in the majority of amplifiers. Bus-pumping is a risk and there is that intuitive feeling that charging a small capacitor very fast is not the same as charging a large capacitor at a slower rate. The presentation clearly showed that at higher power levels the so called LLC switch mode power supply was much more efficient than the conventional power supply.

In an overload situation where the switch mode power supply was in over current protection, a recovery to the normal situation was reached much faster than with the traditional power supply. **Figures 5–7** show a comparison between an Hypex DS 8.0 "traditional" power supply and a Hypex NC502mp switched-mode power supply (SMPS).

Not only did Hypex grant that wish for a small, efficient power supply, it proved that a well-adapted switch mode power supply is better suited for an audio amplifier than a traditional one. With the introduction of SMPSs capable of feeding audio

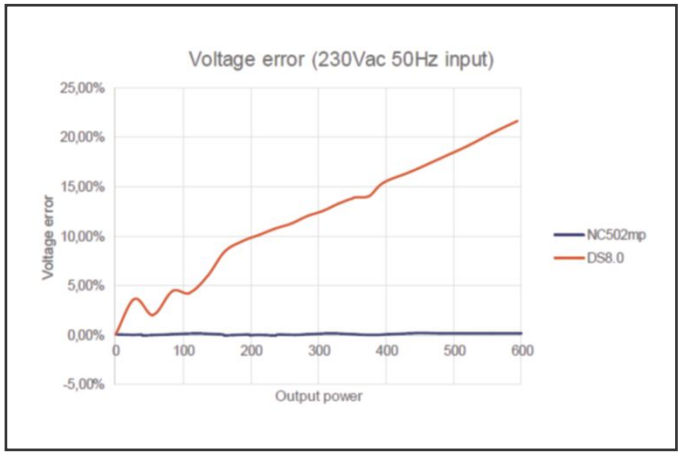


Figure 6: Voltage error

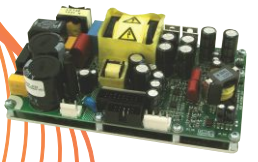
hypex NCxxxMP series

overview

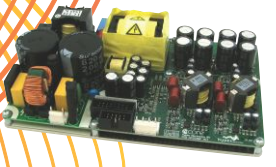
The NCxxxMP amplifier module incorporates a low power standby power supply (meets 2013 ERP Lot 6 0.5W requirements), a highly efficient switch mode power supply and a high- performance Class D amplifier in one compact and easily applicable power brick.



NC122MP



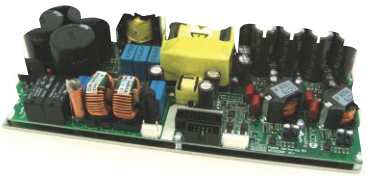
NC250MP



NC252MP



NC500MP



NC502MP

Highlights

- High efficiency
- Universal mains operation
- Flat, fully load-independent frequency response
- Low output impedance
- Very low, frequency independent THD
- Very low noise

Features

- One or two channel amplifier
- 5W standby SMPS
- Advanced over current protection
- External controlled operation
- Auto-switching (115/230V)
- Low weight
- Compact

Applications

- Monitor loudspeakers for recording and mastering studios
- Audiophile power amplifiers for professional and consumer use.
- Public address systems
- Active loudspeakers

Add-on Module NC100HF
The NC100HF is a dedicated tweeter-amplifier which fits the NCxxxMP series.



ncore audio power conversion

hypex electronics
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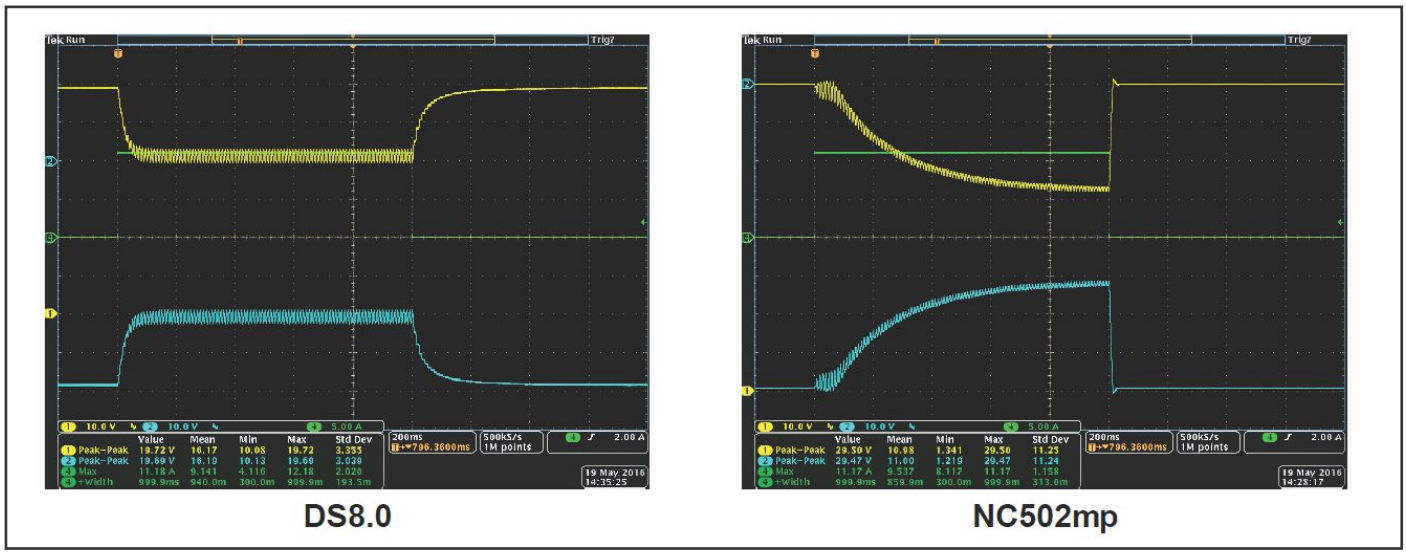


Figure 7: The pulse load at 11 A per 1 second

amplifiers, the development of high-power, high-efficiency amplifiers has entered a new stage.

The EMI Compromise

One thing not mentioned here is one of the negative aspects of switch mode products, whether they are power supplies or Class-D amplifiers. That is electromagnetic compatibility (EMC)/ electromagnetic interference (EMI).

High efficiency does come with its own design challenges. A Class-D audio amplifier's switching topology produces EMI. EMI creates radiated and conducted noise that causes interference. High dV/dt and dI/dt are generated, where dV/dt is the intantaneous rate of voltage change (volts per second) and I is the instantaneous current through the capacitor. This means that any part of the circuit large enough to act as an antenna will do so. So PCB tracks and cabling will radiate unless a lot of effort has been done to prevent high-frequency signals from reaching them. In other words, printed circuit layout is critical. Unfortunately, there are even more ways to turn a perfect Class-D amplifier design into a transmitter.

On the Hypex website (www.hypex.nl), you will find an application note regarding EMI. It explains in a funny and scary way the do's and don'ts of reducing EMI. So yes, taking care of the EMI problem takes knowledge and experience.

Until recently, Hypex offered amplifiers and separated power supplies to the market. The new products offer integrated solutions, which provide a complete platform that is ideal for audio companies not having in-house development capabilities in the field of electronics. Due to the different type of electronics development needed for this kind of

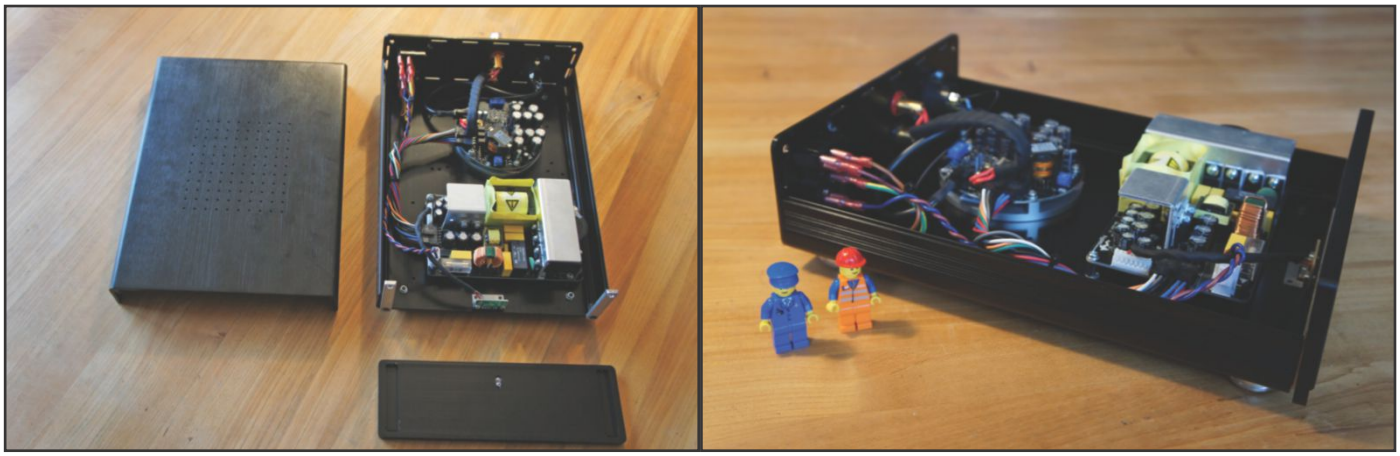


Photo 3: This is the complete DIY Nc 400 mono block provided by Hypex for our tests. In the same "universal" chassis, you can adapt almost all existing Hypex modules.

amplification, it is also a solution for companies not willing to start a development in that direction and enabling them to focus on their specialties.

Testing One, Two, Three...

As the flagship Ncore products are, according to Hypex, the first products that surpass traditional amplifiers in every aspect relevant to sound quality, Hypex believes that we are likely to see a landslide effect in every aspect of the amplifier market. It is already visible (and audible) in the newest generation products.

I received two Hypex Nc 400 mono blocks for testing (see **Photo 3**). There are plenty of Nc 400 reviews focusing on hi-fi and DIY applications. So, I decided to test them under completely different conditions.

For the first test, I used it as the central amplifier on a cross country running event. I used a row of several small Electro-Voice (EV) loudspeakers in parallel, forming a load of about 2 Ω per channel, to broadcast background music and announcements. The complete installation was fed by a small portable mains generator. The experience was pleasant. No problems at all and the amplifier chassis did not even become warm to the touch.

For the second test, I used the Nc 400 as the driving force of a large studio system. We played the system loud with all kinds of music. The result was mostly the same as with the previous experience, except that the Nc 400 got slightly warmer this time. To be honest, these are the outcomes I expected for these tests.

The third test, however, was the one about which I was most curious. Hypex claims the Nc 400 is a high-end amplifier and as such comparable or better than products available in the high-end scene. I managed to get a selection of expensive amplifiers and speakers together on a Saturday afternoon for a shootout. That turned out to be an afternoon to remember. Let me give you the results first. The Nc 400 was fully on par with all the transistor amplifiers available. In the bass region, it proved to be capable of providing the most controlled and tight bass, better than the rest. Only in mid/high region did a very modified tube amplifier sound, let's say... "friendlier."

After the dust settled, in the discussion that followed, the "friendlier sound" was directly attributed to the well-known tube amplifier behavior with its pleasant distortion "color." The most remarkable fact was that nobody disputed the presence of the Nc 400 in this company of high-end amplifiers costing thousands of dollars more. Neutral and transparent reproduction, "neither dirt



Photo 4: The well-organized warehouse in the company's headquarters in Groningen, The Netherlands, is mainly used to keep DIY and sample products available for delivery.

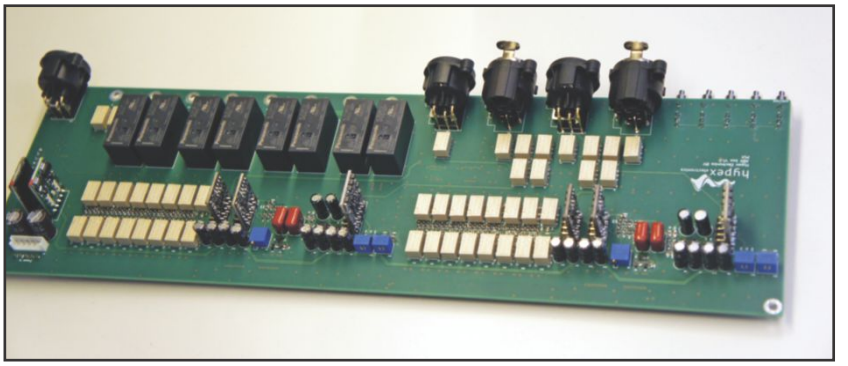


Photo 5: This is a house-made switcher for amplifier listening tests.

LLC Switch Mode Power Supply

The LLC switch mode power supply is based on a resonant inverter (i.e., a system that converts a DC voltage into a sinusoidal voltage or more generally, into a low harmonic content AC voltage) and provides AC power to a load. To do so, a switch network typically produces a square-wave voltage that is applied to a resonant tank tuned to the fundamental component of the square wave. The resonant tank consist of an LLC network. Hence the name.

Ncore

According to Bruno Putzeys, Ncore is mainly about the exact oscillation criterium and accompanying large-signal behavior. Different from UcD is the exact analysis method. It seems that the correct mathematics for Class-D did not get the attention it deserved. Predictability of a Class-D amplifier has been greatly improved. With Ncore, it is possible to increase the loop gain without getting instability problems. With the higher possible feedback, the distortion dramatically decreases. The Ncore control circuit also ensures the return to a stable situation after clipping.

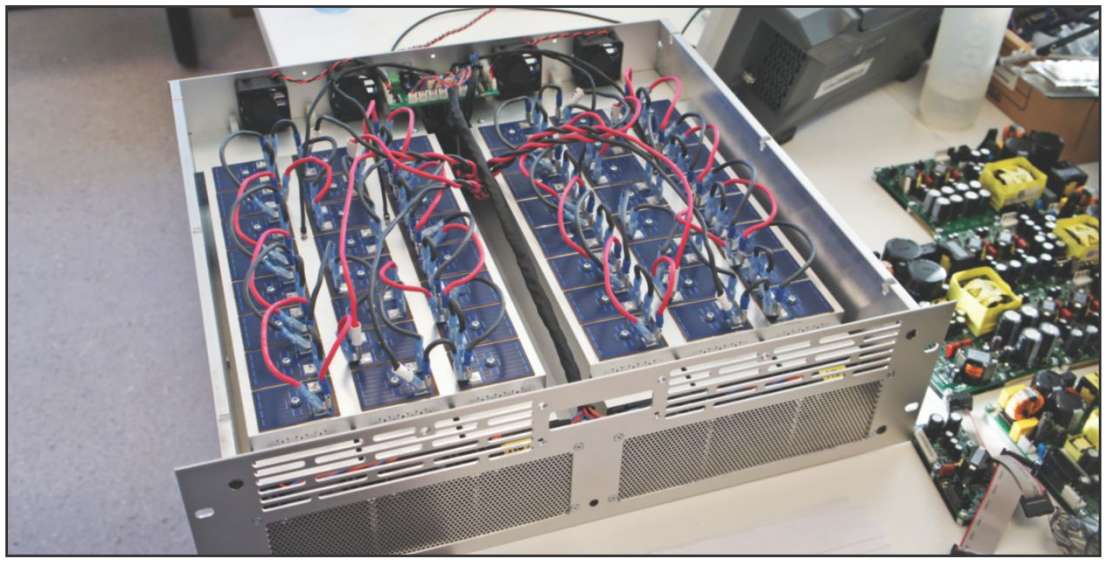


Photo 6: A house-made load on the bench is able to handle even the most powerful products.

nor fairy dust" as Hypex promised, and that is what the test audience got.

After mentioning the fact that this Nc 400 had a price tag of 650 Euros (approximately $727), it was decided then it would be better not to mention the brand names of the other products tested in the session.

As Putzeys already mentioned in his 2011 nCore paper (see Resources), "I understand that one of the joys of audio writing is generating baroque prose to describe the sonic vagaries (or pleasantries) of products designed expressly so that reviewers can have something interesting to write about them, but as an engineer I can't help preferring doing a good job."

"If you want to forget completely that you're listening to an amplifier, how it's made and how it works and instead you just want to get sucked into the music, this amp is for you," he adds.

Kattegat 8

Visiting Hypex was a nice experience. The reception was friendly and also very open. A guided tour through the premises always shows something about a company's culture. I have been in many production plants and research laboratories, however, I have not often seen such a degree of purpose.

The Kattegat is in an uninteresting industrial area of Groningen, in the northern part of the Netherlands. The mentality in that part of the Netherlands is often described as "no frills, no thrills." Hypex knows it is changing the world, and its building, warehouse, lab, and even its logistics software radiates purpose.

About 15 people work for Hypex in Groningen. Most of those employees have been there since Hypex began and have grown with the company and are very motivated. The first floor of the building is mainly for development. Different from many other R&D departments, it is a bright and well-organized working environment. On the ground floor is the logistics department and the company's demo room. The management team also has offices on the first floor. In the back of the building is a small but well-organized warehouse where the DIY orders are shipped and where first-production samples are received (see **Photo 4**). There is also a small corner where units that have come back with a failure are immediately checked. The main production flow never gets to Groningen, since products go directly to OEM customers. Again, there is an almost serene atmosphere of quiet control.

Hypex also keeps a small group of three to four people working on the development of power supplies in Eindhoven, which is in the southern part of the Netherlands. The former presence of Philips in that region makes Eindhoven a strategic place for finding talent and for high-tech development. So yes, Hypex blends in nicely with the surrounding areas.

Mola Mola (www.mola-mola.nl), a finished product hi-fi brand, shares the same building as Hypex in Groningen. It shamelessly has the same company address and Putzeys and van Amerongen are listed as founders. Putzeys' hand is visible in the Mola Mola products: the hardware, the software, the user interface.... With no limitations besides his own vision, he has once again created something extraordinary. The Mola Mola products are modest in all aspects except in their performance.

In the lab we also found some Mola Mola products under test. The company follows the same approach that, "Once you've removed everything that isn't the music, what remains is the music."

As Hypex claims to provide high audio quality, naturally it cannot shy away from objective testing and we found plenty of evidence of that at its premises. In **Photo 5**, you can see the development results of a homemade switch unit, which enables them to switch sources, amplifiers, and speakers. No matter how good the measurements are, they only mean something if listening confirms that the design objectives are met. As always, if you are on the edge of what is technologically possible, you need measurement equipment that is not on the market. And at Hypex, equipment that cannot be bought is tailor-made for internal use (see **Photo 6**). That underlines the fact that the company knows what it's doing.

I expect more to come from Hypex in the near future. I was able to see a lot, but not everything... Hypex is obviously gearing up for even more innovations. We will return to it in a future article about further developments.

The Shape of Things to Come

Class-D amplifiers are not only here to stay, I expect they will soon represent a massive share of the amplifier market in a couple of years—in all application segments, and that includes high-end. They can provide high power, are inexpensive, efficient, and very good. From a commercial point of view, that is an unbeatable combination.

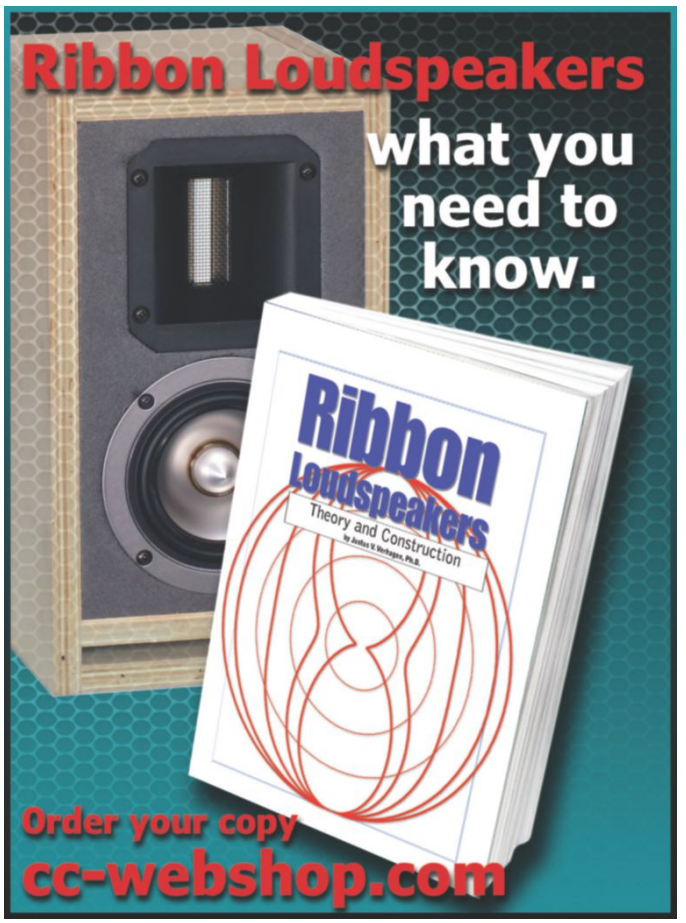
The only thing remaining for established amplifier brands is how fast they are willing to adapt to the changing market situation. You can try and cover up a Class-D amplifier inside some nicely designed aluminum casing. I already saw some "hidden" Class-D amplifiers from companies not ready to admit it to their customers. Others are choosing the "powered by" approach. Watch carefully for upcoming product introductions. It is the shape of things to come.

Project Files

To download additional material and several papers and application notes on this topic, visit
http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Resource

B. Putzeys, Ncore Technology White Paper, Hypex Electronics BV, February 2011, www.hypex.nl/img/upload/doc/an_wp/WP_Ncore_Technology.pdf.





Earthworks M50 Omnidirectional Measurement Microphone

Learn more about Earthworks and its recently released M series microphones, targeted at the middle of the market and ranging in price from $500 to $1,300 per microphone.



Photo 1: The M 50 is a member of Earthworks' M Series. The M Series has become the accepted standard for affordable, reliable reference and measurement microphones that are accurate in the time domain and frequency response.

By
Stuart Yaniger
(United States)

Speaker designers, acoustics researchers, and people with home labs are certainly living in a golden age of measurement. From the transducer end, there's good, capable microphones available from general purpose semi-quantitative units priced under $100 to calibrated specialty lab microphones in the multiple-thousands of dollars range, the choice of which depends on specific application requirements. Earthworks, a well-regarded US-based brand, manufactures and markets the M series microphones.

For those who are unfamiliar with the company, Earthworks was started by David Blackmer as a measurement microphone company back in the late 1990s before branching out to other markets and developing proprietary acoustics and electronics technology. Blackmer's goal was to produce the most accurate measurement microphones in the market, since he felt that most of the then-existing products did not give reliable results. The basic philosophy at Earthworks is optimization of the time-domain response. If the time domain is done well, frequency response will automatically be correct. The inverse is only true for minimum-phase systems, and Earthworks argues that non-minimum phase issues in competitive microphones (e.g., from reflections off grills and other internal structures) are not accounted for. According to Taoufiq Bellamine, Senior Design Engineer at Earthworks, "Everything starts with the understanding of the acoustic impulse response of the microphone, which no other microphone company to this day tests for. All major companies are still set in their ways to focus on the magnitude frequency response alone, which is only half the picture. You need to combine the magnitude and phase responses, to get an impulse response, which is the only accurate test for the true time response of the microphone."

M50 Omnidirectional Measurement Microphone
Earthworks Audio
37 Wilton Rd.
Milford, NH 03055
info@earthworksaudio.com
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The Product

The M50 under review here ($1,299 retail price) is the top entry of the M series, which mainly differ in high-frequency cutoff between models (see **Photo 1**). The capsule used in the M series is an electret with a plastic diaphragm. In general, plastic diaphragms can be lighter and less expensive than metal diaphragms, but have greater variation in properties with changes in temperature and humidity. As with all engineering, it's a tradeoff. The capsule diameter is 5.8 mm. This is also a tradeoff, this time between noise (smaller

capsules are noisier) and polar pattern (larger capsules are more directional). All M series microphones are omnidirectional (free-field) and use the same electret capsule (see **Photo 2**). The microphones operate off 48 V phantom power and require 10 mA of current.

Because the capsules are the same, the maximum sound pressure level (SPL) and sensitivity remains the same for the entire M series. The "50" in the M50 denotes response flat to 50 kHz; not surprisingly, the M23 is flat to 23 kHz, and the M30 is flat to 30 kHz. The high-frequency cutoff difference is determined by a tuning process.

On the electronics end, Earthworks uses a low noise filter to flatten out any resonances in the capsule and extend the response to high frequencies. They also do an acoustical tuning by adjusting tip geometry—introducing washers, capsule support inner stocks, and other mechanical tweaks. The mechanical design avoids having large cavities that might cause any reflection issues (non-minimum phase) or resonances inside the capsule housing. An M30 (with a 30 kHz cutoff), for example, can be tuned more quickly than an M50, which is reflected in the price differences. Further, to extend response to 50 kHz flat, they use a different ultrasonic stimulus and apparatus for the M50, based on a spark method.

Earthworks uses the sparker apparatus in order to provide a single, very fast, high-energy acoustic spark that simulates an ideal impulse. It can exercise the microphone's entire frequency range and show any issues with the mechanical construction or electronics. Taoufiq says, "I personally have a locker of all the high-end measurement microphones you can think of, and when we expose them to an acoustic spark, we can see some excess ringing and phase issues, not only because of their large diaphragm construction but also because of all the external grills used in them. These issues cannot be seen on a frequency chart alone."

Although some microphone companies use electrostatic actuators for calibration purposes, at least at higher frequencies, Earthworks does the calibration using acoustic signals, which it claims is an advantage of its approach. In a sense, a bug becomes a feature. Electrostatic actuators do not work properly on plastic diaphragms, but are usually used for metal diaphragms. In any case, the use of acoustic sources for calibration is documented in the IEC 61094-8 standard, and this standard is the basis of how I check microphone calibrations in my home lab. As later determined, the correspondence between Earthworks' calibration method and mine is quite good.

The microphone itself is finished in attractive brushed steel, and packaged in a padded case along with a mounting clip, foam wind screen, and a 0.25" calibration adapter (see **Photo 3**). An individual, serialized calibration sheet is included, and the calibration data may be electronically obtained at no charge when the microphone is registered online.



Photo 2: The M50's extended frequency response makes it suitable for loudspeaker measurement.

Measurements

Unless otherwise noted, all measurements were performed using an Audio Precision APx515 analyzer and the companion APx1701 transducer interface.



Photo 3: The microphone itself is finished in attractive brushed steel and packaged in a padded case along with a mounting clip, foam wind screen, and a 0.25" calibration adapter.

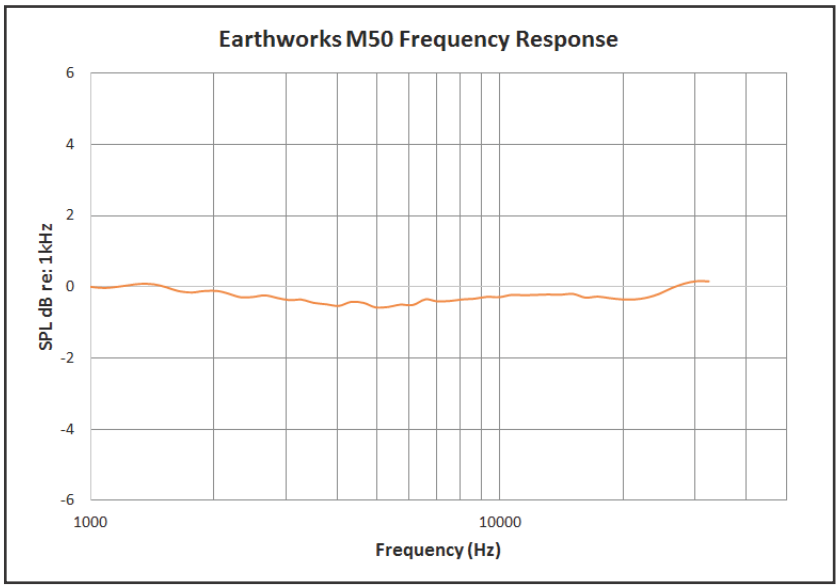


Figure 1: The M50's frequency response was obtained by a gated measurement, stays within a 1 dB window, and corresponds very closely to the supplied calibration curve.

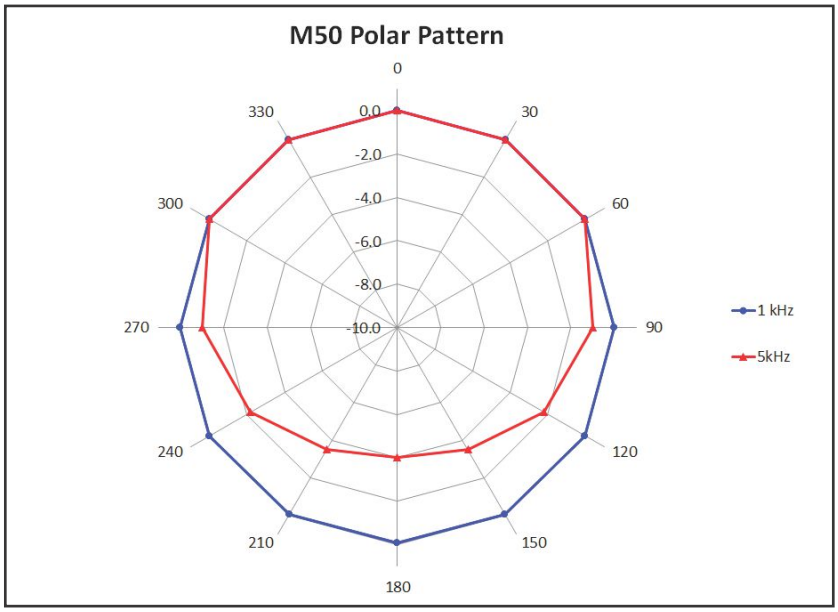


Figure 2: The polar pattern of the M50 shows ideal omnidirectional behavior at 1 kHz and only a 3 dB drop-off at 180° at 5 kHz.

Resources

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I measured frequency response by comparison to a NIST-traceable reference microphone using the same acoustic source (a small full-range speaker flush-mounted in a large absorbent planar baffle) and carefully duplicating the mic position. I do not have an IEC-grade anechoic chamber, but a section of my basement with heavy wall and floor coverings, in conjunction with gating (that is, cutting off the time domain impulse response just before the first major reflection) gives me data very consistent with proper IEC measurements. Gating the impulse response before the first reflections (quasi-anechoic) limits low-frequency data, so I show response only above 1 kHz. The acoustic source outputs little energy above about 30 kHz. So on the high end, my data cut off at that frequency because of the uncertainty. Nonetheless, the response was quite extended to the limits of what I could measure. The frequency response, shown in **Figure 1**, is within 0.6 dB of the calibration data supplied by Earthworks and is reasonably flat, even before applying correction. This is a very good result and confirms that the M50 conforms to the frequency response limits of Class 1 under IEC 61672, at least at normal room temperature (my lab is about 21°C). At higher frequencies, increased temperature could have a significant effect.

To check the sensitivity, I used a Bruel & Kjaer 4231 calibrator with the 0.25" adapter supplied by Earthworks. The calibrator was set to at 94 dB ref 20 µPa, and a 1 kHz excitation frequency. I measured 36.5 mV output from the microphone, almost exactly the number supplied in Earthworks' calibration data (36.0 mV), well within the error limits of the calibrator and Earthworks' measurement.

Figure 2 shows the polar pattern data at 1 kHz and 5 kHz and were taken at 90 dB SPL. The microphone is nearly omnidirectional at lower frequencies, with increasing directionality at higher frequencies because of the microphone geometry. Nonetheless, at 5 kHz, the rear sensitivity is only 3 dB lower than the front sensitivity. This good performance is largely attributable to the small diaphragm and mechanical design of the housing.

Noise level is specified as 22 dB, A-weighted, 20 kHz bandwidth. I measured the noise using my Basement Lab Acoustic Noise Killer (BLANK) setup. This basically involves wrapping the microphone in a plastic bag, wrapping the bag in duct tape to seal it, then burying the wrapped package in about 200 lbs of a mixture of sand and kitty litter in a plastic tub sited on a concrete cellar floor. The tub is covered with thick blankets, and the microphone cable is isolated from any hard surfaces with wadded rags. This setup, though seemingly crude, effectively enables noise to be measured below 5 dB SPL A-weighted

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without using a special isolated anechoic chamber. Next, I measured the noise of the M50 to be 23.1 dB SPL A-weighted, slightly higher than the 22 dB SPL A-weighted specification, but reasonably close. This is relatively low noise for a 0.25" capsule and entirely satisfactory for loudspeaker measurement and front-of-house equalization measurement, but a bit higher than you would want for recording. Earthworks has other microphones available for that application, which have significantly lower noise—though they may be less suitable for metrology.

One should be cautious when comparing distortion data measured acoustically with specifications from some manufacturers. I've noticed in the fine print of some spec sheets that the manufacturer bypassed the capsule completely and the reported distortion numbers were actually from the electronics alone. This obviously does not give a true picture of distortion as the microphone is intended to be used, but can be useful in specification wars for marketing purposes! With this in mind, I only measure microphone distortion acoustically.

Figure 3 shows the distortion vs. SPL. This was obtained using a 6" loudspeaker driver measured near field (about 2 mm distance to the microphone from the cone), with an amplitude-swept sine wave. The measurement frequency was 1 kHz, though lowering or raising the frequency by a factor of two had no effect on measured distortion. My test limit was 130 dB SPL due to threats of divorce—at that level, distortion measured about 0.93%. Using one of my reference microphones, I confirmed that the distortion of the source was under 0.3%, so the measured distortion here really is that of the microphone.

Figure 4 shows the distortion spectrum of the M50 at 120 dB SPL and 1 kHz. The second and third harmonic dominate, but there is a spray of higher-order products as well, though these are low enough in level that they would be unlikely to affect most measurements. If you extrapolate the curve, the 3% distortion point would be expected to be at about 140 dB SPL, which is the specified acoustic limit for the M50.

Wrap-Up

The M50's measurements confirm the manufacturer's claims—moderately low noise, accurate sound pressure measurement, even and extended frequency response, and a good omnidirectional polar pattern. There are competitive microphones that do better in one or two of these areas, but there are none that I'm aware of that offer this particular balance of virtues. For a small test lab, sound reinforcement engineer, or serious loudspeaker designer, this is a compelling product.

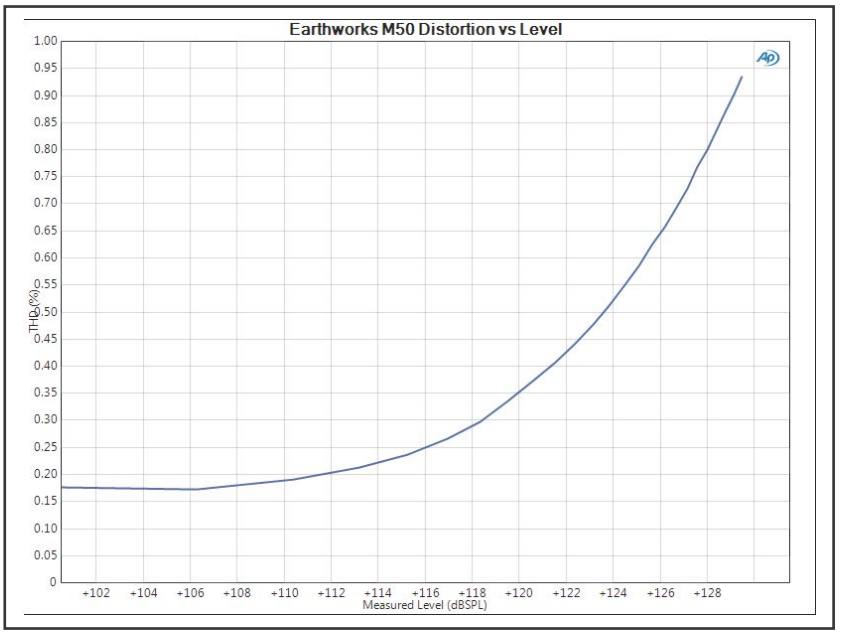


Figure 3: The M50's distortion remains below 1% at levels up to 130 dB SPL.

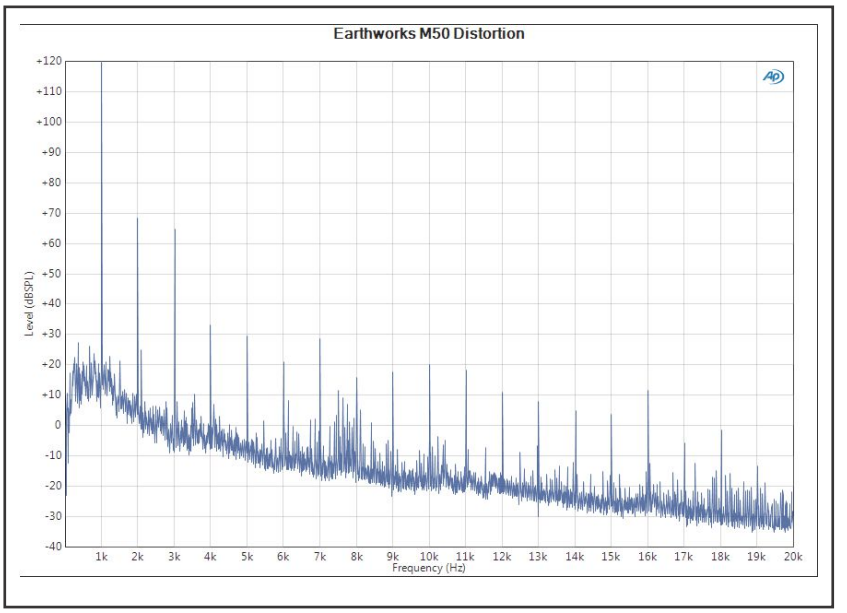


Figure 4: The distortion spectrum of the M50 at 120 dB SPL is under 0.35% and is dominated by the second and third harmonics.

About the Author

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

Benchmark DAC3 HGC Stereo D/A Converter

Benchmark Media Systems, Inc.'s DAC series of high-performance digital-to-analog converters (DACs) have earned the respect of audio enthusiasts and professional users alike for their state-of-the-art performance, including exceptional audio transparency and vanishingly-low levels of noise and distortion. Now Benchmark has made its DAC3 even better with the upgrade to ESS Technology's new ES9028PRO DAC chip.



Photo 1: The 64-pin ES9028PRO digital-to-analog converter (DAC) chip is a drop-in replacement for the older ES9018. Total harmonic distortion (THD) compensation is among the significant improvements in the new chip. (Photo courtesy of ESS Technology)

By
Gary Galo
(United States)

The heart of Benchmark's DAC2 is ESS Technology's ES9018 digital-to-analog converter chip, widely recognized in the industry as one of the best DAC chips and, until now, the best in the view of Benchmark's Vice President and Director of Engineering John Siau. Although several other manufacturers of DACs and stand-alone digital players used the ES9018, Benchmark's implementation ensured the best possible operating environment for the chip. But, ESS Technology hasn't let grass grow under its feet. Last year, it announced a new line of audiophile-grade, 32-bit DAC chips, including the ES9028PRO (see **Photo 1**). Now, Benchmark has recently introduced its DAC3 line of D/A converters based on the new ESS chip (see **Photo 2**).

The ES9028PRO has a 64-pin footprint, and is pin-compatible with the ES9018, enabling Benchmark to drop it into the same PC board used in the DAC2. This makes the DAC3 a cost-effective upgrade. Siau notes that the DAC3 is like a DAC2 on steroids. In addition to the new ESS chip, the software



Photo 2: This is the front view of Benchmark's new DAC3 HCG digital-to-analog converter (DAC), shown with its remote control. The unit offers state-of-the-art performance, and the same flexible array of digital and analog inputs and outputs as its predecessor. (Photo courtesy of Benchmark Media Systems, Inc.)

DAC3 HGC Stereo D/A Converter
Benchmark Media Systems, Inc.
203 East Hampton Place, Suite 2
Syracuse, NY 13206
800-262-4675
www.benchmarkmedia.com
sales@benchmarkmedia.com
Price: $2,195 with remote control



Photo 3: This is the rear panel of the DAC3 HGC. The digital inputs shown on the right include two S/PDIF coaxial, two TOSLINK optical, and one USB 2.0. Analog I/O includes two stereo RCA inputs, two stereo RCA outputs, and stereo balanced outputs. (Photo courtesy of Benchmark Media Systems, Inc.)

and the analog gain structure are different from those in the DAC2.

Like its predecessor, the DAC3 is available in three versions—the DAC3 HGC, the DAC3 L, and the DAC3 DX. The DAC3 HGC reviewed here combines an analog preamplifier with the DAC. There are two stereo, unbalanced (RCA) analog inputs and two stereo, unbalanced analog outputs, along with one stereo balanced (XLR) analog output (see **Photo 3**). All outputs can be controlled with the front-panel volume control, eliminating the need for a separate analog preamp.

Digital inputs include two S/PDIF coaxial, two Toslink optical, and one USB. The USB Audio 1.1 mode supports pulse code modulation (PCM) up to 96 kHz/24-bit and doesn't require any driver on the host computer. The USB Audio 2.0 mode requires the free, downloadable driver on Windows computers, and supports PCM up to 192 kHz/24-bit, along with DSD at 2.8224 MHz in the DoP 1.1 format. No driver is needed for the Macintosh platform. Users who need an AES/EBU (XLR) digital input can opt for the DX version, which doesn't include the analog inputs. The L version is an HGC without the headphone outputs.

DAC3 HGC's Key Features

My 2015 review of the DAC2 DX is still available on the *audioXpress* website (see Resources). And, since much of the technology in the DAC2s and DAC3s is the same, I'm not going to repeat all the details here. I recommend reading the DAC2 DX article as a prelude to this review. The functionality of the HGC, the DX, and the L versions is essentially the same for the DAC2s and the DAC3s and is also covered in the DAC2 DX review. Here's a short list of the key features and technologies that the DAC3 HGC converter has inherited from the DAC2 HGC:

- Hybrid Gain Control (HGC) combining analog and 32-bit digital volume controls for the lowest possible noise floor
- Native Direct-Stream Digital (DSD) conversion
- Advanced USB input using an SMSC USB3318 high-speed USB transceiver and an XMOS programmable microcontroller. Benchmark's Multi-Mode Asynchronous Advanced USB Audio eliminates jitter-induced distortion, offering the convenience of computer playback without the usual compromises
- Texas Instruments (TI)/National LME49860 op-amps for all analog circuitry
- TI/National LME49600 high-current audio buffers on headphone outputs
- Switching-mode power supply, reducing AC line hum to -133 dB even at full output, and -160 dB at idle
- Front panel display of both sampling rate and digital word length

DAC3 HGC Improvements

ESS's Sabre ES9018 DAC chip is now seven years old, and Benchmark began working with the new Sabre ES9028PRO after ESS announced its new chip last October. Measurements and listening tests convinced Benchmark of the new chip's superiority. The block diagrams for the two chips are shown in **Figure 1**, and reveal the ES9028PRO's increased complexity. The ES9028PRO has four key areas of improvement over its predecessor:

- Total harmonic distortion (THD) compensation
- Improved oversampling filters
- Improved phase-locked loop (PLL)
- Improved power supply distribution

Benchmark considers the THD compensation to be the most significant improvement. The 32-bit

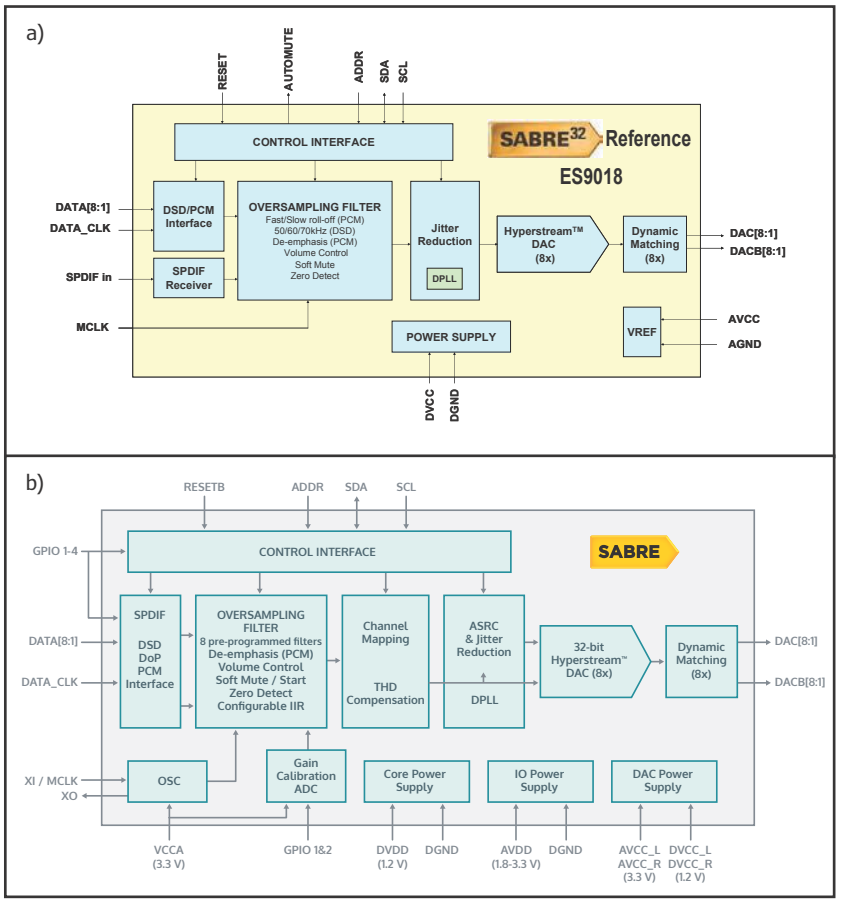


Figure 1: Block diagrams for the ES9018 (a) and ES9028PRO (b) DAC chips. The ES9028PRO's advances include harmonic distortion compensation, and improvements to the oversampling digital filters, the phase-locked loop, and power supply distribution. (Images courtesy of ESS Technology)

digital processing in the ES9028PRO not only compensates for harmonic distortion at the chip's own analog outputs, but it can also be adjusted to compensate for distortion in the post-DAC analog circuitry. The second and third harmonic distortion components can also be independently nulled. **Figure 2** shows the results. The black trace shows second harmonic distortion with the DAC3 in a special test mode, with the THD compensation disabled. The results are similar to those from the DAC2. The red trace shows a significant reduction in second harmonic distortion with the THD compensation enabled. In the midrange, where the ear is most sensitive, the improvement is 8 to 12 dB.

The new ESS DAC chip offers a selection of eight digital filters, either pre-set or user-programmable. Benchmark has implemented its own linear-phase filter, optimized for the lowest passband ripple. Like its predecessor, the DAC3 uses a TI/Burr-Brown SRC4392 sample rate converter, upsampling all digital inputs to 210.9375 kHz. This shifts the digital interpolation filter's corner frequency out of the audio passband—above the Nyquist frequency—even for sampling frequencies of 192 kHz. Benchmark notes:

"This unique Benchmark system prevents the near-Nyquist errors that would normally be produced by the filters in the D/A converter chip. The 211 kHz upsampling system allows us to implement a filter that is optimized for minimum passband ripple instead of using the default filter in the ES9028PRO. The DAC3 and DAC2 both use the Benchmark 211 kHz upsampling system, and both provide outstanding frequency-domain and time-domain performance. But, due to the improvements in the ES9028PRO, the DAC3 has slightly less passband ripple than the DAC2."

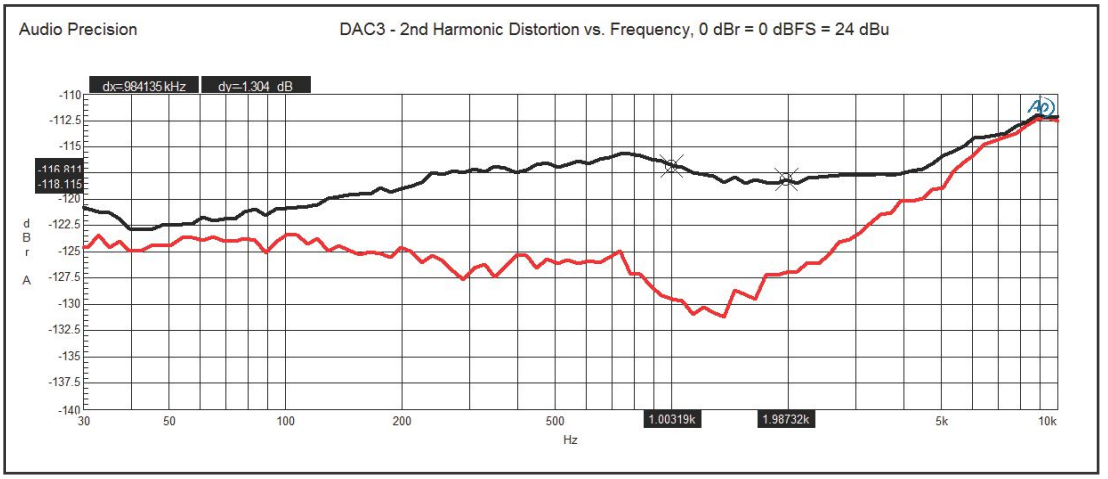


Figure 2: This graph shows the DAC3's second harmonic distortion with the ES9028PRO DAC chip's THD compensation disabled (black) and enabled (red). The THD compensation yields a significant reduction in distortion in the critical midrange, where the ear is most sensitive. (Image courtesy of Benchmark Media Systems, Inc.)

Faster Lock Times

ESS has improved the PLL in the ES9028PRO, giving the new chip faster lock times than the ES9018. Benchmark has used this to good advantage, changing the timing in the UltraLock2 system used in the DAC2 to create UltraLock3 for the DAC3s. Jitter attenuation in the two systems is virtually identical and exceeds the performance of the PLL in the ESS DAC chips. The graphs in **Figure 3** show lock times for Benchmark's three DACs. The lock/unmute time for the DAC1 was 50 mS (see **Figure 3a**), but it took that converter another 2 seconds to settle. In the DAC2, the mute circuits have to be held on for 450 mS while the processor waits for the ES9018 to lock (see **Figure 3b**). With the ES9028PRO, the DAC3 muting time is reduced to around 6 mS (see **Figure 3c**). Both the DAC2 and the DAC3 are fully settled at the end of the mute cycle.

The DAC3's faster lock and settling time makes it possible to conduct A/B comparisons between digital inputs. One obvious application would be to compare an S/PDIF coaxial interface to a Toslink optical, fed from the same digital player. The DAC3 remote control will then enable seamless switching between the two inputs without leaving your listening chair. You can even compare the analog outputs of your digital player as a stand-alone unit, to the DAC3 conversion. However, this comparison won't be meaningful unless the analog output levels of the digital player and the DAC3 are matched.

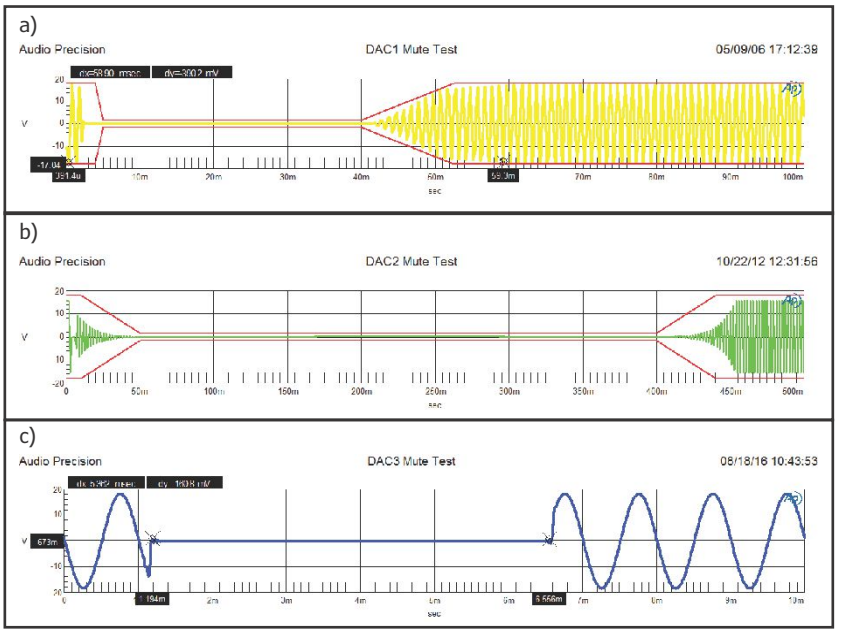


Figure 3: These three graphs show the improved lock time of the DAC3 compared to its predecessors. The DAC1 (a) unmutes in 50 mS, but takes 2 seconds to settle. The DAC2 (b) takes 450 mS to unlock, but is complete settled at the end of the mute cycle. The DAC3 (c) is unmuted and completely settled in 6 mS. (Images courtesy of Benchmark Media Systems, Inc.)



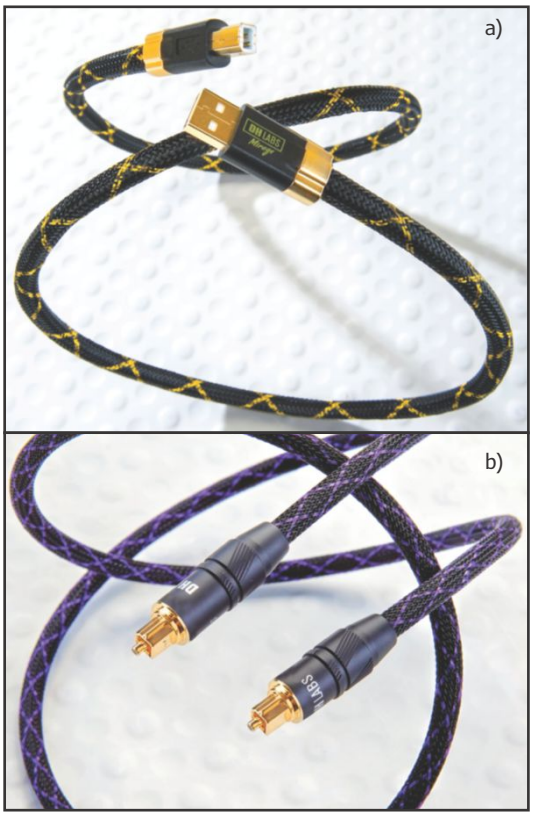


Photo 4: These cables from DH Labs will ensure the best-possible performance from USB and Toslink digital interfaces—the Mirage USB (a) and the Glass Master Toslink Optical cable (b). (Photos courtesy of DH Labs)

Optimum Operating Environment

Although many stand-alone digital players and outboard DACs have used the ESS chips, Benchmark correctly notes that the chip alone is no predictor of performance. The operating environment for the DAC chip is critical and varies widely from one manufacturer to another. As with the DAC2, Benchmark has taken the necessary steps to insure the best possible performance from the ES9028PRO chip. These include:

- 4:1 channel summing, improving the signal-to-noise ratio (SNR) by 6 dB
- External current-to-voltage (I/V) converters, lowering noise and distortion
- Precision differential amplifiers, which remove common-mode distortion
- Advanced power-supply filtering, with very-low-noise voltage regulation (Benchmark's discrete design reduces noise and distortion)
- UtraLock3 jitter attenuation, providing virtually perfect jitter rejection, reducing jitter-induced sidebands to -149 dB
- 211 kHz upsampling, eliminating time-domain errors caused by the DAC chip
- DSP with 3.5 dB of headroom above 0 dB, eliminating clipping of intersample peaks
- Six-layer circuit board with external ground planes, reducing noise and providing RF shielding

ESS also introduced two other DAC chips last year—the ES9038PRO and the ES9026PRO. The ES9026PRO is a drop-in replacement for the previous-generation ES9016S. Like its predecessor, the ES9026PRO is manufactured in a 48-pin package and offers a simpler implementation, albeit with slightly lower dynamic range (124 dB vs. 129 dB for the ES9028PRO), and somewhat higher distortion (-110 dB vs. -120 dB for the ES9028PRO). But, the real confusion is in trying to decipher the differences between the ES9028PRO and the ES9038PRO. ESS calls the ES9038PRO the flagship of the ESS PRO series, claiming a dynamic range of "up to" 140 dB, and THD+N of -122 dB.

The fine print in ESS's Product Brief for the ES9038PRO notes that 140 dB of dynamic range is in the mono mode, presumably with all eight channels connected in parallel. I contacted them for clarification on this and received a reply from Shawn Scarlett, Marketing Director for ESS. He notes that the two chips have the same circuitry. But, the ES9038PRO has "significant portions in parallel for each channel" to optimize noise performance. He further notes that "when customers use the part in

About the Author

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

stereo mode—they are creating a massively parallel system."

ESS's Product Brief states that the ES9038PRO "was designed for premium home-theater equipment including Blu-ray players, preamplifiers, all-in-one A/V receivers, and more." This makes sense. If a single ES9028PRO is used in a 7.1-channel system, only one DAC section is available for each of the eight outputs. The ES9038PRO enables designers of multi-channel equipment to take advantage of the performance improvements of paralleled DACs, with a single DAC chip, just as stereo designers are able to do with the ES9028PRO.

Nonetheless, there are already stereo DACs available that use the ES9038PRO, some for substantially less money than the DAC3 HGC. One, from a company I respect, claims both a dynamic range and SNR of 120 dB. But, the Benchmark DAC3 HGC achieves a 126 dB unweighted dynamic range and S/N figure using the ES9028PRO. As noted above, a manufacturer's implementation of a DAC chip is as important as the chip itself. Benchmark has provided a state-of-the-art operating environment for the ES9028PRO chip. As such, the DAC3 HGC can almost be guaranteed to outperform less expensive products using either of the new ESS chips, where the implementation lacks the level of sophistication found in the DAC3 HGC.

Digital Cables

Although Benchmark's UltraLock3 jitter suppression system is arguably the most advanced in the industry, I still recommend using good quality digital interconnect cables. I use several DH Labs cables, including the Mirage USB and the Glass Master Toslink (see **Photo 4**). The Glass Master Toslink cable is manufactured with 300 strands of high-purity glass fibers to yield extremely wide bandwidth. The glass fibers are terminated with a machined connectors and precision polished optical-grade lenses for low jitter. My own listening has shown that this cable narrows the gap between coaxial S/PDIF and Toslink more than any other Toslink cable I've used.

With this cable, the biggest limitations in the Toslink interface are probably the transmitter and the receiver modules used in the connected equipment. Although the Toslink interface will often operate without glitches at 192 kHz, sound quality audibly degrades above 96 kHz. Benchmark, like other manufacturers offering Toslink interfacing, specifies 96 kHz as the maximum sampling frequency supported by its Toslink inputs.

DH Labs' Mirage USB cable is manufactured with separate shielding for the data and the power

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Canare Corp. of America | www.canare.com

Mirage USB and Glass Master Toslink optical cables
DH Labs | www.silversonic.com/docs/products/USB.html

ES9018 and ES9028PRO DAC chips
ESS Technology, Inc. | www.esstech .com

USB Audio Player Pro V. 3.4.4 for Android 4.0 and up
Google Play Store | play.google.com/store/apps/details?id=com.extreamsd.usbaudioplayerpro&hl=en

High-Resolution PCM and DSD files
HDtracks | www.hdtracks.com

HAECOAX audio embedder
KanexPro | www.kanexpro.com

BDP-105 Universal Digital Player
OPPO Digital | www.oppodigital.com

HD-24PCE HDMI Cable
Pangea | Audio Advisor, www.audioadvisor.com/pangea

High-Resolution files and discs
Reference Recordings | referencerecordings.com

Galaxy Note 4
Samsung | www.samsung.com

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.*

Measurement Tests	DAC1	DAC2	DAC3
SNR—A-weighted	116 dB	126 dB	128 dB
THD+N, 1 kHz, -1dBFS, 20 kHz LPF (dB)	-106 dB	-109 dB	-113 dB
THD+N, 1 kHz, -1dBFS, 20 kHz LPF (%)	0.00056%	0.00035%	0.00022%
Frequency response, 20 Hz to 20 kHz	±0.1 dB	0, -0.04 dB	0, -0.015 dB
Passband ripple	±0.005 dB	±0.003 dB	±0.002 dB
Intersample headroom	0 dB	3.5 dB	3.5 dB
PLL lock time	60 ms	450 ms	6 ms
Jitter attenuation system	UltraLock	UltraLock2	UltraLock3
Converter chip	AD1853	ES9018	ES9028PRO

Table 1: Here is a comparison of the specifications for the three DACs.

supply lines. Each line is double-shielded, making a total of four shields. The cable uses silver-plated, continuous-crystal conductors and a low-density dielectric. DH Labs claims uniform impedance, minimizing reflections, which is critical at the high sampling rates used in high-resolution digital audio. The Mirage USB is the most transparent USB cable I've heard. The Glass Master Toslink and the Mirage USB are priced at $200 each for 1 m lengths. If this exceeds your budget, try the Deluxe Toslink Optical at $45 or the Silver Sonic USB at $70. On a high-resolution system, these will still yield a worthwhile improvement over generic cables.

Listening

My evaluations of the DAC3 HGC were done using my OPPO Digital BDP-105 Universal Digital Player (which I reviewed in a 2013 article, see Resources) as the source for CDs, DVD-Audio, and Blu-ray audio discs and my Samsung Galaxy Note 4 with USB Audio Player Pro software for playing high-resolution media files via USB.

The HDMI 2 output of the OPPO Digital BDP-105 player feeds a KanexPro HAECOAX audio de-embedder, which is powered by my own outboard 5 V linear, regulated supply. The Kanex (which I reviewed in a 2016 article, see Resources) ensures that high-resolution DVD-Audio and Blu-ray audio discs are passed to the DAC3 HGC's S/PDIF input at their native sampling rate. The OPPO Digital player's own S/PDIF output is down-sampled to 44.1 or 48 kHz, when instructed to do so by the DVD or Blu-ray media. I used a Pangea HD-24PCE HDMI cable from the OPPO Digital player to the KanexPro, and DH Labs D-75 coaxial, fitted with Canare 75 Ω RCA connectors, from the KanexPro to the DAC (the D-75 has been replaced with D-750).

Since last year, my reference DAC has been the DAC2 HGC, so making comparisons between the two was very easy. The DAC's balanced outputs feed a pair of Benchmark AHB2 power amplifiers, one for each channel. (See Resources for information about my 2015 *audioXpress* AHB2 power amplifier review.) The AHB2 amplifiers power my Audio Concepts Sapphire III/Sub-1 loudspeaker systems in a passive bi-amp configuration. Between the feed-forward error correction in the AHB2 and the THD compensation in the DAC3 HGC, this combination might yield the lowest measureable THD and noise of any digital playback system.

The DAC3 HGC retains all the virtues of its predecessor, including its remarkable transparency and a seemingly non-existent noise floor. There's also a sense of effortlessness and a complete lack of strain even on the most complex musical scores. But, the most obvious areas of improvement are in the reproduction of inner detail and soundstage delineation.

In my 2016 review of the $10,000 Bricasti M1 Special Edition (see Resources), I noted that what set this DAC apart from others is how it "reproduces the acoustic space between individual instruments and sections of a symphony orchestra. The musicians are connected by that space with an uncanny level of realism." That's what I'm hearing in the Benchmark DAC3 HGC compared to the DAC2.

By better delineating the acoustic space in which the musicians perform, localization of the musicians is rendered with greater dimensionality, which adds to the impression of greater detail. The bass performance is also improved, mainly in resolution and control. On Reference Recordings' 176.4 kHz/24-bit files of Igor Stravinsky's *Song of the Nightingale* (RR-70), the delineation of the various bass drum attacks is noticeably improved. It's not just a matter of changes in volume, but also the subtle differences in timbre and weight.

A smartphone is a convenient method for media file playback, although there are certainly more exotic and more expensive ways of playing high-resolution files. The results are not quite as smooth and detailed in the treble as DVD-Audio discs made from the same files, with the OPPO Digital BDP-105 feeding the DAC3. But, the smartphone via USB is very good, nonetheless.

HDtracks is now selling the DSD files that were used to master the RCA Victor Living Stereo SACDs made by Analogue Productions. The DSD files for the Reiner/Chicago Symphony recording of Béla Bartók's *Concerto for Orchestra* are more richly textured and three-dimensional than the SACD played on the OPPO Digital BDP-105 as a stand-alone player

(the OPPO Digital BDP-105 must be used as a stand-alone player for native DSD conversion of SACDs).

At this level of quality, we're often splitting hairs when comparing the best DACs on the market (see **Table 1**). Some might be described as leaning toward ambience and warmth, while others appear more analytical. The improvements I've observed in the new DAC3 HGC are consistent with Benchmark's design philosophy of making the most accurate, transparent equipment possible. What the DAC3 does is simply extract more of the information already on recordings. The improvements are readily audible on both CDs and high-resolution sources, and the end result is as warm and as musical as the source—no more and no less.

Benchmark has an interesting application note on its website titled "How Should I Buy an Audio System?," which delves into the issue of euphonic coloration vs. transparency. Its author John Siau,, acknowledges that one piece of euphonic equipment in an otherwise transparent chain may be desirable to some listeners. But, the end result, however pleasing and musical, can't be described as totally transparent.

Many of my reference recordings are digital remasterings of early stereophonic material from the mid-1950s through the mid-1960s, from labels including Mercury Living Presence, RCA Victor Living Stereo, and British Decca (formerly called London Decca in the US and Japan). Though these recordings captured many outstanding performances in superb sound, I admit that these recordings also have some euphonic coloration from the analog tape and tube electronics. I prefer to play them on a transparent system, so I can hear an accurate reproduction of those recordings—the DAC3 HGC/AHB2 combination provides that.

With the DAC3 HGC, Benchmark has taken an already remarkable product and moved it to the next level. The performance improvements over the DAC2 HGC are well worth the relatively modest increase in cost, and Benchmark's DX and L versions offer the same performance upgrades in configurations to suit a variety of audio enthusiast and professional needs. The DAC3 HGC ensures Benchmark Media's continued leadership in high-resolution digital audio, and I highly recommend DAC3 HGC.





Noise in Dining Spaces



It is notoriously difficult to have a conversation in most restaurants due to a lack of acoustical design.

Public spaces, in particular restaurants, are inherently difficult places to control acoustics. To those about to build or remodel such a venue, our acoustical expert offers some recommendations.

By
Richard Honeycutt
(United States)

A couple of decades ago, my wife and I joined my son and his date for dinner in a national chain restaurant in Winston-Salem, NC. The food was fine but communication was difficult due to the intense noise created by customers talking in a room with only acoustically reflective surfaces. We probably would have discussed how badly this restaurant needed my services as an acoustical consultant had not the act of discussion been so challenging.

A few years later, I was visiting a client in New Orleans, LA, and stopped at a famous old restaurant that had a giant dining room with plaster walls and ceiling, wooden floors, un-upholstered wooden chairs, and not many customers at the time. With some difficulty, I ordered my food and somehow the waiter understood the order correctly, despite the strong reverberation and flutter echo.

Subsequently, in a city near my home, I found another chain restaurant whose Chilean sea bass is extraordinary. But my wife and I have had to stop going there because the "background" music levels are so high that we had to shout to the waiter, who had to shout back to us. Needless to say, conversation over dinner was a non-starter.

Restaurant Sound Levels Vary

I have been told that restaurant consultants advise their clients that the best sound level for a restaurant depends upon the clientele: The younger the clientele, the louder the restaurant should be. I do not remember the original source of this information, nor have I seen the research (if there was any) upon which this conclusion was based. However, I usually suspect over-generalizations such as this one.

I did, however, find more information in Jess Thomson's article in the August 2014 edition of *Seattle Magazine* (http://www.seattlemag.com/article/noisy-restaurants-how-much-noise-too-much).

> "Sound is a double-edged sword in restaurants," says [Ethan] Stowell, whose nine Seattle restaurants are known as being among the city's loudest. "Having a reasonable noise level is crucial for business. A restaurant has to be loud enough, because no one goes to restaurants that are too quiet. But of course, every owner faces noise complaints. And you don't want your clientele to be uncomfortable."

Stowell doesn't design his restaurants with an optimum sound level in mind, but waits to see how the sound situation plays out once a restaurant opens. For example, at Red Cow, his steak frites spot in Madrona, WA, Stowell says he used previous experience to assess the noise level from the day it opened. "We work backwards with sound protection," he explains, referring to the sound-absorbing paneling many restaurants install in an effort to strike the right balance. At 8:30 p.m. on a Friday night, he wants to hear a loud din that reflects a spot's popularity, but hates the kind of

sharp noises that interrupt conversation.

Eric Hentz, owner of Mallet, a Seattle, WA-based design-and-build firm with restaurant clients, says no owner is actually designing a restaurant to be noisy, though they want it to be busy. Owners have finite budgets and they need to navigate a complex set of choices when building restaurants. Their first priority is to open; they often need to come back after the fact and address any acoustic problems. "They want [the noise level] to generally appeal to a wide demographic, but project budgets are such that people tend to cut sound attenuation out of the design approach very early on when they see how much it costs," he explains.

Hearing Challenges

While acoustical absorption is certainly not free, there are many options available, and a good consultant can help a client control noise in a restaurant and still stay within budget. Certainly the ability to converse over dinner is important to many people, not just the over-50 bunch.

In the April 2017 issue of *New Yorker*, David Owen writes:

"Two years ago, in Dallas, I went out to eat with a large group in a noisy restaurant. I was seated near one end of the table and couldn't hear what anyone at the other end was saying. I assumed that that was because I was the oldest person there, but then a young guy sitting across from me asked the young guy sitting next to him whether he could hear anything. 'No,' he said. 'I'm just nodding and smiling.' A little later, one of the young guys cupped his hands around his ears, like an ear trumpet, and the other asked whether doing that helped. He said that it helped a lot, so we all tried it—and, indeed, I found that I could even home in on particular speakers."

Owen, who is himself hearing-challenged, brings up a valid point: those who have trouble hearing are even more negatively affected by noisy dining spaces. This is especially true of those who wear hearing aids, as only the most expensive aids avoid boosting the noise to an uncomfortable degree when the user is in a space having high ambient noise. While at first blush, this might seem to support the "louder-for-the-younger" scheme of restaurant design, we do well to remember that about one-fifth of all hearing aids sold in the US are purchased by the Department of Defense. Many of those are used by younger servicemen and women who have recently been discharged.

Another danger of noisy dining spaces is illustrated by school cafeterias. In research presented to the North Carolina Regional Acoustical Society of America (ASA) chapter in 2001, and referenced in an *ASA Scitation*,[1] Joseph H. Bridger found that some school cafeterias actually exceeded Occupational Safety and Health Administration (OSHA) guidelines for noise exposure. Also, concerns had arisen that teachers and other school personnel could suffer from vocal abuse due to having to shout to communicate in the cafeterias. Conditions have significantly improved since then, and most good architects use the services of acoustical consultants in order to keep noise levels in school cafeterias and other social spaces controlled.



Multipurpose venues offer special acoustical challenges.

Church fellowship halls, community centers, and rental event spaces present particular challenges with regard to noise. On the one hand, these are usually multi-purpose spaces, and they may need to support good acoustical projection when used for theatrical, concert, lecture, or other events. But they also need to keep noise under control when



Acoustical absorption can be used as a design element. (Photo courtesy of Armstrong Ceiling and Wall Solutions)



This is another example of acoustical absorption used as a design element. (Photo provided courtesy of Armstrong Ceiling and Wall Solutions)

used for dining, reception, and other social events. Meeting these two needs requires a careful balance between hard surfaces to provide helpful reflections and absorption to control noise.

For many years, acousticians believed that by designing for the proper Reverberation Time (RT), they could achieve a good acoustical result in any room. (If your only tool is a hammer…) While RT is important, in that high RT means that sound stays around a long time and continues to build up, it is not the entire story.

In a noisy room, people unconsciously tend to speak more loudly in order to be heard. This, in turn, makes the room noisier, so the people (all 300 to 400 of them in some cases) speak even more loudly. This vicious circle continues until speaking loudly enough to be heard becomes physically uncomfortable. What is needed is to design the room so that it has enough acoustical absorption for the maximum number of occupants. If this is done, the room will often be deader than the ideal for concert events, though it would work pretty well for theatrical presentations.

So far, we have considered only people talking, which is indeed the predominant noise source in most dining spaces. But street noise entering through open doors or single-glazed windows can add to the "social noise." Background music can also be a problem, as illustrated by my avoidance of the sea bass restaurant mentioned earlier. And noise also serves a useful purpose: table conversation is more comfortable when the participants have a sense of privacy. To a certain extent, controlled background music or noise can help with this.

Acoustical Solutions

So finally we get to the bottom line: How do we control noise in dining spaces? There are several important considerations to keep in mind:

- The space must incorporate the proper amount of acoustical absorption for the maximum occupancy. This is really a signal-to-noise-ratio (SNR) problem. We must keep the noise at or below the level of typical table conversation. Actually, our ear/brain mechanism is so awesome that we can communicate quite comfortably with the noise level 1 to 2 dB higher than the conversation level.
- The absorption must be spread around evenly enough to prevent acoustical hot spots, such as at bars, serving windows, and so forth.
- Care must also be taken with regard to noise from sources external to the room: street noise, kitchen noise, etc.
- Music levels should be set by policy, not at the whim of individual staff members.
- The least expensive place to put acoustical absorption is on the ceiling. In rooms using a blacked-out ceiling, the absorption can also be painted black, rendering it unnoticeable. (This only applies to certain kinds of absorptive materials, as many are not paintable without losing their absorptivity.)
- Acoustical absorption can be used creatively if it is treated as a design element.
- It is far less expensive to design and build any space with good acoustics in mind than to try to retrofit later. To twist the old saying, "there's never enough money to do it right, and there certainly may not be enough money to fix it!"

The best advice is that wise restaurateurs should secure the services of a good acoustical consultant near the inception of the design phase for a new or renovated restaurant.

Reference

[1] J. Bridger, "School cafeteria noise—The impact of room acoustics and speech intelligibility on children's voice levels," *The Journal of the Acoustical Society of America,* Volume 111, Issue 5, http://asa.scitation.org/doi/abs/10.1121/1.4778394.

Resources

D. Owen, "High-Tech Hope for the Hard of Hearing," *The New Yorker*, April 2017.

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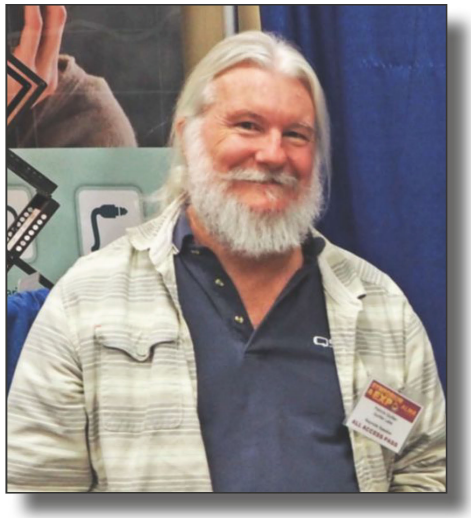
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Industry Icon Goes Back to His Roots

An Interview with Pat Quilter of Quilter Laboratories



Pat Quilter founded QSC Audio Products and upon his retirement 43 years later, he and several partners started Quilter Laboratories, hoping to fulfill Quilter's lifelong goal of developing a modern lightweight guitar amp that "thinks it's a tube amp."

By
Shannon Becker
(United States)

SHANNON BECKER: Tell us how you got interested in audio design.

PAT QUILTER: I was inspired to go into the audio business by my younger brother Matt Quilter, who was getting started as a rock musician in the heady days of the late 1960s. As the nerd of the family, I decided to take over the world with high power guitar amps. There were no really satisfactory choices for hard rock when we started in 1968. The "old school" amplifiers by Fender and Vox are treasured today, but they didn't "go to 11" like we wanted in the psychedelic era. The rock-god amps such as Marshall amps sounded awesome, but would often fail during a single show. And none of the early solid-state amps did overdrive well, although some had a decent clean tone. So I set out to make a high energy amp that "cranked" and could be played at full power all day long. I figured you could just design all the components to handle full peak power and it should never fail. How hard could it be?



Pat Quilter is shown circa 1970. A time when QSC started to deliver its first amplifiers to other audio manufacturers.

I had been experimenting in college, and slowly worked my way up to a homemade 5 W amp and speaker, built into a small cabinet, running on 12 V batteries with a tuner and tape player—a homemade boom box before such things became common. This taught me the rudiments of electronics. When my brother started his high school band, his bass player couldn't afford a name-brand amp. I had just read an article in *Radio-Electronics* magazine about a power boosting technique called "bridging." I looked up the available transistors and determined I could get 100 W using this method. Of course, all of this work was based on transistor technology—"everybody knew" that tubes were on the way out.

SHANNON: Tell us about your first company, Quilter Sound Company? What did you want to accomplish?

PAT: I wanted to provide amps with controllable overdrive and extremely dynamic performance—like

hot rods or race cars. But the show must go on so the amps had to be reliable. Amps have a definite peak power so it seemed like all I needed to do was make sure the speakers and power devices could continuously take that power. That part worked out pretty well, but there were still mystery failures, and musicians were complaining about shortcomings in my tone. We all had a lot to learn in those days, but people showed me now-classic tube amps that had the sound they wanted—if only they weren't so heavy and fragile! I finally built a few experimental tube amps to see what was really going on. This was very instructive, although at the time I didn't have the engineering chops to fully implement these behaviors in solid state. Meanwhile, as the 1960s progressed into the 1970s, guitar amps were becoming ever cheaper and less well defined, and we even wondered whether the electric guitar had run its course. By now I had several working partners trying to make a success of our little business, and we decided we needed a broader and steadier market. We took stock of what we had learned in the school of hard knocks. The power section of the guitar amps was the hardest to get right, so we decided to exploit this knowledge and become a power amp specialist in the pro-audio market. We left Quilter Sound behind and adopted the more corporate sounding "QSC Audio Products."

SHANNON: What was the biggest difference between the Quilter Sound Company and QSC Audio?

PAT: Since guitar amps are designed to produce tone, even today, people need to "try before they buy." We were small with very few dealers, so this was a real hindrance. But power amps are supposed to be clean and predictable, with objective performance measurements. You can buy them off the spec sheet. This enabled us to reach more customers and begin a long and steady campaign to extend our lineup and earn a reputation for reliability and value.

SHANNON: Do you have a favorite design? Can you share some of the challenges you faced while designing it?

PAT: High power transistors throw off a lot of heat during normal operation, so they need a cooling system—commonly called a heatsink, which absorbs the heat and dissipates it, sometimes with the help of a fan. The transistors must be electrically isolated, but thermally coupled to a common cooling surface so they all run at the same temperature. Unfortunately, materials that insulate electricity also block heat flow. So we had to provide each power transistor with a thin mounting film, covered with thermal grease on both sides to improve thermal coupling, and clamped down with insulated mounting screws. There can be 8 to 12 power devices in each channel of a large amp, so all of this finicky assembly work took time and money, and the thermal grease often dried out over time, causing the devices to run hotter. Early transistors suffered from thermal fatigue from hot-and-cold cycling, which resulted



This is an early advertisement for Quilter Amps—the Duck Amps instrument amplifiers—from 1968. The artwork for the ad was admittedly "shamelessly ripped off from Gilbert Shelton's 'Freak Brothers' comic strip with new dialog" added to it.



Pat Quilter demonstrates one of his earlier Series Three designs at a trade show in the 1980s, making serious inroads into the concert sound and installation markets.



Pat Quilter ,with QSC's co-founders Barry and John Andrews, celebrate the company's 40th anniversary during the National Association of Music Merchants (NAMM) Show 2008. And yes, that's a cake... not real amps!

in unpredictable failure. Not good! In one of my few legitimate "Aha" moments, I realized that you could sort of turn the amp inside-out so the power transistors were grounded instead of the power supply. This meant we could quickly and reliably screw them directly onto a metal heat sink with ideal metal-to-metal contact. This greatly boosted our production rate AND our long term reliability, and some of these 1980-era "Series One" amps are still playing today in clubs and movie theaters. Combined with some clever packaging ideas, these became our first fully competitive, yet affordable, power amp line.

SHANNON: How has the technology changed since you first started designing amplifiers?

PAT: For a couple of decades, 1970-1990, there were steady improvements in the quality and performance of power transistors, and with careful packaging, fan cooling, and other tricks, the power of a "big" amp increased from less than 500 W to maybe 2000 W. However, this required larger and heavier power transformers to supply the internal power, and amplifier weight became an increasing limitation. In the mid-1990s, QSC developed a particularly robust and cost-effective lightweight replacement for the AC power transformer, which we used in our PowerLight and PLX series amps. We also added circuitry that reduced the heat loss by 50% or more, so we could package more power in a smaller chassis. This enabled us to reach almost 4000 W with a fraction of the weight, and these amps remain some of the cleanest and most reliable on the planet.

But in the last decade, "Class D" has become more feasible. This is a more complicated method of producing high power audio signals, with new tricks to learn, but when properly done, the heat loss is greatly reduced, which not only allows for smaller, lighter packaging but saves on energy costs. This allows amp modules to be packaged inside speaker cabinets and other non-traditional spaces with obvious advantages. The leading pro-audio companies are now reasonably fluent in this technology, but it has only begun to impact guitar amp design. Of course, this technology is one of the key features of our Quilter amplifier program.

SHANNON: How does your design process work?

PAT: Typically we try to identify gaps, either in our line-up or customer needs, which we can meet with our skill set. These are the areas where we can add value and offer products that enhance our customers' lives.

SHANNON: How have your techniques evolved?

PAT: At the start of my career, I took a sort of perverse pride in going back to the earliest roots of electronics and seeing if I could re-invent the basic elements of circuit design. This extreme hands-on technique gave me a good feel for whether things want to work or not, but to make real progress you eventually have to stand on the shoulders of the giants who have gone before you. And so, as lab instruments improved I had better tools for identifying performance problems, and as computer-aided design and simulation became more affordable, this also became a great enabler, especially for the more complex designs we now found ourselves doing. The ability not only to speed up work, but re-cycle portions of successful designs, is a great aid in meeting customer needs with faster

Pat Quilter is pictured at Prolight+Sound 2008 with some of QSC's best selling pro audio amplifiers.



design cycles. But interestingly, I still reach for my pencil and paper when I really want to THINK about a circuit idea.

SHANNON: What do you know now that you wish you had known when you started?

PAT: In a very real way, it's a good thing I didn't know how long it was going to take to achieve success. I probably wouldn't have had the gumption to go for it! But once you're knee deep in the swamp, you have no other choice but to press on and try to work through the problems. So fortunately, I would say that this allowed success to come at a rate that I could absorb without losing my footing or going under.

SHANNON: Quilter Laboratories has been in business since 2011. Why go back to your roots with guitar amplifiers and what products are you most proud?

PAT: I got into audio because I wanted to support my brother's career, and by extension, all performing musicians. Full-service electric guitar was the art form that blossomed in my formative years, and it combined two of my passions—making harmony and high-performance technology. And so, the guitar amp was my first love, if you will. Power amps are an essential tool and a great challenge, but the mission is to be a perfectly clear glass of water—and I wanted to provide a richer menu.

Modern technology, especially digital technology, has greatly expanded the range of tones and capabilities you can pack in an amp. But for whatever reason, I don't feel much "soul" in these products. Maybe it's the need for a screen-driven interface, or maybe it's because the digital programmers didn't experience the dawn of high-energy amplification, but the results are notoriously rather cold and sterile. So there's still a big demand for the classic tube technology that the electric guitar grew up with, but we reached "peak tube" around 1960 and tube quality has been going slowly downhill ever since. "Back in the day" we expected a set of tubes to last for years, but now they sometimes go bad in a week. So my goal is to use my knowledge of tube behavior to capture the same dynamic and tonal responsiveness, using modern, stable, mass-producible analog electronics and intentional fine-tuning of the key factors.

At Quilter Labs, I'm proud of all my "children," for different reasons. I personally gravitate towards the simple-but-effective designs like our 101 Mini Head or Block-series amps, but the Mach 2 Micropro is a



Quilter Labs was at the National Association of Music Merchants (NAMM) Show 2013, where Pat Quilter met other musicians and held impromptu jam sessions.



The Quilter Pro Block 200 is is the miniature-sized Tone Block 200 head, a 200 W, all analog, solid-state head.



The Quilter 101 Mini Head is an incredibly light, yet powerful amplifier.



The Quilter 8" Mach 2 combo is the ultimate grab-and-go performance amplifier. The smallest and lightest of the Mach 2 combos (19 lbs), it features a potent high power pro audio driver.

strong statement of how flexible and multi-faceted we can make analog technology that "just sounds good."

SHANNON: You have had a very successful career, but instead of retiring, you opted to start Quilter Laboratories. How does this company differ from you other business ventures?

PAT: I want to have some fun, and maybe do something more creative, with the skill set I developed at QSC. I have always enjoyed making stuff and I want that to continue. My QSC career was indeed very rewarding both from a financial as well as technological perspective, but I spent 40 years designing ever-more-intricate and refined power amplifiers, which is a deep and narrow specialty, and not really the product I expected to be known for, although I feel quite satisfied with my body of work. But when spring rolls around and the sap rises, my thoughts return to the project I never quite finished—getting world-class guitar amp tone using modern high efficiency circuitry. So now I am having fun, along with a small collection of partners, using my design skills to improve tone and functionality for the performing musician. And we have many happy customers, so we must be doing something right!



Pat Quilter accepts the ALMA Beryllium Award at AISE in 2016.

SHANNON: Tell us a little about being awarded the ALMA Beryllium Award in 2016 and the Parnelli Audio Innovator Award.

PAT: Pro audio power amplifiers are pretty much a "backstage" product category with little public exposure. They are essential to concert sound, but you pretty much get all of the responsibility with none of the glory. So yes, it was very nice to be recognized for a long and productive career. I certainly didn't do it all by myself, but I feel proud that QSC has steadily worked its way up to its position as a leading pro-audio supplier, while treating people right and doing good honest business in a challenging but interesting field.

SHANNON: What's next for you and Quilter Laboratories?

PAT: If there's one thing I could hope for, which is of course rather "aspirational," I would love to bring new excitement and income opportunities to the live music industry. Musicians don't really get paid any more than they did 50 years ago, and that professional devaluation extends down to the equipment providers, who have to struggle to meet price points despite the impact on quality and local sourcing, because the market can't afford to pay what good stuff really costs. So the big squeeze results in a worse experience for everyone, music is not supported in our schools, and the world becomes a somewhat bleaker place.

My hope is to establish a business foundation with our mainstream amplifiers that we can build on with more "wild card" projects that bring some new possibilities to our art. I grew up with rock music - I love the energy - but the last thing I would have expected, when I started in 1968, was that we'd still be hearing those same tunes on the radio 50 years later!

Build your dream system

You Can DIY!

Improving the Stax SRM-T1 Electrostatic Headphone Amplifier

Explore ways to improve the drive capability and the sound quality of the Stax SRM-T1, but be aware that the modifications involve potentially lethal high voltages so this is not the best project for a beginner.

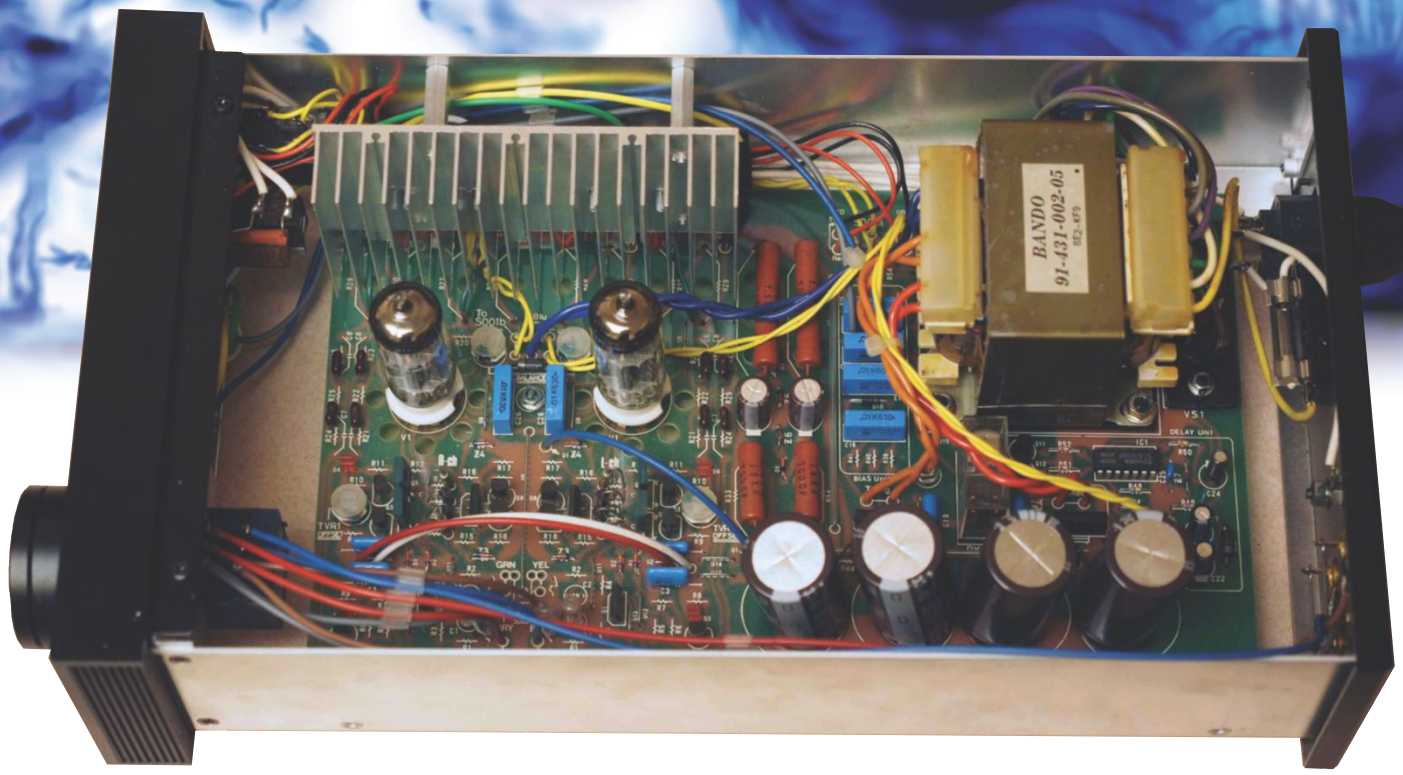


Photo 1: The modified Stax SRM-T1 has increased drive capability and sound quality.

By
James Lin
(United States)

Recently, I purchased a used Stax SRM-T1 electrostatic headphone amplifier "as-is." Now, I already had an amplifier—the SRX Plus described in an article I wrote for *audioXpress* in 2015. However, it never hurts to have a backup, and it came at an irresistibly low price, so I took a chance.

The T1 was first released in 1987, updated to the T1S with balanced XLR inputs in 1993, and to the T1W with extra switching facilities in 1994. It was reintroduced as the SRM-006t in 1999. All the T1 variants and the earlier SRM-006s have output sockets for both the standard 6-pin (230 V) and the Pro 5-pin (580 V) bias headphones, but the current SRM-006tII only has pro bias sockets.

In a visit to the Stax factory in 2012, it was reported that the T1 was used as a base amplifier for designing headphones until the late 2000s. However, that has recently changed. Headphone amplifier guru Kevin Gilmore commented that it "is actually a really good unit, but it uses 6FQ7s, which really cannot withstand the 600 V. So as it gets louder and louder, the sound becomes more restricted, and…the amount of high-frequency energy slowly disappears." The consensus is that it is somewhat underpowered

for the more difficult to drive Stax headphones (e.g., my SR-007), sounding somewhat dark and lacking in detail, with flabby bass. The modifications I made on the Stax SRM-T1 (see **Photo 1**) are outlined in this article and will significantly increase its drive capability and sound quality. However, it involves potentially lethal high voltages and DIYers should proceed with caution.

Basic Design

The T1 is a hybrid amplifier with a DC-coupled topology from input to output, a differential FET input stage, balanced transistor intermediate amplification stages, a tube output stage, and overall feedback from the output stage back to the input (see **Figure 1**). The high voltage supply is unregulated, while the input stage has a Zener shunt regulated supply. All the modern Stax hybrid amplifiers share this topology, differing only in the output stage.

The output stage is a differential amplifier with the outputs taken off the tube plates. The T1 and the SRM-006 uses one dual triode 6CG7/6FQ7 tube per channel with 66 kΩ plate resistors and a B+ of 320 V. The limited edition SRM600 released in 2009 used one ECC99 tube per channel with 60 kΩ plate

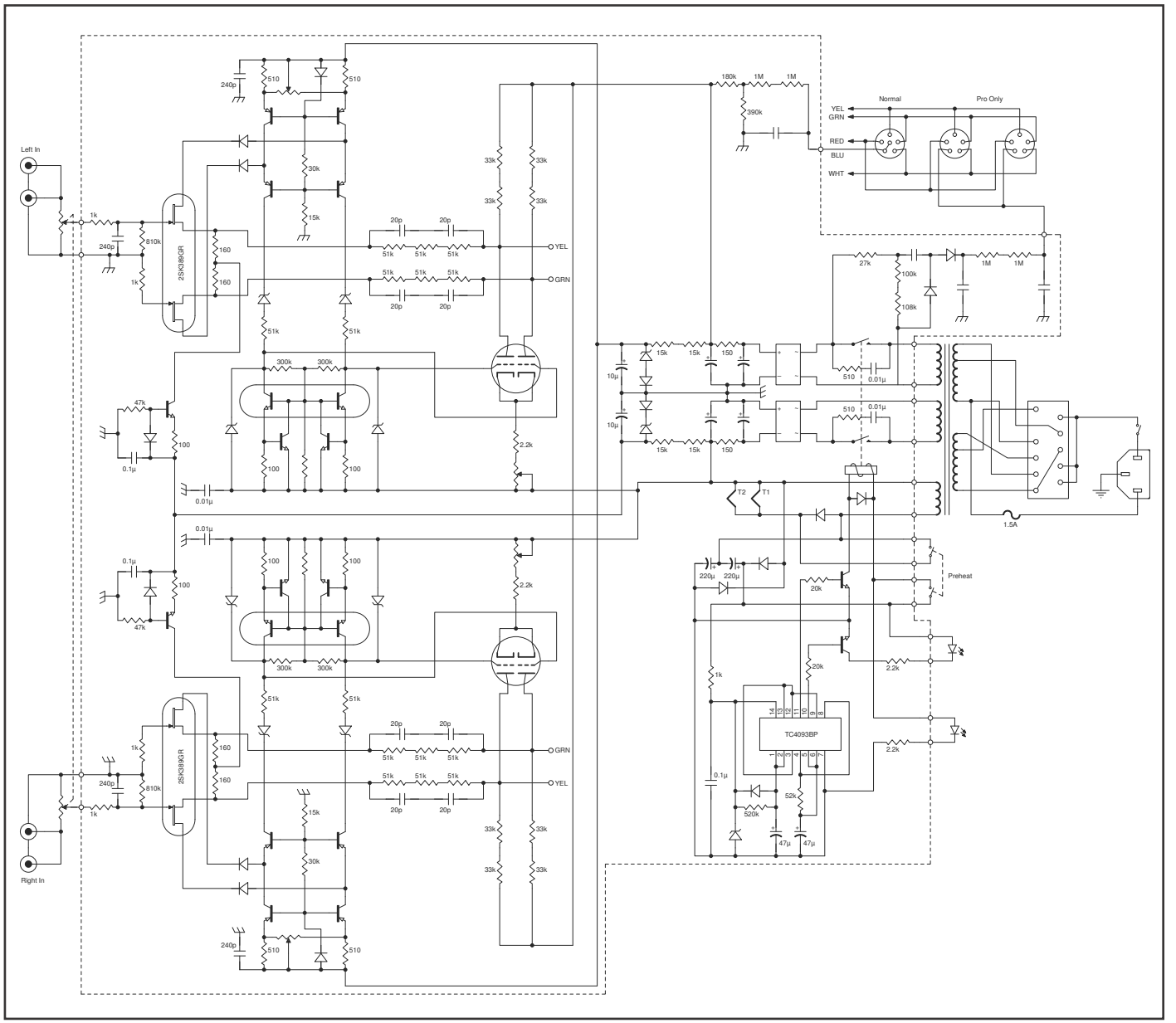


Figure 1: This is the schematic of the Stax T1 electrostatic headphone amplifier. The modifications are limited to the output stage, which is just to the left of middle.

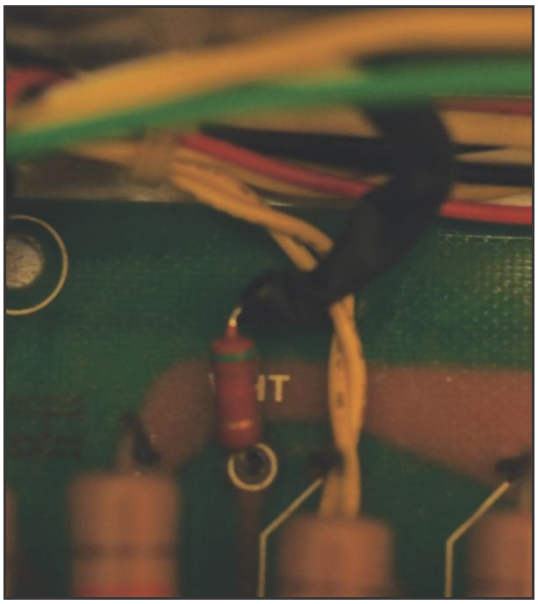


Photo 2: This is one of four output safety resistors installed on the circuit board.

resistors at a B+ of 350 V. The SRM-007tII, released concurrently with the SR-007 headphones, uses two 6CG7/6FQ7 output tubes per channel, with paralleled tube sections, driving 47 kΩ plate resistors at a B+ of 350 V.

Although the 6FQ7 tubes are run at close to maximum plate voltage, they are only at 55% of maximum power dissipation in the T1, which should result in long tube life. The output stage has two trimmers to adjust the offset and balance between the plate sections to get the output voltages to 0 V with respect to ground.

Refurbishment

Electrolytic capacitors deteriorate with time. Because the T1 is 24 to 30 years old, I recommend replacing all the electrolytic power supply capacitors before powering it up the first time. This is basic maintenance. There are four large high voltage power supply capacitors and six lower voltage caps. Since the 1990s, electrolytic capacitors have become smaller for the same capacitance, so the power supply can be improved by replacing the original capacitors with higher capacitance values, as long as the original voltage rating is maintained. This by itself may improve the sound simply by restoring the amplifier to factory specifications or better. The Parts List shows the original and the replacement values in my amplifier.

To avoid confusion, I removed and replaced the capacitors one at a time—be sure to install them in the same polarity. Then, power up and make sure it is working. It is always a good idea to confirm a unit is functioning before modifying it. After replacing the capacitors, my unit worked perfectly.

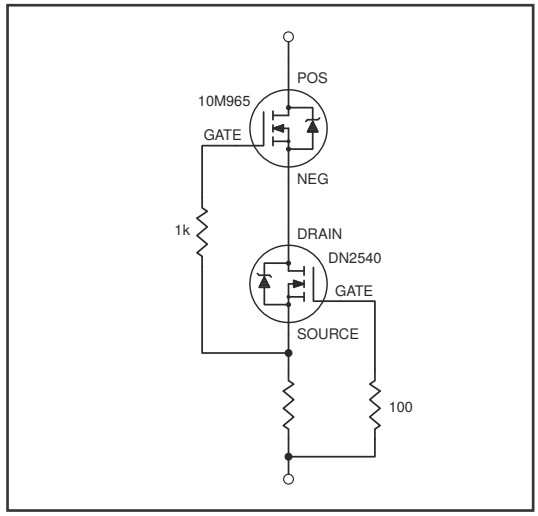


Figure 2: Here is the schematic of the cascode MOSFET constant current source (CCS). The unmarked resistor has to be chosen individually for each current source, or it can be replaced by a 261 Ω resistor plus a 100 Ω trimmer pot.

Modifications

My modifications are limited to the output stage, to improve safety and drive capability while decreasing distortion.

First, add 5.1 kΩ/500 V resistors between each tube plate and each output socket terminal. This can be done at the circuit board by desoldering the green, yellow, white, and red output wires from the circuit board to the sockets and interposing the resistor (see **Photo 2**). These safety resistors are present on all current Stax amplifiers to protect the headphones and the listener in the event of a catastrophic output stage failure.

Generally, amplifier output stages produce the most distortion because that is where the voltage and current variations are the greatest. Stax specifies the maximum output of the T1 as 300 VRMS at 1 kHz. With the SR-007 producing 100 dB with 100 VRMS at 1 kHz, the T1 should be able to drive them to 110 dB, which would be plenty loud. Although the sonic flaws of the T1 have been blamed on the relatively puny 6FQ7 output tubes, the most significant problem is the use of plate resistors in the output stage.

With plate resistors, any signal voltage requires a corresponding signal current to be consumed by the resistor, according to Ohm's law. Electrostatic headphones are inherently a high impedance load, but the plate resistors must be significantly lower in impedance in order to provide adequate signal current to drive the headphones.

As a result, the vast majority of the signal current is always wasted driving the plate resistors, while another fraction is consumed by the feedback resistors, with only a small percentage of the total current produced by the output tubes actually driving the headphones. Thus, at its specified maximum

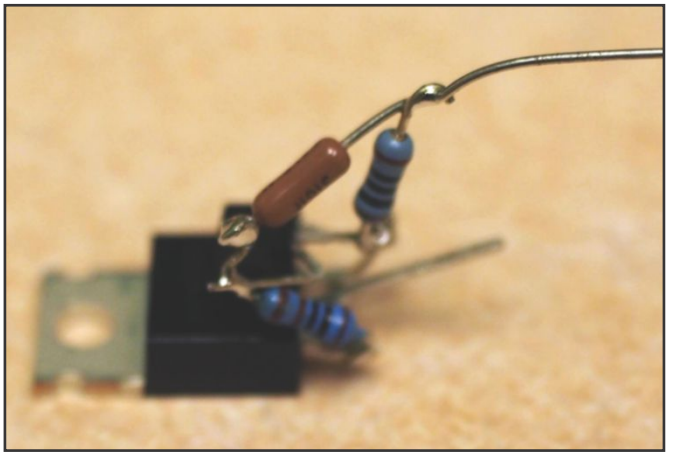


Photo 3: The constant current source (CCS) is hard wired—the blue resistors are the gate stop resistors and the brown resistor sets the current.

output of 300 VRMS, 6.4 mA peak current is required to drive the plate resistors, plus another 2.8 mA for the feedback resistors, a total of 9.3 mA. With a standing current of 9.7 mA, 0.5 to 3 mA at most remains for the headphones. If you're wondering how 9.7 mA minus 9.2 mA equals 3 mA, remember that electrostatic headphones somewhat resemble a capacitor so the current drive to the headphones can be up to 90° out of phase. No output device sounds its best when it is strained to the limit. If we eliminate the current burned up in the plate resistors, the drive capability, distortion, and sound quality should be improved.

The solution is to replace the plate resistors with a good constant current load. This decouples the signal current and

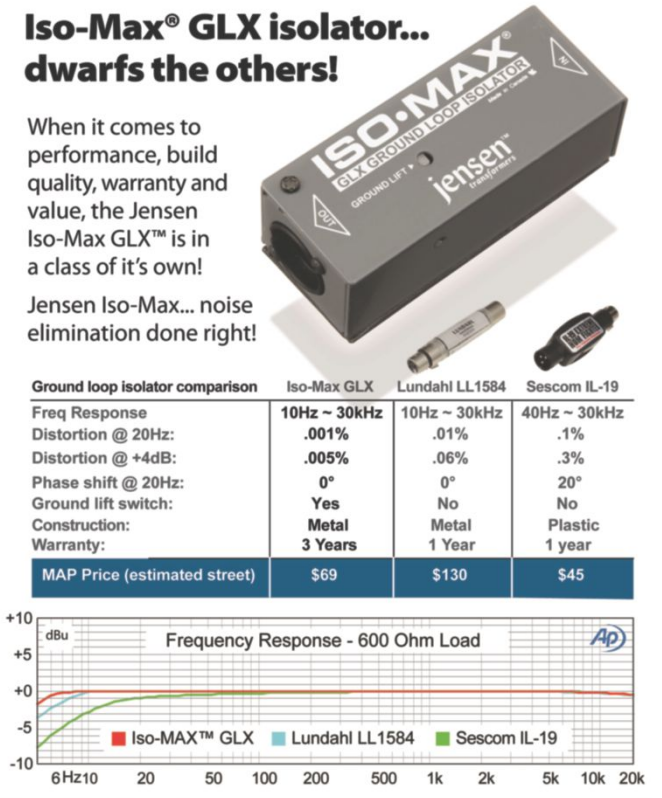
Parts List

Power supply (original values in parentheses)
4 – 330 (100) µf/400 V electrolytic capacitors
2 – 33 (10) µf/50 V electrolytic capacitors
2 – 47 µf/35 V electrolytic capacitors
2 – 270 (220) µf/10 V electrolytic capacitors

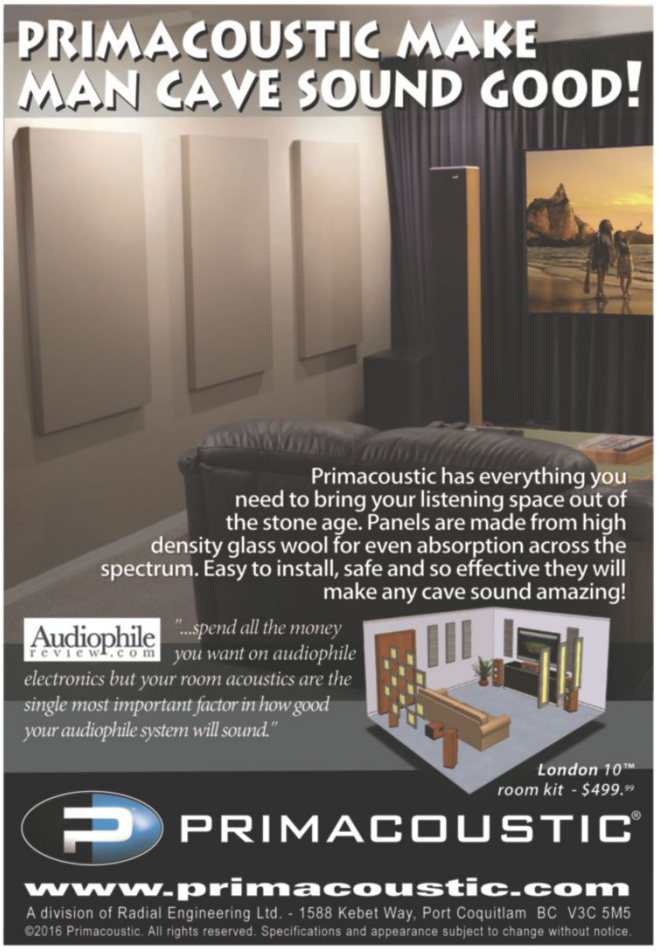
Output section (both channels)
Resistors (0.25 W unless otherwise specified)
4 – 100 Ω gate stop resistor
4 – 300 Ω approximate (current setting resistor, see text, adjust value to get 4.9 mA) or 261 Ω resistor plus 100 Ω trimmer pot
4 – 1 kΩ gate stop resistor
4 – 5.1kΩ/2 W/500 V resistor (safety resistors for the output stage)

Active devices
4 – 10M90s TO220 FET current source
4 – DN4250 N-channel MOSFET

Other
1- finned heatsink approx. 4" × 2.5" × 1"
4 – high voltage ceramic TO220 insulator
Heatsink compound
4 - 4-40 ½" screws for mounting the 10M90S FET
4 - #4 Isolating washer
4 – 4-40 1" standoffs
4 – 4-40 3/8" screws for mounting heatsink standoff to side wall







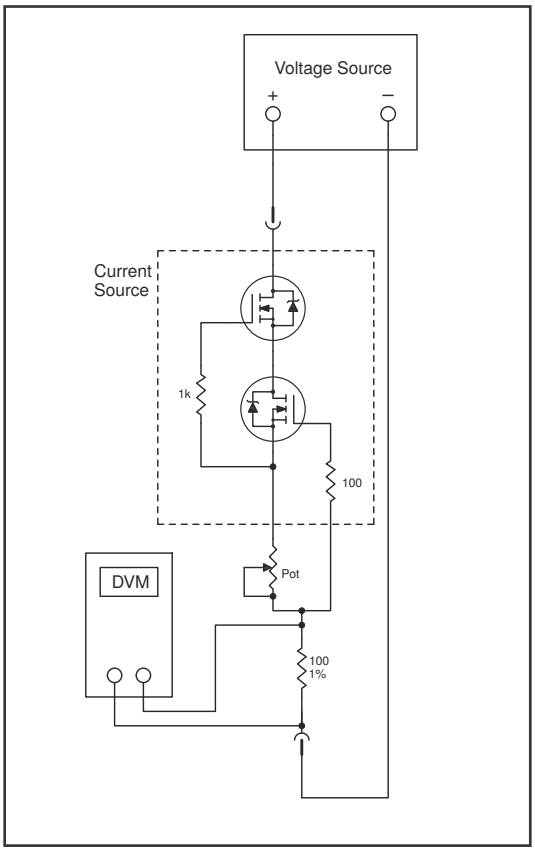


Figure 3: The testing jig is used to determine the source resistor for the DN2540 to set the current.

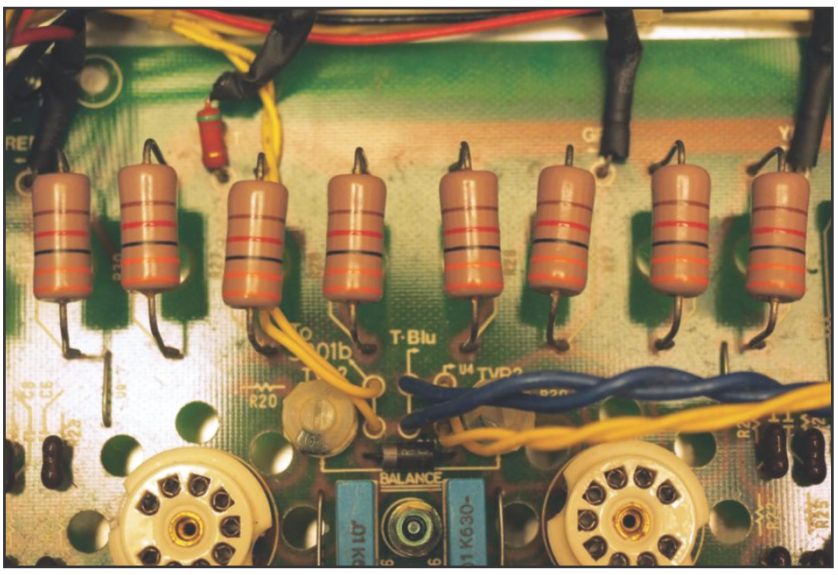


Photo 4: These eight plate resistors will be replaced by four current sources.

voltage, enabling the output tube to swing its entire voltage amplitude without consuming any appreciable signal current, thus relieving the strain on the output tubes. Constant current output loads are used in a number of electrostatic amplifier designs including Stax's contemporary solid-state designs.

High voltage N-channel MOSFETs make excellent constant current loads. You can read about MOSFET CCSs and loads in Walt Jung's *audioXpress* articles (see Resources). Cascoded constant current loads using MOSFETs provide excellent performance with only two MOSFETs and three resistors.

The measured DC impedance of the constant current loads in this modification is more than 150 MΩ, so each output tube will swing the full voltage output of the amplifier while consuming less than 3 µA—less than 0.1% of the standing current. The constant current load sets the operating conditions of the output stage, but is practically invisible to the output tube—all it sees is the headphones and feedback resistors. At the stock T1's same 300 VRMS maximum specified output, 2.8 mA peak current is consumed by the feedback resistors, leaving 6.8 mA to 9.2 mA of standing current available to drive the headphones, much better than the stock output stage. In other words, using constant current loads has converted an amplifier for driving plate resistors into an amplifier for driving headphones.

With a B+ voltage of 320 V and a plate resistance of 66 kΩ, the standing current for each tube section is 4.9 mA. This will be the nominal value for our current load. Each current load dissipates over 1.5 W, so a heatsink is mandatory.

The schematic of the current load is shown in **Figure 2** and is identical to the current loads used in the SRX Plus. The DN2540 MOSFET alone makes a decent current load with an impedance of a few hundred kilohms. It sits between the gate and the source of the 10M90S MOSFET, which swings the voltage. The gate-to-source voltage of the 10M90S stays relatively constant with large voltage swings, largely isolating the DN2540 from overall voltage variations, resulting in a much higher impedance for the cascode pair. Gate stop resistors on both MOSFETs prevent ultrasonic oscillations, and the source resistor of the DN2540 sets the current. It can be built as a simple hard-wired module (see **Photo 3**).

DN2540s can vary, so the source resistor has to be individually chosen for each current load. I used a simple jig with a voltage source, a variable pot as the current setting resistance, a 100 Ω sense resistor and a digital multimeter (DMM) to measure the current (see **Figure 3**). I adjusted the pot to get 0.49 V across the sense resistor, then measured the pot resistance, soldered in the closest 1% resistor,

and confirmed the current measurement again. For my four current loads, the current set resistor varied between 270 Ω and 340 Ω.

If you don't want to mess with hardwiring an individual resistor for each current load, you can build all four current loads on a piece of perf board, using a 261 Ω resistor plus 100 Ω trimmer pot on the DN2540 source terminal. Use the jig to adjust the pot to set the current to the proper value, and attach the perf board and 10M90S FETs to the heatsink.

Due to the high impedance, the value measured at 15 to 20 V will be the same as at 320 V in circuit. The current loads are mounted on a heatsink with a high voltage ceramic TO220 insulator. There are two wires coming from each current load. The wire coming from the 10M90S will go to B+, and the wire coming from the DN2540 source resistor will go to the plate of a 6CG7/6FQ7 tube section.

Each current load replaces two 33 kΩ plate resistors. These are the eight large resistors that are sitting in a row, elevated off the circuit board (see **Photo 4**).

Out of the 16 terminals for the plate resistors, identify which eight terminals will be used by looking at the back side of the circuit board. Four of the resistors are connected together—these are the B+ terminals and will be connected to the wires going to the 10M90S positive lead. Four of the resistors will have terminals connecting to the green, yellow, red, or white wires going the output socket. These are the plate terminals and will be connected to

About the Author

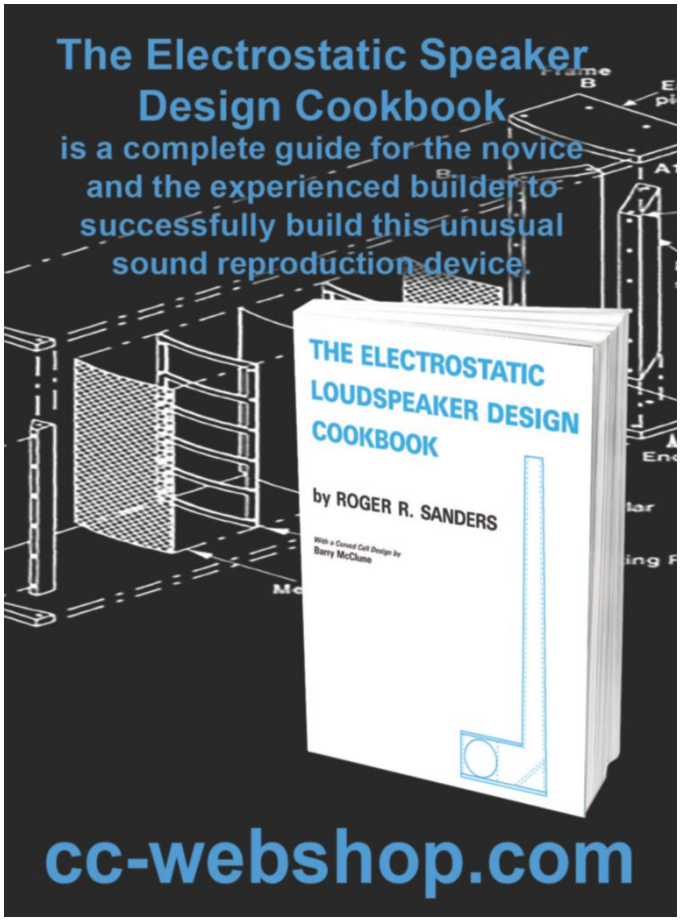
James Lin became interested in DIY audio by reading *The Audio Amateur*. He has written modification articles for *The Audio Amateur, Speaker Builder*, *Glass Audio*, and *audioXpress*. He recently retired from the Veterans Administration in Albuquerque, NM.

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———, "Sources 101, Audio Current Regulator Tests for High Performance: Part 1 and Part 2," *audioXpress*, April and May, 2007, www.audioxpress.com/article/Sources-101-Audio-Current-Regulator-Tests-for-High-Performance-Full-Article.

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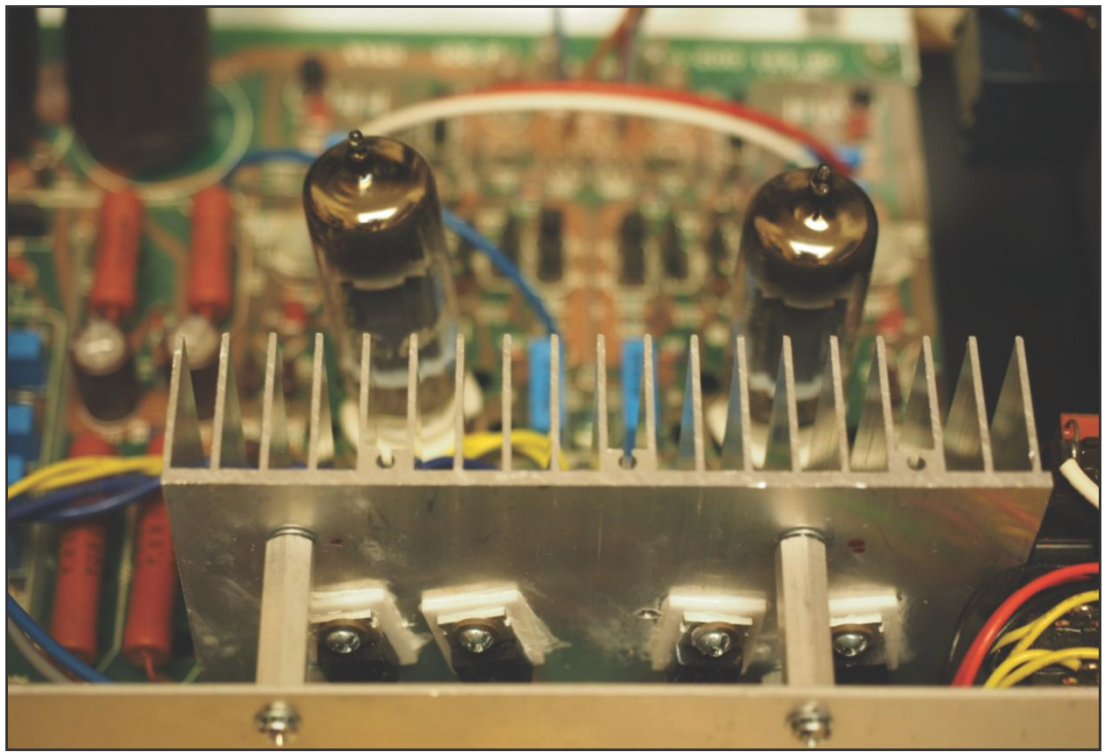


Photo 5: The heatsink with constant current sources in place of the plate resistors.

the source resistor of the DN2540. The other eight terminals connect between pairs of plate resistors and will not be used.

Remove all eight resistors to clear a space to mount the heatsink, insert the wires from each current load into the appropriate terminals, mount the heatsink to the adjacent side wall with standoffs, and then solder the wires in place on the circuit board (see **Photo 5**).

When you plug in the amplifier and turn it on, you will need to adjust the output voltages to zero.

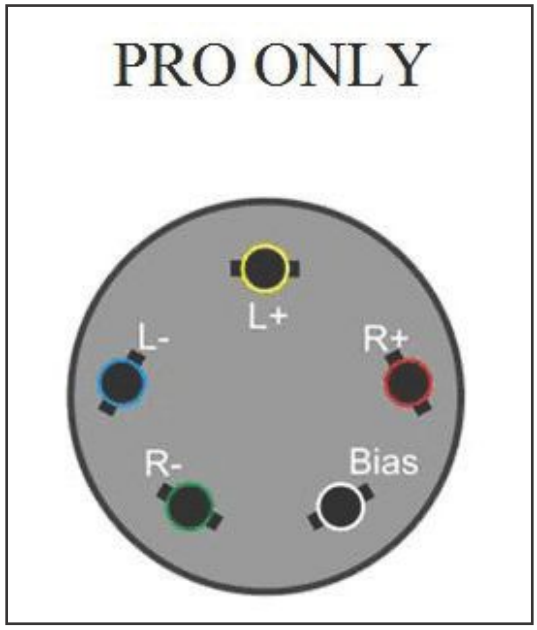


Photo 6: Diagram of the Stax output socket, from the front side

This is checked by inserting a DC voltage probe into the appropriate output terminals on the headphone socket in front (see **Photo 6**). There is a ground terminal at the back of the chassis that you can use as a reference point.

There are two adjustment pots for each channel. The right channel is closest to the front panel and the left channel is behind it. TVR1 (offset) adjusts the voltage between the positive and negative outputs of the channel. TVR2 (balance) adjusts both output voltages up or down in tandem to bias the tubes and is located close to the output tubes. Use a plastic handled screwdriver to make any needed adjustments. First adjust TVR1 by putting your probes in the + and – terminals of the same channel and adjust to 0 V, then remove one of the probes and attach it to the ground terminal to adjust the balance control close to zero. Then do the same for the other channel.

The offset adjustment is somewhat finicky, so go slow. The offset is pretty stable after the first few minutes, but the balance will change as the amplifier warms up. I do a preliminary adjustment after 5 to 10 minutes, then put the cover on and allow the amp to warm up to steady state for an hour or so, and readjust. Adjust the output voltages as close to zero as possible, but it doesn't pay to be too compulsive about it, as there will be some variation with warm up, with changes in line voltage, and as the tubes age. Stax specifies offset and balance voltages within ±15 V of each other or zero.

I had problems getting one channel zeroed due to a faulty pot and tube socket. Once they were replaced, the adjustments went smoothly. With the amplifier warmed up, the heatsink temperature was around 55° to 60°C.

Once the output voltages are adjusted, you can close up the unit, attach the source and the headphones, and enjoy!

Other Changes

The safety resistors and current loads are by far the most important improvements for this amplifier. Some modifiers have also replaced the inputs with better quality RCA jacks and upgraded the input wiring and the volume control.

The volume control is a 50 kΩ Alps RK27 with two sections (four sections in later models, which allow balanced inputs) and concentric shafts to adjust the volume independently in each channel. The knobs pull off. The front part of the volume knob is connected to the smaller concentric shaft, and the rear part to the larger shaft. They are friction linked, enabling it to act as a volume control when they are turned together, and as a balance control by holding the rear knob and independently turning the front part. If you decide to replace it, you will lose the balance function and need to replace the volume knob itself, so there are pluses and minuses to that modification.

Conclusion

These modifications can be done to any T1, T1S, or 006T. In the T1W, a heatsink mount will need to be devised. These modifications should be able to be adapted to the SRM-600 and SRM-007T with appropriate size current loads and heatsinks, but I have not actually done this.

So, how does it sound? Compared to a stock unit, the modified T1 retains the character of the original but sounds cleaner, clearer, more neutral, and more powerful with better bass punch and control, particularly as you crank up the volume.

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.*

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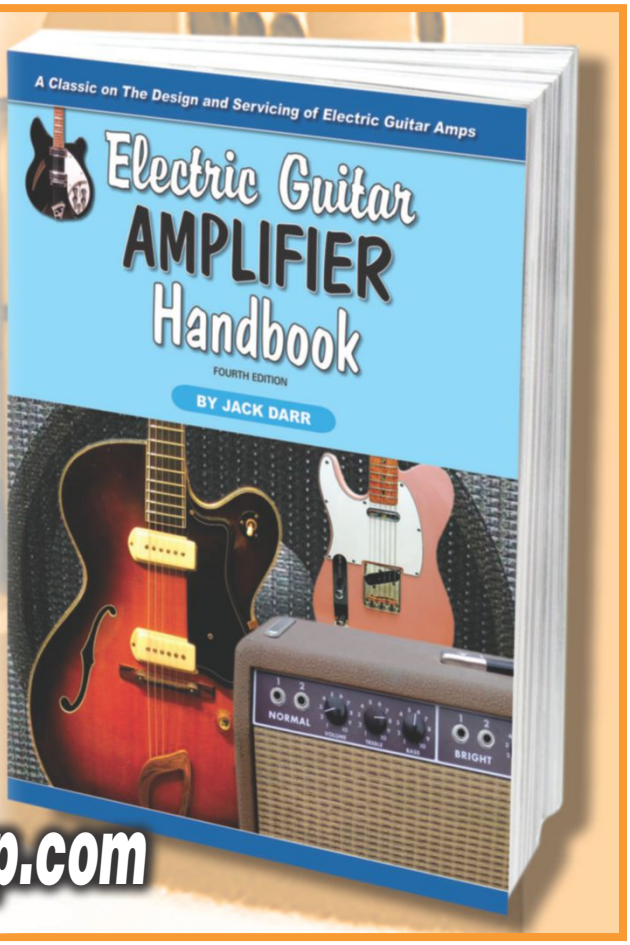
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Cherry Bomb—A VFET SET Amplifier (Part 2)

Construction and Listening Tests

Learn more about this VFET SET amplifier, which was built after this author discovered two Yamaha 2SK77 transistors buried in his parts bin. In this issue, we will share more information about the design of this unique amplifier and discuss the results after testing and listening to it.

By
Thomas Perazella
(United States)



Photo 1: This is the painted chassis with mounting and feed through holes

The Cherry Bomb VFET SET amplifier's driver stages are not quite conventional. Quiescent current is set through the action of Q201 and D201. Q202 and Q203 provide the positive and negative drives to the VFET. Q204 is the basis of another current control that has some negative feedback to prevent conflicts withQ201. It is the control element driven by the servo used to set the quiescent current of the VFET.

Output Stages

Being single ended, the output stage is relatively simple if unconventional. There are a number of protection devices considering the rarity of the VFETs. Drive to the gate of the VFET Q205 is provided by current through R211. Its value was chosen to be low enough to provide sufficient current to overcome the gate capacitance while being high enough to work with the first protection device D209. That diode is a Zener that limits potentially destructive reverse source to gate voltages. The second protective device is an N channel power MOSFET Q206 that is biased in the off condition by connecting the gate to the source and connected directly across the VFET drain to source. It essentially acts as a very robust power Zener to protect the VFET from reverse voltages or excessive forward voltages. That particular device, an IFRB3307Z, has a forward voltage rating of 75 V. Although the avalanche voltage is not specified, it is generally about 10% to 20% higher than the working voltage. That would place the avalanche voltage at 90 V

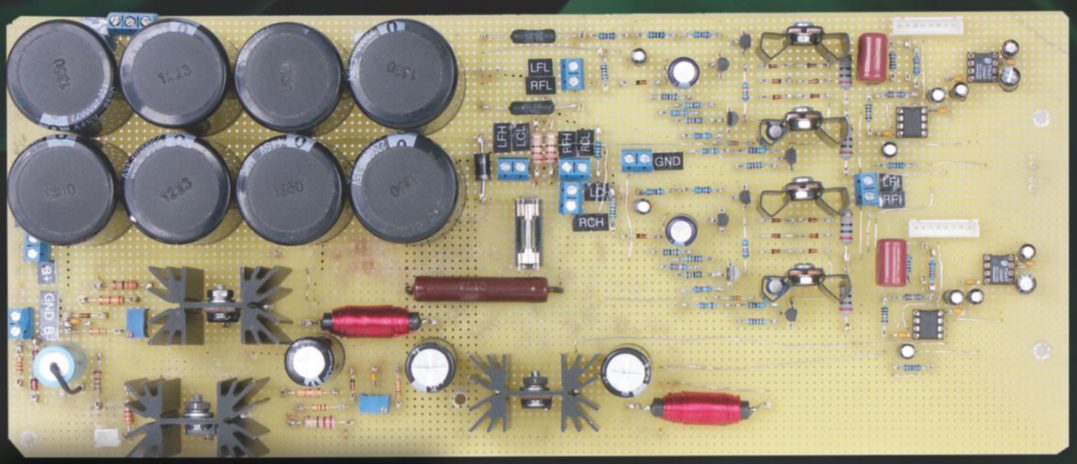


Photo 2: Here is the top of main circuit board.

maximum, which is well below the 200 V rating of the VFET. It also has an internal source to drain diode with a forward voltage drop of 1.3 V. This combination will protect the VFET against any conceivable over-voltage problems in either direction.

To protect against excessive current and power dissipation in the chokes and VFETs, both chokes are fed through F2, a 6.2 A fuse. If something were to go wrong with a servo and the current drive to a VFET were to spike, the fuse would blow. The current rating of the VFET is 20 A. As far as power dissipation, if one of the servos were to fail, the current in the other VFET would still be 2.5 A, leaving a balance of around 4 A before blowing the fuse. That would result in a dissipation of around 85 W in the VFET. With a thermal impedance of the Delta 441 heatsink at around 0.6ºC/W and an estimated junction-to-case thermal impedance of 0.5ºC/W with that oversized case for a total of 1.1ºC/W, the junction temperature would reach about 94ºC—well within the rated 150ºC maximum allowed. Note that the transistors are directly mounted to the heatsink with thermal compound to reduce the case-to-sink thermal resistance. Any insulating material that is reliable and not toxic has a thermal impedance of around 1ºC/W, which would essentially double the resistance and junction temperature so none was used. This fuse solution proved to be much simpler and more reliable than all the active solutions we tried. Two other passive components were necessary, R9 and D4. They provide a safe path for discharge currents of the power supply capacitors if the fuse should open. As a bonus, we added R8 and LED1 to provide an external indication of a blown fuse.

Construction

Since this was a one-off project, the construction materials and methods had to allow for flexibility. For this type of work, the circuit boards are constructed using FR4 glass epoxy boards punched with 0.042" holes on a 0.1" grid pattern. For most of the smaller components, especially where values might need to be changed, we used push-in terminals so that connections could be made from the top side of the board. With the number of changes that were made during the design of this amp that capability was invaluable.

We chose a conventional aluminum rectangular chassis with a bottom cover. The dimensions are 17" × 13" × 3". Mounted in the rear of the chassis are the input power IEC-type receptacle with internal fuse holder, input RCA and XLR connectors, speaker connectors, level controls, and a switch to select balanced or unbalanced inputs. The front of the chassis has the power switch, a power LED, a blown fuse indicator LED, and the VU meters. The top and sides of the chassis have holes for the chokes, transformers, and heatsinks and other miscellaneous parts.

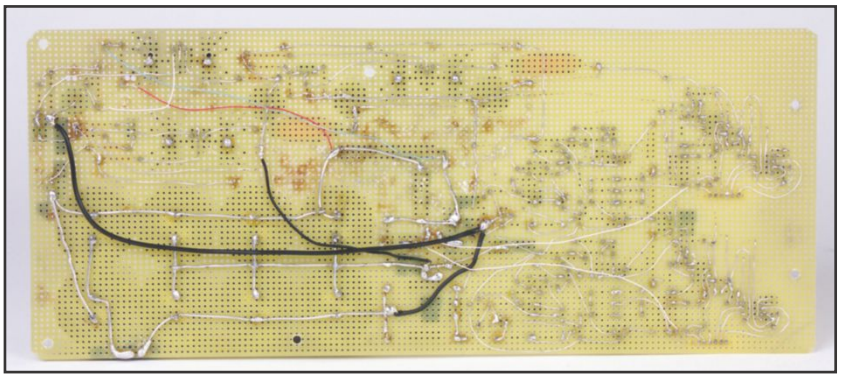


Photo 3: This is the bottom of main circuit board.

To make this conventional chassis unconventional, I decided to try an attention-getting finish. In the period when the original VFETs were used, Hot Rodders often used bright color finishes on their cars with heavy metal flake compositions. I purchased several samples of auto touch-up paint that were similar to Candy Apple red with metal flakes, but most were too subdued for such a small chassis. Then at a crafts store, I ran across a family of spray paints from Krylon called Glitter Blast. One of the colors that proved irresistible to me was called Cherry Bomb. It is dark red with lots of large metal flake, which made it really stand out on a project chassis. Every time someone saw the completed amplifier they asked about the finish and it shortly picked up the moniker Cherry Bomb. **Photo 1** shows the painted chassis with holes.

Circuit Boards

Originally, we tried to put most of the components on one circuit board. With the changes necessary in the power supply, it eventually became necessary to split some of the components for the two small power supplies to a second board. The main board contains most of the components that are not directly



Photo 4: The VFET was mounted on the heatsink.

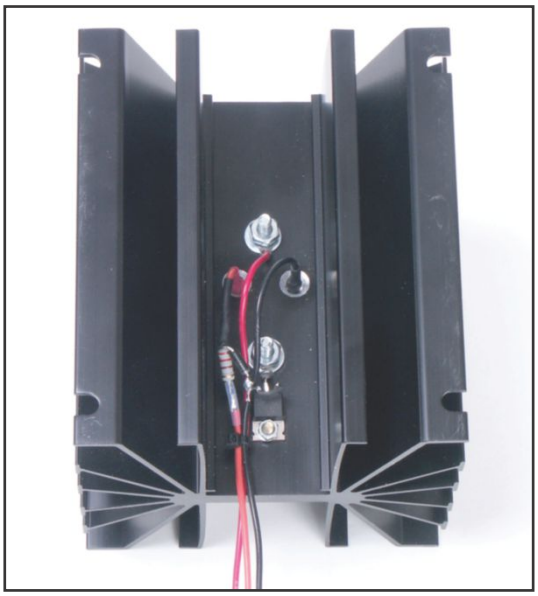


Photo 5: The wiring of VFET and protection MOSFET and Zener has been completed.

chassis mounted including the large electrolytic filter capacitors of the main power supply, the various power supply regulators, the input and driver stages, some of the output components, and the VU meter circuit. The second board, which is mounted to one of the inside sides of the chassis, contains the two small power supply transformers, rectifiers, and first stage filter capacitors. **Photo 2** shows the top of the completed main circuit board in its final configuration, but before the addition of the VU circuits. **Photo 3** shows the bottom of the main board, again before the addition of the VU components. Surprisingly, considering the extensive design changes that were made during development, few jumpers were needed.

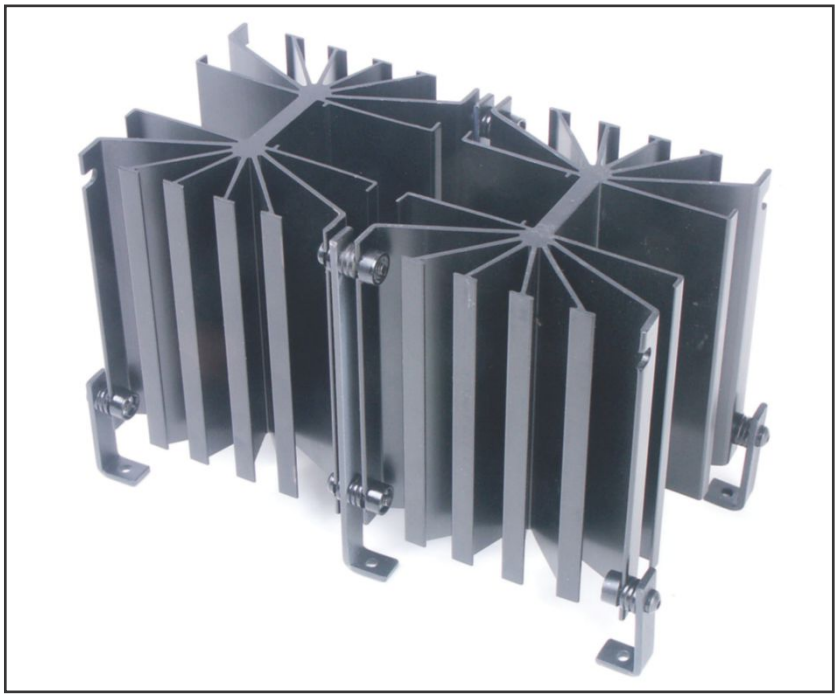


Photo 6: Here is the complete VFET heatsink assembly.

The VFET Heatsinks

Because the amplifier operates in Class A, the dissipation on each transistor is around 50 W. To maintain reasonable temperatures, substantial heatsinks are necessary. I already had one Delta model NC-441 heatsink in my parts bin. Luckily I was able to find another online. In order for the sinks to operate at maximum efficiency, they must be oriented with the fins in a vertical direction and thermal joint compound must be used on the transistor's mounting surface. **Photo 4** shows a VFET mounted on one of the sinks. It is also important for the protection MOSFET, Zener, and gate resistor to be in close proximity to the VFET. I mounted those three parts on the back side of the sink directly to the VFET leads. **Photo 5** shows that mounting arrangement.

Another requirement for proper operation is to have sufficient airflow across the fins, which means they must be raised above the chassis. To accomplish that and still fit them on the chassis, I decided to mount them side by side using three pairs of custom brackets. Since the transistors are electrically connected to the sinks and voltages above and below ground will occur with output signals the sinks must be isolated from each other and the chassis. I used insulators to fasten the sinks to the brackets. Four of the brackets connected to the outside of the conjoined sinks were short, reaching only from the chassis to the lower mounting holes. For the surfaces of the sinks facing each other, I used two longer brackets, reaching from the chassis to both the lower and upper mounting holes. This provided greater rigidity of the assembly and prevented the sinks from touching each other. **Photo 6** shows the completed sink assembly before it was mounted to the chassis.

Inside the Finished Chassis

Once all the external components were connected to the chassis, I installed the internal components including the two circuit boards, the main rectifier bridge, and a power terminal strip. Next, I completed all the wiring. **Photo 7** shows the inside of the completed amplifier, including the VU meters. Note that even though the transformer and the choke leads are coated, they were still covered with shrink tubing for additional insulation. The Parts List for the Cherry Bomb is included in the Supplementary Material found on the audioXpress website.

The Completed Amplifier

The last construction step was to attach rubber feet to the bottom cover and fasten it to the chassis.

Photo 8 shows an oblique view of the amplifier. **Photo 9** shows the back of the amplifier. **Photo 10** shows the amplifier from the top. **Photo 11** shows front of the amp after the VU meters were installed.

Testing

The next stage in this project was to get some measurements. These included frequency response, gain, power output into different loads, harmonic distortion including total, first and second, and intermodulation (IM) distortion. Power measurements were taken with a resistive load bank with both channels driven. I used an oscilloscope to view the waveform for signs of clipping. That is always an approximate method but is sufficient for this project. Frequency response, harmonic distortion, and IM distortion measurements were conducted using a PC with the application Liberty Audiosuite (LAUD). Gain was measured using a signal generator and a high-quality DVM to read input and output levels. Those readings were then converted to gain figures.

Starting with gain, the left channel had 21.0 dB and the right 20.9 dB. The gain matching we did at the input gain stage compensated for the gain differences of the VFETs very well.

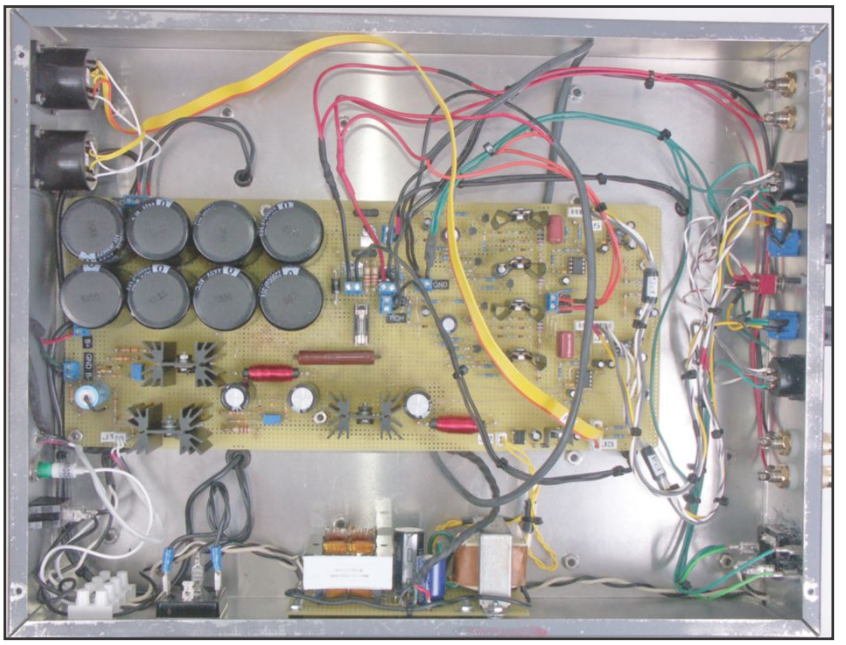


Photo 7: This is the inside view of the completed chassis.

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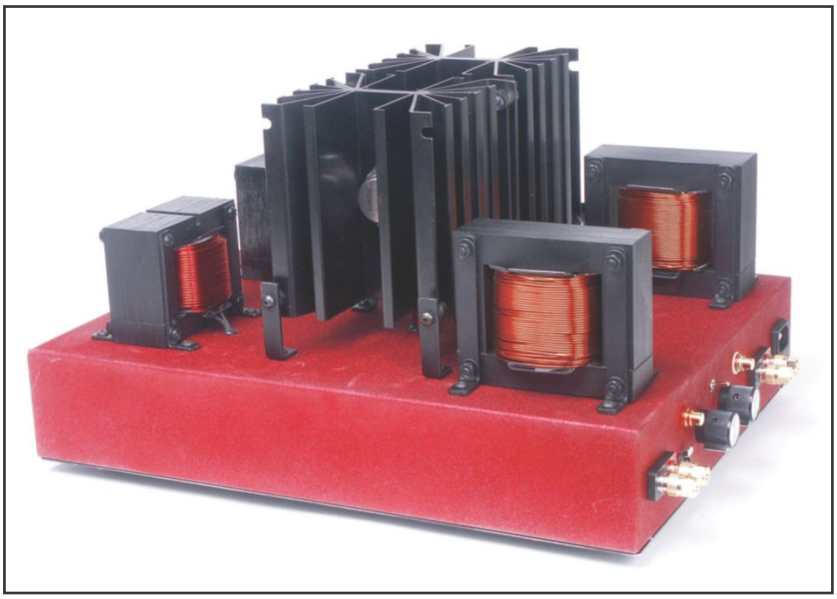


Photo 8: Here is an oblique view of completed amplifier.

Output Power

Power was measured with three loads, 8 Ω, 6 Ω, and 4 Ω. Six frequencies from 20 Hz to 20 kHz were used for the measurements. Power levels will be given as average power from 100 Hz to 20 kHz because they were as close as this measurement method would allow and another level at 20 Hz because that was slightly lower.



Photo 9: The rear view of completed amplifier shows the output connections.

- For the 8 Ω load, the left channel average was 15.3 W. The right channel was 15.2 W. At 20 Hz the left channel was 14.4 W and the right channel 14.8 W.
- For the 6 Ω load, the left channel average was 17.6 W. The right channel was 17.8 W. At 20 Hz the left channel was 13.2 W and the right channel was 13.7 W.
- For the 4 Ω load, the left channel average was 11.6 W. The right channel was 11.7 W. At 20 Hz the left channel was 9.6 W and the right channel was 9.9 W.

Our design goal was to optimize power output at 6 Ω and that appears to have been achieved. It is also interesting to note that the single-ended 300B amplifier that my friend David Berning (The David Berning Co., http://davidberning.com/techology) designed produces about 6 W into 6 Ω. The power from this amplifier is closer to the output he gets from his single-ended 845 amp. We were definitely pleased with the power levels produced.

Frequency Response

Frequency and phase response measurements were made using LAUD. Again, we were pleasantly surprised with the results. Not only was response very flat, but the phase response was also flat with some small deviations at both extremes of the frequency range which is to be expected. Detailed numbers and a graph of the response are available in the Supplementary Material found on the audioXpress website.

Harmonic Distortion

Distortion measurements were made into various load impedances and power levels. Overall, the figures were quite good with a few surprises. More information can be found in the Supplementary Material for this two-part article. However, harmonic distortion levels were as expected for this kind of design and quite respectable. In almost all cases, the distortion levels between the two channels were very similar.

Intermodulation Distortion

IM distortion was measured at a power level of 6 W into 8 Ω. With transformer output single-ended amps, the IM distortion products are usually a significant problem. With this amplifier at that power level, the IM distortion was just above 1%—a level that would be hard to achieve for most transformer coupled Class-A amplifiers. Again, we were quite satisfied. The data can be found in the Supplementary Material or by using the link for Project Files at the end of this article.

Listening

Although I am a firm believer in measurements, you must listen to really determine the final quality of an amplifier. Both of my reference speaker systems are tri-amped so they were of no value in these listening sessions. The primary system used to test the amplifier belongs to David and consists of Merlin speakers with two custom 15" subwoofers in sealed boxes that he and I built.

Since David is in the business of building amplifiers, he needs full-range speaker systems that can be driven with a single amplifier. The crossover that he designed and built to compensate for impedance anomalies at the low crossover frequencies was quite involved and very costly. The results were very good. The Merlin consists of a 6.5" Scan-Speak paper/carbon fiber cone with a low-loss rubber surround in a cast-aluminum basket. The highs are reproduced by a Dyanaudio Esotar D330A dome tweeter. Sensitivity is 89 dB/watt and impedance is 8 Ω.

We also listened to the amplifier on another system that consists of Urei 813C monitor speakers using a 15" coaxial mid/high driver. The tweeter is a horn-loaded compression driver. The low frequencies are handled by another 15" woofer. All drivers are JBL and are in a 12 ft^3 ported box. Impedance is 8 Ω and sensitivity is 101 dB/watt. They were driven full range with the Cherry Bomb but they were also supplemented by two 15" JBL subwoofers in 8 ft^3 sealed boxes through an electronic crossover below 90 Hz with a separate amplifier. **Photo 12** shows that system.

A wide range of source material was used in analog and digital formats, including open-reel tapes, LPs, and CDs. Comparison amps included the 300B and the 845 Class-A amps from David and various tube and solid-state amps on the Ureis. One thing this amplifier had in common with the tube amps was that it could play a lot louder than you would think it should, without sounding bad. With distortion that is primarily second order and relatively soft clipping, short-term excursions well beyond rated power do not



Photo 11: This front view shows the VU meters illuminated.

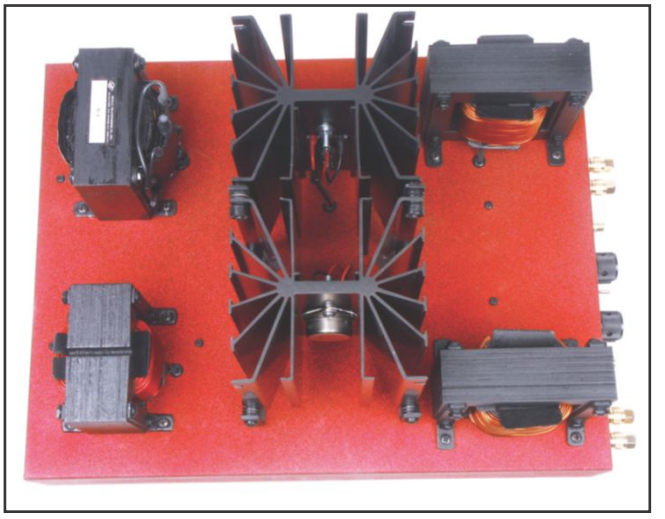


Photo 10: The top view of the completed amplifier shows all the components.





Photo 12: We connected the Cherry Bomb amplifier to a set of Urei speakers.

produce the harshness you would get from some high feedback pentode designs or poorly designed solid-state amplifiers.

The amplifier was also displayed by the DCAudioDIY group as part of its room at the Capital Audiofest 2015 show in the Washington DC area. It was played through a pair of Legacy Audio Studio HD speakers that use an 8" mid/bass speaker and a pleated ribbon tweeter. Early in the show, one of the editors of *Stereophile* magazine came into the booth and was blown away with the performance of the amp. He subsequently gave it a glowing report in his show write up.

Transient response was excellent and the amplifier was very quiet even with the high sensitivity speakers. It had no problem with high frequencies, which is what you would expect since the high-frequency response extended well beyond 20 kHz. It easily bettered some of the tube amps in low-frequency response but David's 300B and 845 amps did somewhat better in that respect thanks in part to his ZOTL design.

One of the most important things in an amplifier, for me, is to be able to properly reproduce the texture of the music. I define that as an ability to separate all the nuances of the instruments and performers in a natural way without overemphasis. It is hard to describe but as the old saying goes, when you hear it, you will know it. This amplifier provided that in spades. It is dynamic, detailed, and easy to listen to.

The Bottom Line

How do you wrap up a project for something that is not only unique but cannot be reproduced? I guess you can say that it is the perfect example of the reason why people DIY. It is not so much about the destination as the journey. Yes, the amplifier is special and should provide lots of listening pleasure. But the process of conceptualizing the Cherry Bomb, designing it, and carrying it all the way through the construction and testing phases to completion are things that expand your abilities and your doggedness to overcome obstacles and achieve a special result. You just feel good about what you have accomplished. It is especially satisfying if you have a friend like David who shares your values and is fascinated by challenges. Every time we work together we learn something from each other. It doesn't get much better than that. So I would say my final words should be go out and DIY!

Correction: The caption for Photo 1 in Part 1 of the "Cherry Bomb—A VFET SET Amplifier" article incorrectly identified the Cherry Bomb. It is in fact a VFET SET amplifier.

About the Author

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

Project Files

To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html.

Resources

D. Berning, "ZOTL Technology vs. Audio Output Transformers," The David Berning Co., http://davidberning.com/technology.

Heatsink considerations. http://sound.westhost.com/heatsinks.htm.

Nelson Pass articles, First Watt, www.firstwatt.com/articles.html.

H. Reichert, "Capital Audio Fest 2015: Day One," Stereophile, August 2015, www.stereophile.com/content/capital-audio-fest-2015-day-one#ShEYMdX18LEbQwVT.97

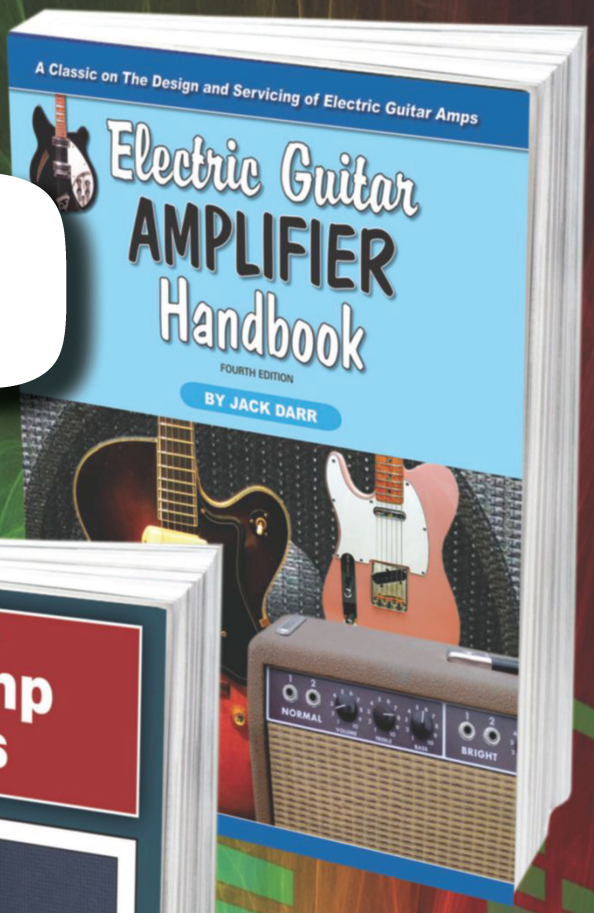
Studio HD speaker, Legacy Audio, http://legacyaudio.com/products/view/studio-hd.

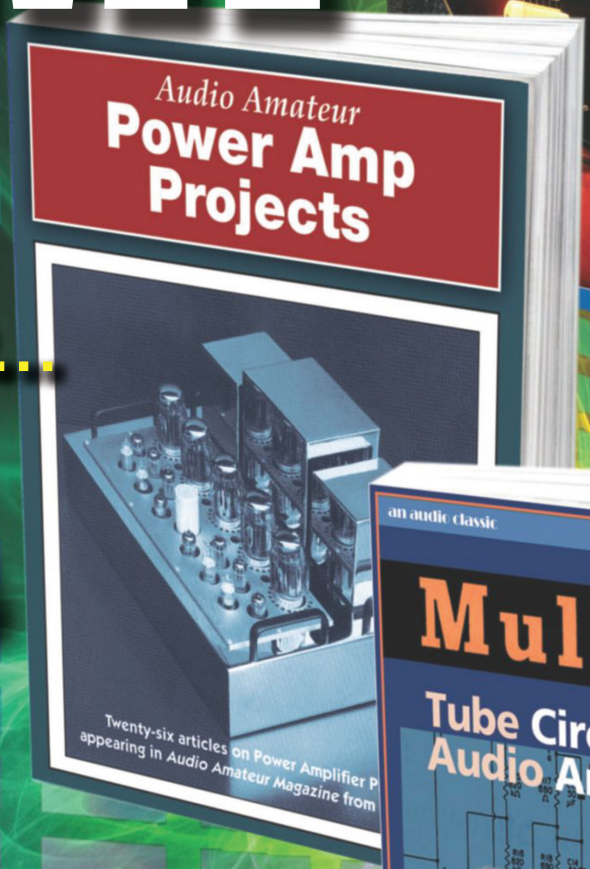
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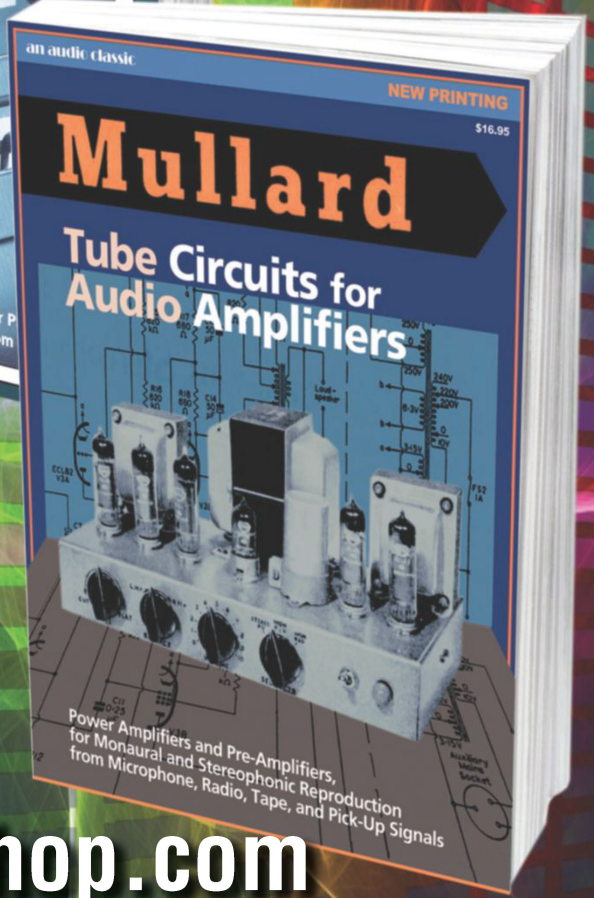
Design it...

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Electric Guitar Amplifier Handbook

Everything you need to know about how each part of the amp works and what to do when it doesn't.

Power Amp Projects

An array of designs for those who prefer to build their own audio equipment.

Mullard Tube Circuits for Audio Amplifiers

A complete guide to building 11 power and control amps for a sound system with vacuum tubes.

Resources for the audio enthusiast...cc-webshop.com

Routine Maintenance for a Hard-Working Bass Amplifier

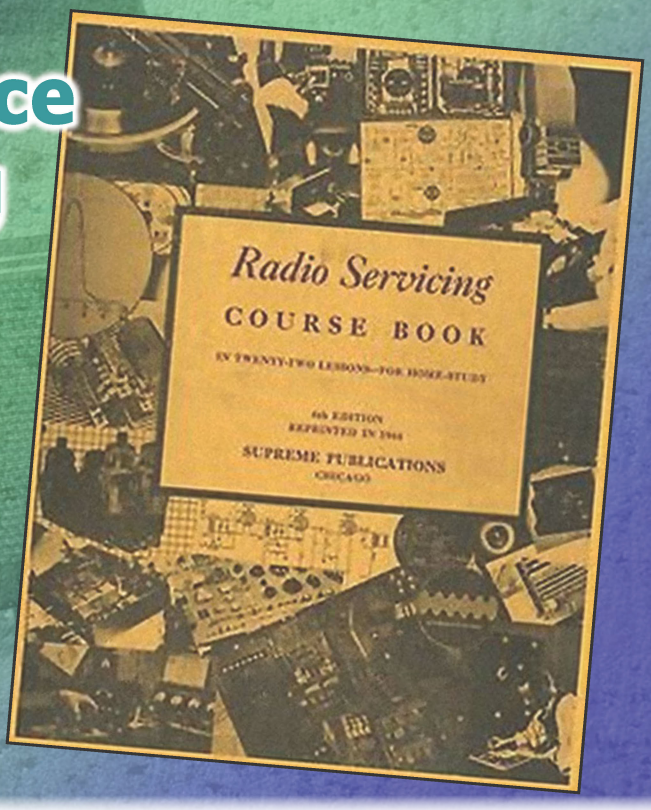


Photo 1: This early book published by Supreme Publications offered unique and useful instructions in repairing hollow-state equipment.

It is possible for professionals and hobbyists to keep classic hollow-state audio electronics in good repair thanks to several companies around the world who now make exact replacement parts.

By
Richard Honeycutt
(United States)

In the old *Supreme Radio Servicing Course Book* (see **Photo 1**), author Morris N. Beitman made the somewhat humorous statement that if a radio serviceman wanted to impress the customer with his knowledge, he could look at a defective radio and predict, "It's probably a bad capacitor." It was a safe assumption at the time that the customer had already checked all the tubes at a local drug store, and although those drug store tube testers were only marginally useful, if they said the tubes were okay, then odds were that a defective capacitor was at fault. While I can't verify the accuracy of Beitman's advice, I can state that capacitors—particularly electrolytics—were frequent causes of electronic equipment failure in the early days.

Electrolytic Capacitors

The earliest electrolytic capacitors were the "wet" type, using a liquid electrolyte to make contact to one accordion-shaped aluminum plate. When voltage was applied to the capacitor, one plate would form a very thin layer of aluminum oxide that served as the dielectric. The outer aluminum canister served as the other plate. Usually, there was a small rubber stopper in the can through which more electrolyte could be added if the capacitor dried out. Early books on electronic servicing gave the concentration of salt water to use to replenish the electrolyte.

The dry electrolytic capacitor was patented by Samuel Ruben in 1925, and used a non-aqueous gel electrolyte rather than a water solution. As late as 1939, some equipment manufacturers were still using wet electrolytics, presumably because of their lower cost.

Although as early as 1956, Western Electric produced electronic equipment such as amplifiers for underwater cable telephone service, having a design lifetime of 40 years, standard commercial electronics did not meet that standard of reliability. As late as 1980, standard wisdom assumed that the expected life of a dry electrolytic in daily use was about 10 years. In the years since, great improvements have been made in the design and manufacture of electrolytic capacitors.

Maintenance

Professional users of hollow-state electronic equipment have sometimes chosen to perform periodic maintenance to avoid untimely failures. For pro users of musical electronic devices, the periodic

maintenance primarily included replacement of output tubes and electrolytic filter capacitors. The output tubes aged more quickly than the preamp tubes because of their higher operating temperatures. Besides, preamp tubes are often self-diagnosing when they fail, since the most common failure mode is that they become microphonic. Stray thumps in the output of a guitar or bass amp or squeal from internally generated feedback can indicate a microphonic tube, which can be checked by thumping the tube while the amp is powered up and the volume controls are turned up.

It might sound strange to say that dry electrolytics fail by drying out, but that's exactly what happens if the electrolyte seeps out at a pressure-release vent. When the capacitors dry out, the capacitance decreases and the equivalent series resistance increases. This happens more often with filter capacitors than with cathode-bypass capacitors, which constitute the other main use for electrolytics in a tube amplifier. The result can be increased power-supply hum. Another failure mechanism is internal shorting, which produces catastrophic failure, along with blown fuses.

In much older hollow-state equipment, manufacturers saved money by using dual, triple, or even quadruple electrolytic capacitors in the same case, with a common negative terminal. These were available in both tubular and "twist-prong" can packages. The latter were so named because the can had three or four flat prongs that were inserted into slots punched in the chassis and then twisted in order to secure the capacitor (see **Photo 2**).



Photo 2: This new-old-stock (NOS) triple electrolytic capacitor has a "twist-prong" case.

Finding Replacement Parts

Procuring replacement parts for tube amplifiers has presented a variety of challenges over the years. In the 1940s, musical instrument amps were usually small and used pretty much the same parts as did radio sets, making exact replacement parts readily available. Beginning in the late 1940s and extending through the 1960s, TVs were the big fish in the "parts pond," and they usually used different values of filter capacitors than high-powered instrument amps: 40 µF/450 V units were about the highest capacitance and voltage units available. Musical instrument dealers usually could supply the common output tubes: 6L6s, 6V6s; and later, EL34s, EL84s, 6550s, and KT88s.

However, when Ampeg unveiled the SVT (see **Photo 3**) at the 1969 National Association of Music Merchants (NAMM) show, it singlehandedly ushered in a new era of procurement challenges. To begin, the 300 W output was provided by a sextet of 6146 power tubes—definitely not found at your local drug store or even your garden-variety music store.

Although Ampeg changed to 6550 output tubes in mid-1970, the 660 V high B+ (A) and even the 462 V preamp B+ (see **Figure 1**) required filter capacitors that were hard to find. Of all the filter capacitors—C8, C9, C10, C11, C12—and the 100 µF/450 V decoupling/filter capacitor in the preamp, only C8 was commonly stocked by electronic parts suppliers in the 1970s and 1980s. Of course, factory parts were available (at a premium price) through musical instrument dealers.

Replacing high-capacitance, high-voltage capacitors often required using lower-capacitance values wired in parallel. So, replacing a 100 µF/40 µF, 450 V dual capacitor (e.g., C9 or C12 in the SVT) would



Photo 3: The "Super Valve Technology" (SVT) amplifier designed by Bill Hughes and manufactured by Ampeg is one of the truly fine classic bass amps.

require three tubular capacitors: two 50 µF/450 V units in parallel, and a separate 40 µF/450 V unit. Replacing capacitors having higher working voltages than the common 450 V maximum could require wiring lower-voltage units in series.

Replacing C11, which was 30 µF/600 V, required two 60 µF/300 V (or higher voltage) units in series. There is always some risk in using series-connected capacitors to achieve a higher voltage rating: if the capacitances and/or leakage resistances are not very nearly equal, the DC voltage will not divide equally across the pair, and one capacitor might seem too high a voltage and consequently short out. There are two solutions to this potential problem: Parallel each capacitor with a 1 MΩ resistor so that the resistor pair acts as a voltage divider and makes the capacitors split the voltage equally, or use capacitors with higher working voltages than half the desired result. If we used two 60 µF/450 V capacitors in series to replace C11, it is likely that the two capacitors would be closely enough matched that neither would be exposed to more than its rated voltage.

A friend recently brought me his 1970s-vintage SVT for replacement of the output tubes and filter capacitors, giving me an opportunity to look at the present availability of replacement parts.

Photo 4 shows the tightly packed chassis that resulted from another technician's previous work—replacing the filter capacitors using series and parallel units as necessary. Before taking the photo, I stood all the capacitors up on end in order

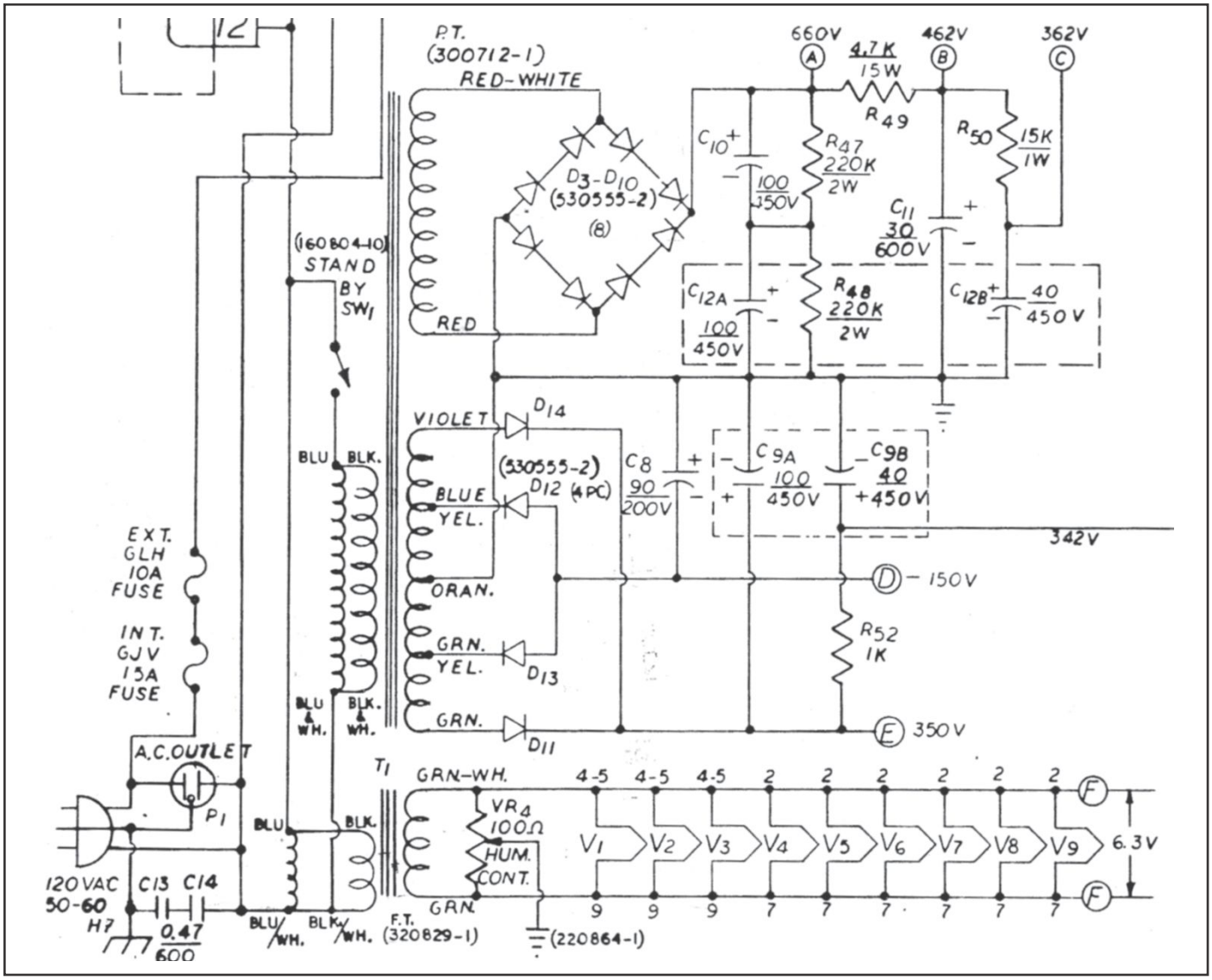


Figure 1: The SVT power supply schematic reveals several components that have been challenging to procure.

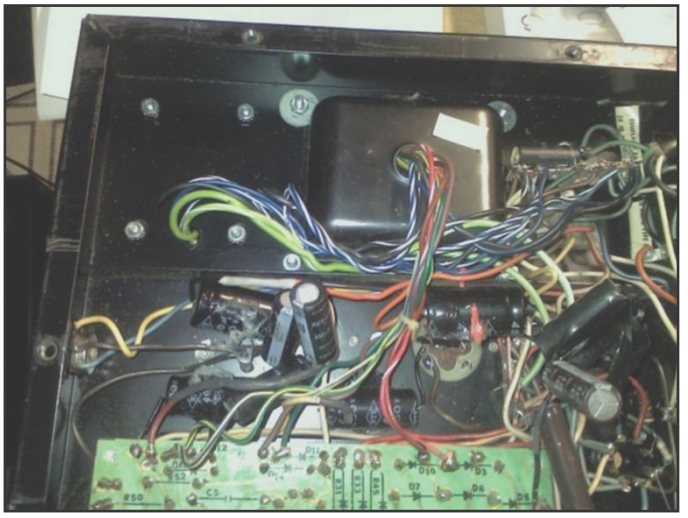


Photo 4: Replacement work using multiple series and parallel pairs can create a mess!

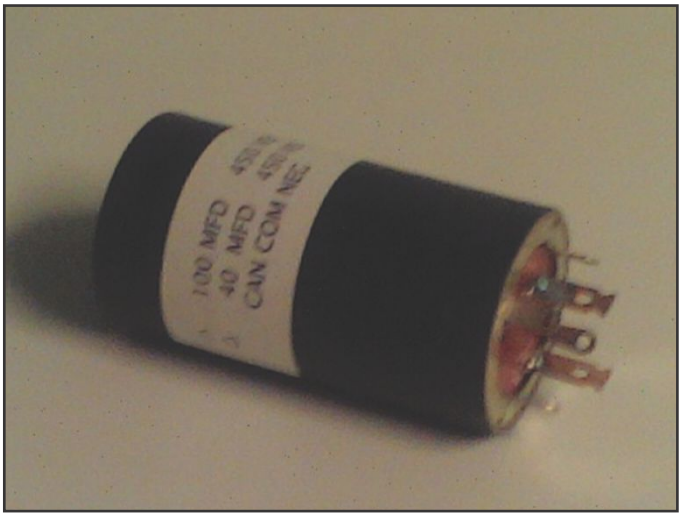


Photo 5: This new twist-prong dual capacitor is an exact replacement for C9 and C12.

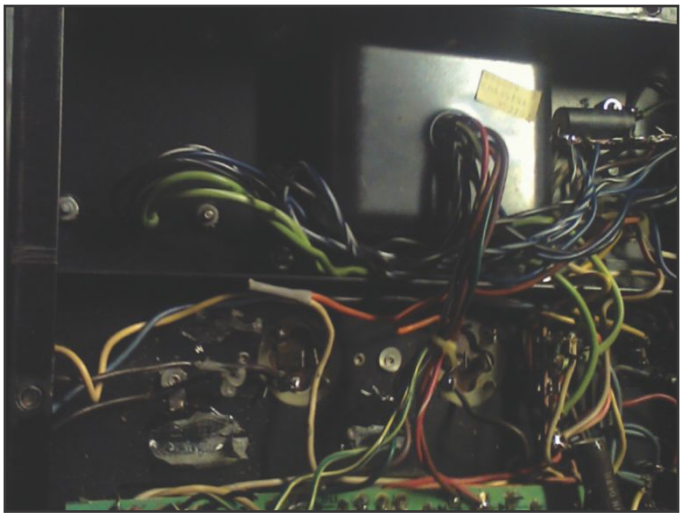


Photo 6: After restoration to the original parts layout, the chassis has more room for air to circulate, keeping components cooler.



Photo 7: The re-tubed and re-capped SVT is now back at home in its case (with the back plate not yet installed).

to be able to read the values. When lying down in their normal positions, they took up pretty much all the available space in that part of the chassis.

The "parts pond" has changed since the 1980s. These days, TVs are usually replaced rather than repaired, meaning audio and musical electronics are now the major inhabitants of a much smaller pond. The result for hollow-state enthusiasts is that small manufacturing companies can now produce those formerly hard-to-get parts and make enough money to stay in business. Thus, I was able to procure a matched sextet of Electro-Harmonix 6550s and exact replacements for all the capacitors with the exception of C8 (90 µF/200 V), for which I used a 220 µF/300 V replacement.

Photo 5 shows the new 100 µF/40 µF, 450 V dual capacitor of which I was able to get two, for replacing C9 and C12. Admittedly, special parts such as this one are not inexpensive; but when you're working on a classic, they're worth the cost.

Photo 6 shows the completed repair, matching the original design using a total of five capacitors (including the two twist-prong duals) to replace nine used in the previous job. Another 100 µF/450 V unit was used in the preamp, which is on a separate chassis. In **Photo 7**, the re-tubed and re-capped amp is shown in its case, with the perforated steel back plate not yet installed.

The only performance advantage obtained by restoring this amplifier to its original parts complement is better air circulation under the chassis, potentially resulting in longer life. Yet it is satisfying to avoid the "shade-tree mechanic" feel we had to endure during the years in which the only way to get exact replacements was to go to a musical instrument dealer, placing special orders and paying high prices. Although Ampeg is still in existence (owned by LOUD Technologies of Woodinville, WA), it no longer stocks parts for this particular model of SVT, but only for later models that don't use dual twist-prong capacitors.

Professionals or hobbyists doing an Internet search for exact replacement parts for classic hollow-state audio electronics will be rewarded with the websites of several excellent companies who specialize in making our lives easier!

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.

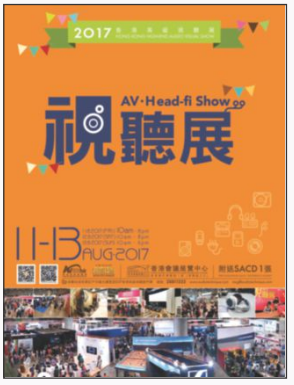
July 13–15, 2017
Summer NAMM
Music City Center, 700 Korean Veterans Blvd., Nashville, TN
www.namm.org/summer/2017

Summer NAMM is traditionally the "reinforcement" trade show for the music instruments industry in the US following its earlier exhibits at the vibrant Winter NAMM. Since settling at the Nashville Music City Center, Summer NAMM has shown positive signs of growth, continues to be one of the most vibrant shows for the music instruments industry, and is finally showing that it is an important event for the audio industry, with a strong emphasis on recording technology. The annual summer music industry gathering is always a business-friendly environment mixed with great entertainment and networking opportunities, also benefiting from an amplified presence of pro audio brands. As the National Association of Music Merchants (NAMM) reinforces, the Summer NAMM event provides a dynamic platform for emerging and established brands to create lasting impressions as buyers plan for the second half of the year. And NAMM has been creating the right initiatives to create growth dynamics and differentiation for the Nashville event, like the highly-anticipated Top 100 Dealer Awards, NAMM U sessions, TEC Tracks, Advanced Audio + Applications Exchange (A3E) educational sessions, and a strong connection with the house of worship sector. Before the show officially opens, on Wednesday, July 12, members can enjoy an all-day intensive training at Retail Boot Camp. The growing pro audio focus has also helped the event attract more purchase decision-makers representing a variety of major global companies and venues.

July 28–30, 2017
California Audio Show
Hilton Oakland International Airport, 1 Hegenberger Rd., Oakland, CA
www.caaudioshow.com

The good American tradition of promoting audio shows in all the main cities and attract both the manufacturers, retailers and potential buyers to regular meetings is keeping the high-end and consumer audio sectors afloat. For the last two years, those shows have been clearly on an upswing, opening up good perspectives for other less consolidated events, such as The California Audio Show (CAS). The second longest-running audio show in the US, now in its 7th year in the beautiful San Francisco Bay Area, CAS takes place at the Hilton Oakland International Airport. The space offers plenty of rooms of the presentations, accommodation for fair prices and plenty of parking. The event will feature the most recent consumer audio offerings presented by more than 120 brands in 50 plus rooms, and includes an entire section dedicated to headphones and related products, and a CD, vinyl, and HRA sale shopping area.

August 11–13, 2017
2017 Hong Kong High End Audio Visual Show
Hong Kong Convention and Exhibition Centre in Wan Chai, China
www.audiotechnique.com/avshow/11700/



This three-day event, sponsored by Far Eastern Music & Audio Enterprises, Ltd., provides a good opportunity in Asia to showcase all types of audio products in a vibrant market, find new suppliers, and do business, with plenty of displays in personal audio, headphones, speakers, cables, accessories, and all audio components.

August 21–24, 2017
SET EXPO 2017
Trade Show: August 22–24
Conference: August 21–24
International Technology Fair on Equipment and Services for Television Engineering, Broadcasting, and Telecommunications
Expo Center Norte Red Pavilion and Convention Center, São Paulo, Brazil
www.setexpo.com.br

The Brazilian Society of Television Engineering (SET) is one of the most prestigious associations in Brazil and Latin America with direct connections to the National Association of Broadcasters (NAB) and the International Broadcasting Convention (IBC) industry associations and respective events counterparts. Even though this is a broadcast-oriented event, SET Expo is undoubtedly the largest and most dynamic business opportunity for the audio and video industries in the Latin American region. SET is also increasing its cooperation with the local section of the Audio Engineering Society (AES), given the strong showing of audio companies at the show. With a strong focus on radio—which is still a vibrant segment in the region—the SET Expo benefits from a strong direct presence of manufacturers and country delegations, with more than 21,000 m² for exhibits and over 400 brands exhibiting.

August 30–September 2, 2017
3rd AES International Conference
On Sound Reinforcement—Open Air Venues
Struer, Denmark
www.aes.org/conferences/2017/sr

This conference on sound reinforcement is a follow-up to the 2015 conference in Montreal and will focus on sound reinforcement in open-air venues. It will bring together recognized hands-on experienced practitioners, scientific experts, self-taught gurus, young technicians, and students in a creative environment. There will be workshops, demonstrations, and live experiments combined with paper presentations, engineering briefs, and tutorials. Live performances will demonstrate the most advanced and up-to-date sound systems operated by renowned sound engineers on an outdoor stage. Technical papers and tutorials will be presented by recognized sound system designers and acoustics specialists. A one-track scheduling format is planned. However, it is expected that parallel tracks for various hands-on sessions and demonstrations as well as theoretical sessions can be organized to suit individual preferences.

September 1–6, 2017
IFA Berlin (The International Funkausstellung)
Berlin ExpoCenter City, Messedamm 22, Berlin, Germany
www.ifa-berlin.com

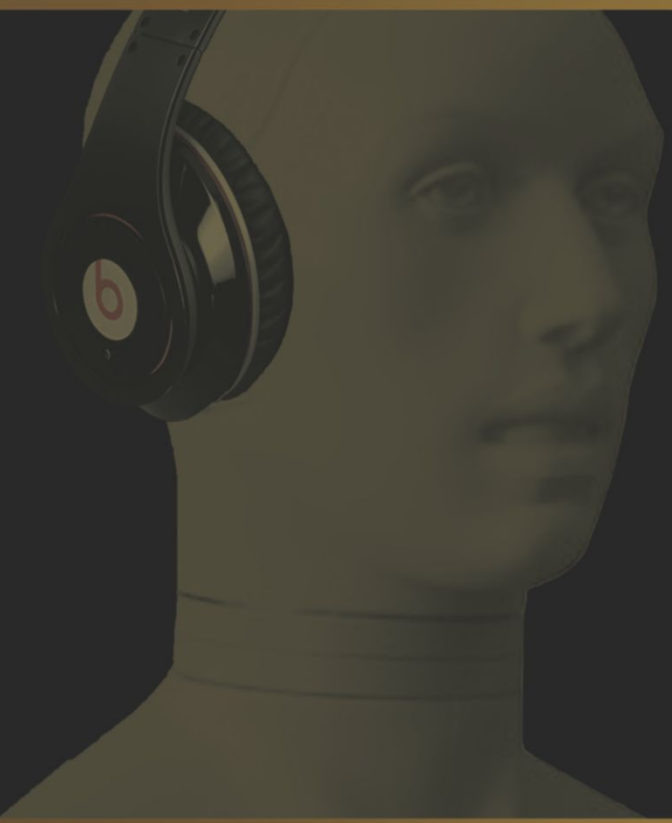
IFA Berlin (The International Funkausstellung) is probably the most important annual event for the consumer electronics industry, introducing new products in time for the most important season of the year for sales. If the CES in Las Vegas, N.V., is the place were future trends and innovation are highlighted and get the most attention, the IFA Berlin show is where many of those products and brands make it or break it. Audio at IFA is mainly focused on personal/mobile devices and the more accessible and lifestyle-oriented products. A wide range of products, solutions, and presentations make a visit to IFA 2017 worthwhile. IFA 2016 featured more than 1,800 exhibitors and attracted 240,000 visitors, with the majority of trade visitors coming from outside Germany: 68,000. The 2016 floor space was already the largest in the show's history. Expect the 2017 edition to be even larger!

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