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Fresh From the Bench

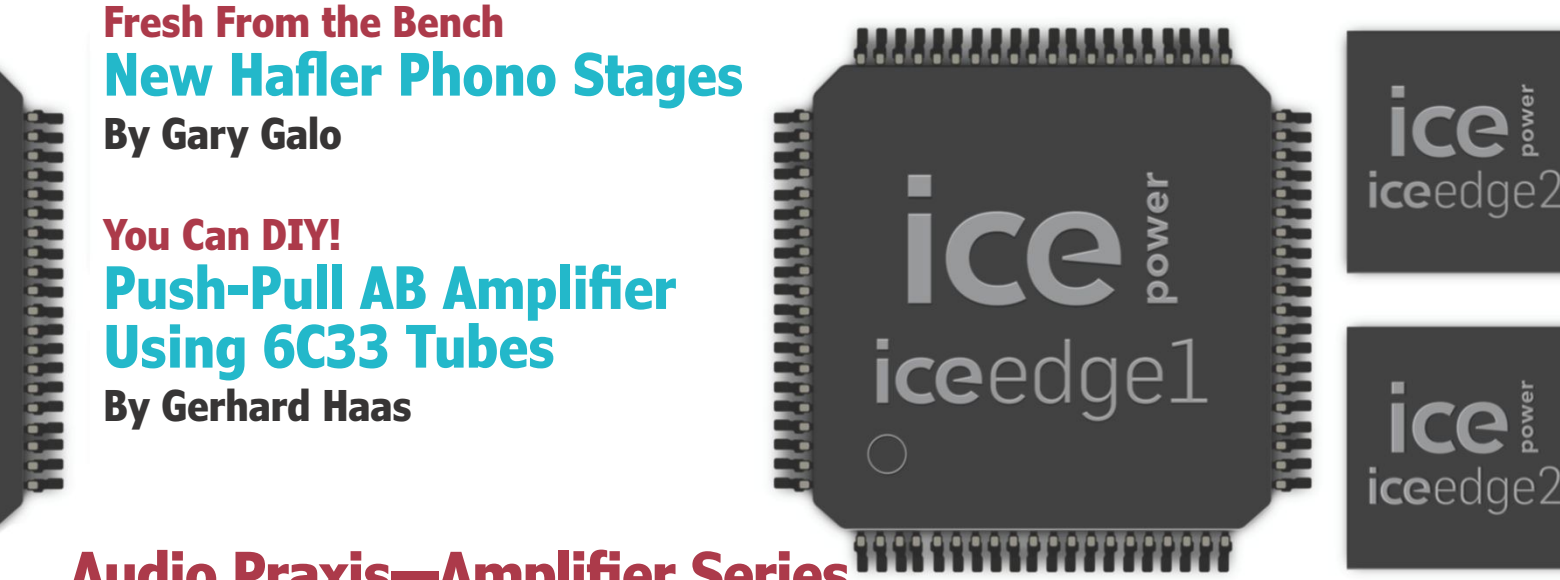
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For the Benefit of Our Readers

While working on the content for this April edition of *audioXpress*, I couldn't help but notice how restrictive the number of pages in a magazine becomes when compared with the limitless possibilities we have on the audioXpress.com website or even our increasingly popular weekly email newsletter, *The Audio Voice*.

The frustration with those restrictions is that so many up-to-date audio technology topics that could be of great interest to our readers take months to convert into the solid content we include in the pages of the magazine. Meanwhile, news and innovations continue to flood our email, forcing us to mainly work online and issue brief updates.

Therefore, I thought I should use this space to reinforce the appeal for article submissions that could fill that gap. We see so many professional papers that were introduced to a few dozen people who attend convention and seminar sessions that could be of great interest to this publication's audience once they are properly converted to a magazine article format. Increasingly, I also see engineers sharing "drafts" of what could be material of great interest to our readers, posted as updates on social media that only reach a restricted group of "followers." Often, that content contains valuable information that could be expanded into an extremely motivating article. To those authors, please reach out to *audioXpress*. We can help share your efforts with the audio community and open doors to valuable interactions and additional business.

We have interacted with several companies and engineers at recent trade shows and we have seen many interesting developments that we would like to share. Unfortunately, the little information we could collect is not sufficient to create a news report, much less a serious article for the magazine. At certain stages of development, we understand only those in charge can provide the level of information our publication requires. Again, we would like to reiterate that we have multiple channels where we could—working together—make your work known. *audioXpress* is the right channel to discuss technology at an early stage with an interested audience, particularly for development platform providers.

In the last three years, we have invested heavily to reach out and personally meet industry leaders, and we are developing a new article series to reflect much of what we are learning. But as with any technical magazine, we need direct collaboration—even sometimes for things that are quite traditional, such as writing a product review. While some companies are used to submitting their products for media to review, technology and parts manufacturers are less accustomed to the process (or not at all). It took many years, and many of Vance Dickason's industry contacts, for our sister publication *Voice Coil* to establish the—now highly recognized—Test Bench procedure characterizing speaker drivers. We are trying to do the same with *audioXpress* in all the many other disciplines of audio.

audioXpress doesn't intend to evaluate products in a subjective way. We understand that writing for developers and even DIY enthusiasts requires a different approach and we publish reviews that explain why our readers need what you have. Sometimes we need to do that before the product is even generally available. We think we can have an important role in sharing knowledge about wireless audio, new amplifier topologies, new transducer materials, new acoustic treatment materials, and new voice processing technologies, for our audience to actually use them in their projects.

Our recent Test and Measurement Focus issue was a good example of that effort. This month we have started a new series on amplification and we are expanding our process of finished product reviews with objective analysis, from a simple product design validation in any class of product, to key trending technology areas. The entire *audioXpress* team continues to work to fulfill that vision.

João Martins
Editor-in-Chief

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COLUMNISTS

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

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.

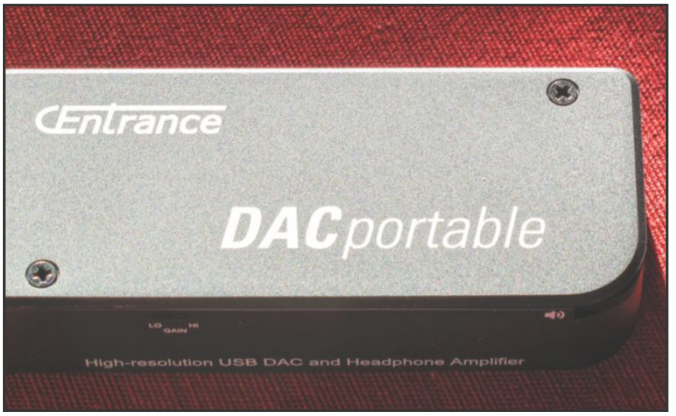
Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957, he started Testronic Development Laboratory, which became TDL Technology, to develop audio electronics. All product sales and services were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

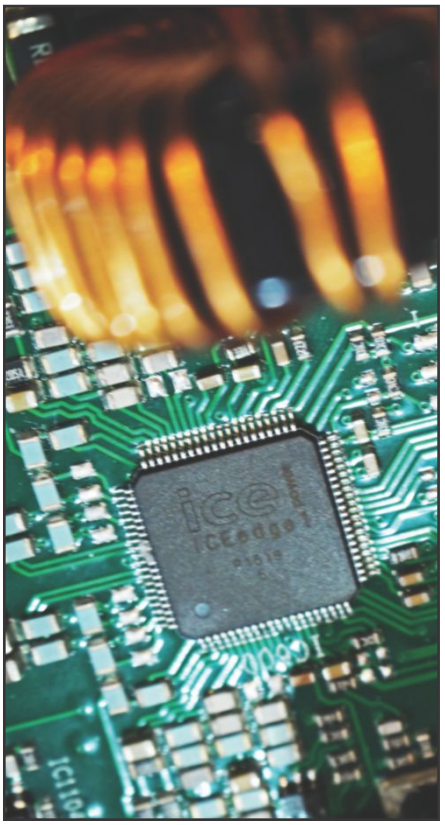


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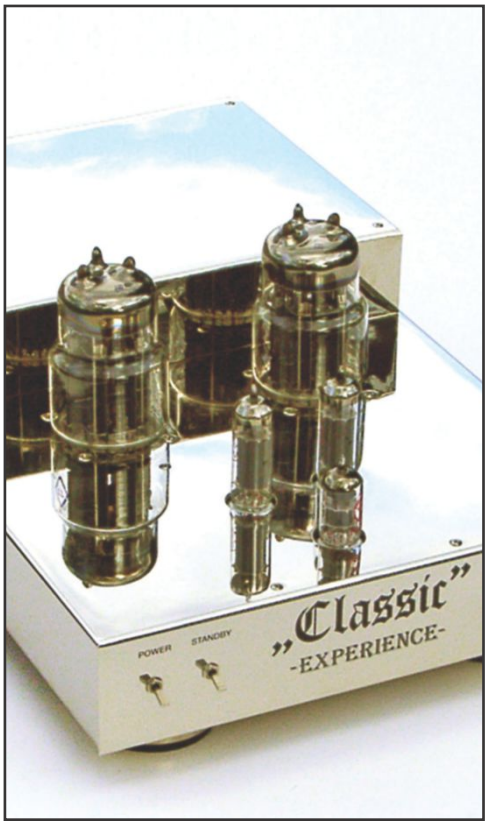
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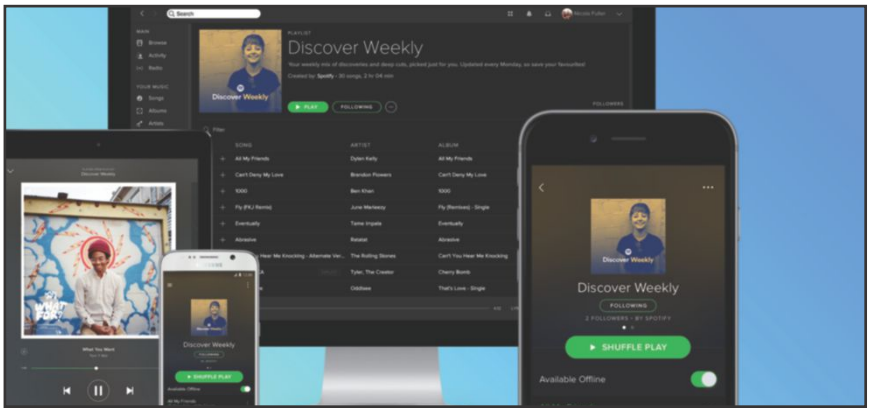
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Streaming Music Experiences (Part 4)

Streaming from Spotify

By
Ron Tipton
(United States)



In this article series, we continue to evaluate the quality of streaming music services. Spotify, founded in Stockholm, Sweden, is currently the world's leading company in this field, with more than 40 million paid subscribers and over 100 million active users, providing access to more than 30 million songs. The quality must be good, right?

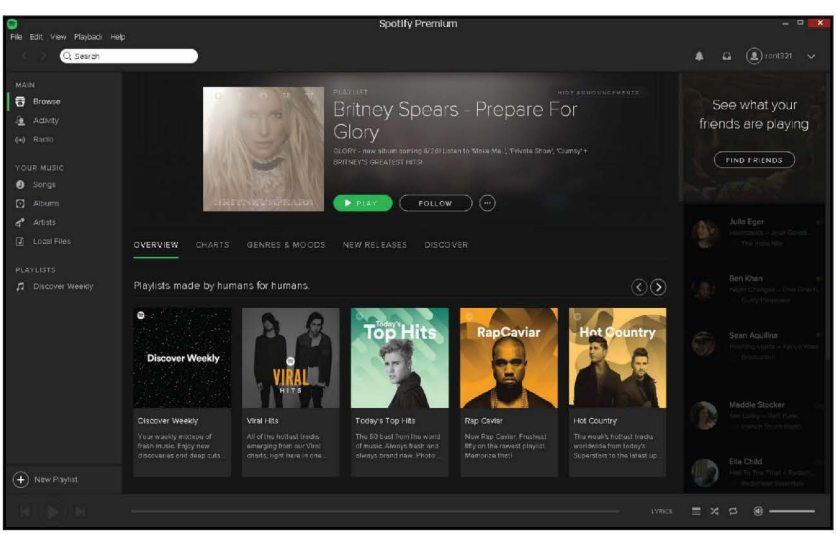


Photo 1: This is the opening screen you see if you have an account. It changes frequently to highlight new albums and playlists. I have a Premium account so Spotify Premium is displayed at top center. The main menu bar is at horizontal center with other options listed vertically on the left side. From here you can explore everything Spotify has to offer.

Spotify is a popular service even though it streams Ogg Vorbis (see Sidebar) at 320 kbps, which is much lower than CD quality music. It does offer a free service, but it is ad supported with commercials and you are limited to 160 kbps (96 kbps on a mobile device), and just six skips per hour. The "Premium" service is $9.99 per month with a free 30-day trial before billing starts. It's ad free, with unlimited skips and offline listening. I signed up for a Premium account, so let's take a look at how to get started and then discuss its performance.

If you are new to Spotify, you can click on "Learn About Spotify" or set up a Free or Premium account. When setting up an account, you will need to fill out an online form that asks all the usual stuff for a new account, with credit card information required if you choose Premium.

If you have an account, in this case Premium, you will see the initial screen shown in **Photo 1**.

Note that it says Spotify Premium at top center and my user name at top right. This screen changes, at least daily, to highlight an album or with useful information. There is a list of options on the left and the main menu bar is across the screen center. Clicking Help loads its menu, where you can view and edit your account data or choose one of the other options.

I often start by choosing Genres & Moods on the main menu bar. This loads a large group of icons in apparently random order. Included are Folk & Americana, Jazz, Classical, Blues, Reggae and many others.

Another option is entering an item in the search box, upper left corner. For example, I entered Dave Brubeck, which displays a partial list of his albums. Clicking "See All" in the upper right corner displays all the cover art—there are 122 in the Spotify collection. Clicking any album cover displays its track list.

Streaming Distortion

As with TIDAL, I asked the question: Does the streaming channel introduce distortion? And again, I recorded the same track streamed at different times and compared them. To be sure of a truly valid answer, this would have to be continuously done. But recording samples on several consecutive days provided a practical answer in TIDAL so it seemed likely that it would suffice once again.

Spotify streams Ogg files and the NCH SoundTap software used for TIDAL records only, .wav, and MP3. During a web search, I found the Streaming Audio Recorder from Apowersoft. It includes Ogg in its longer list of recording formats. Although not free, it is reasonably priced at $39.99. (Mac Audio Recorder does the same job in OSX.) As best I can determine, recording the stream is legal in the US provided it is solely for your personal use.

Set up the file format and save the folder name by clicking the gear icon in the upper right corner (see **Photo 2**). Then "arm" the recording mode by clicking the red dot in the menu bar, lower left corner. Recording does not start until it detects an audio stream. This starts a text counter in the menu bar showing both the elapsed recording time and the file size. During recording, the red dot changes to a red square. Stop recording by clicking it. The saved file names are displayed in a row of icons above the menu bar. As an added touch while recording, "notes" are shown flowing into the microphone.

Comparing the recorded samples is easily done with Audio DiffMaker, a gem of a free program from Liberty Instruments. **Photo 3** shows a screenshot

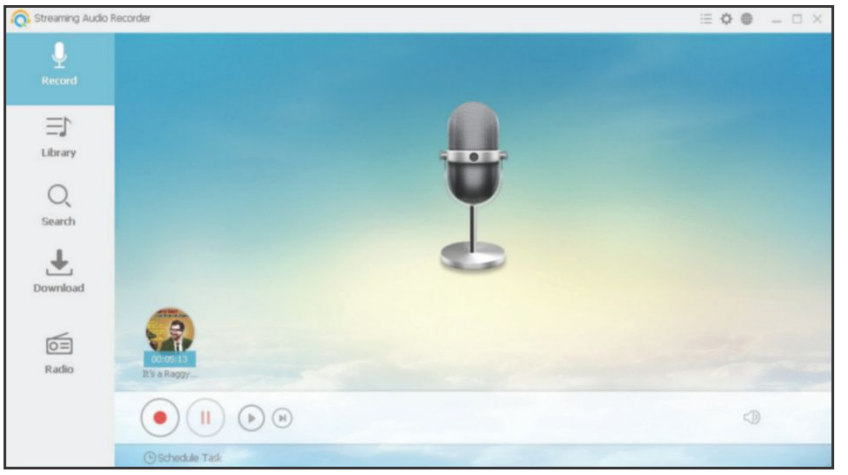


Photo 2: Clicking the red dot in the lower left corner of the screen shown here starts the Streaming Audio Recorder when an audio signal is detected. Musical "notes" come streaming into the microphone from all directions. But first, set up your preferences by clicking the gear icon, upper right. This screenshot shows a recorded file icon, left side just above the menu bar.

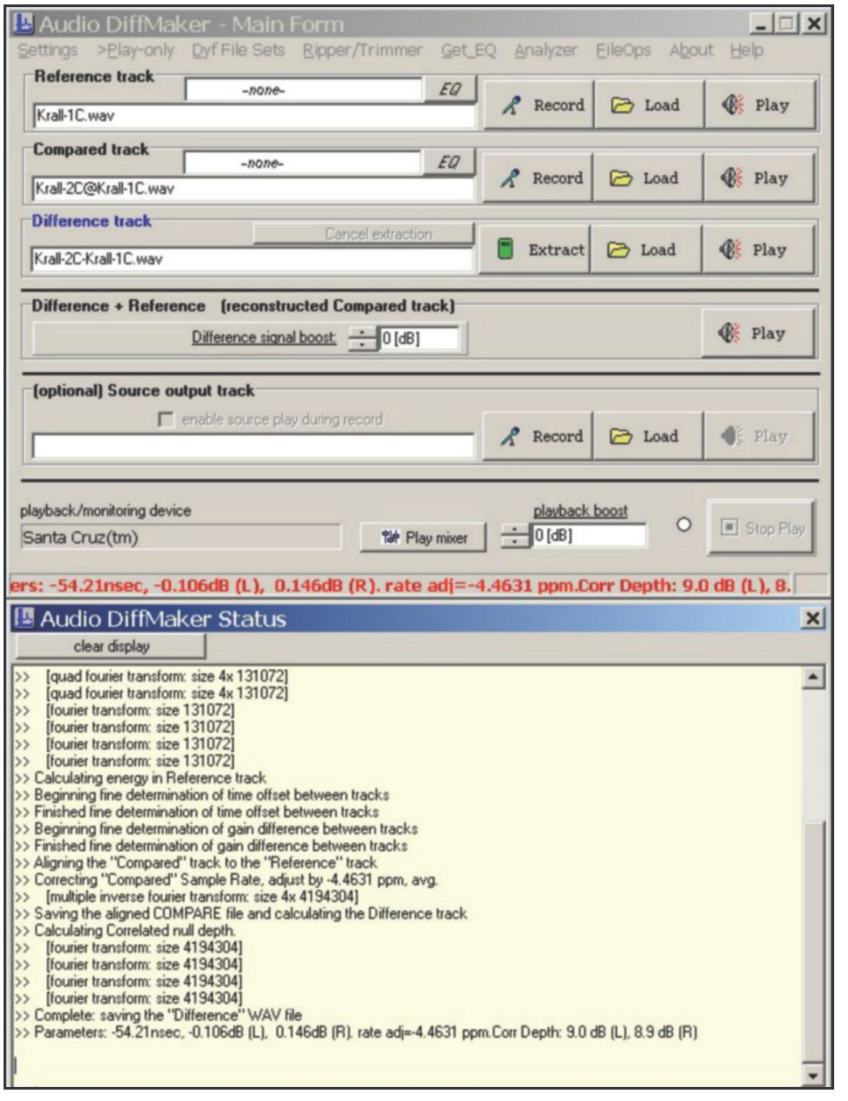


Photo 3: DiffMaker constantly updates the lower text box with the current operation. If the two files greatly vary, the program ends with an error message. Successful completion displays the average sample rate error, the time added to align the files, and the left and right null depths. The peak-to-peak sample rate error would be useful but, unfortunately this program is no longer being updated.

Media	Stream Rate	Sample Rate Error	Left Null Depth	Right Null Depth
Paul Desmond, "Autumn Leaves," *Cool Imagination*, Marathon Media International	415 kbps	9.87 ppm average	7.8 dB	8.3 dB
Diana Krall, "Dancing In The Dark," *Look of Love*, Verve	464 kbps	1.53 ppm average	6.2 dB	6.9 dB
Willie Nelson, "Always On My Mind," *Always On My Mind*, Sony BMG Music	403 kbps	1.99 ppm average	12.5 dB	11.2 dB
Dave Brubeck, "It's A Raggy Waltz," *Dave Brubeck's Greatest Hits*, Sony Music	401 kbps	4.45 ppm average	4.8 dB	6.4 dB

Table 1: This table displays the sample rate error and the left and right null or correlation depths for four Spotify tracks from different albums. Media information is also included. The Sample Rate Error and Null Depths were produced by the DiffMaker software using the same tracks recorded on different days. The Stream Rates were measured from the recorded Ogg files by both Audio Identifier and MediaInfo.

of a completed comparison. You start by using the setup screen to select your sound card and other parameters. Then you enter the two file names for comparison using the LOAD buttons. Then click EXTRACT. The program time and amplitude aligns to two files and then subtracts one from the other, leaving the residual noise plus any file differences. The average sample rate error is also calculated and displayed in the text box at the screen bottom.

Table 1 summarizes my results of comparing four music tracks from different albums. The fairly small numbers in the Null Depth columns indicates

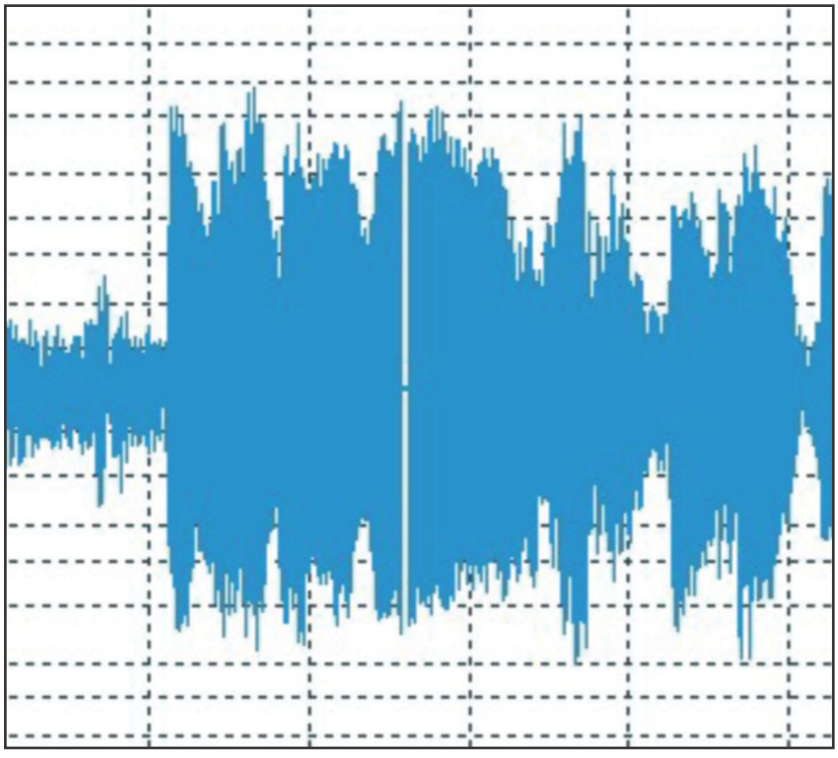


Photo 4: This screenshot of a portion of Paul Desmond's "Autumn Leaves," take one, left track, shows an apparent dropout that is heard as a "glitch" in the playback. The take two and take three tracks did not have a glitch at this time position.

Project Files

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

Resource

R. Tipton, "Signal Dynamics and Loudness," *audioXpress*, May 2015.

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Apowersoft, Ltd. | www.apowersoft.com

TIDAL
Aspiro AB | tidal.com

Audio Identifier 0.7.1
Download.com free software | http://download.cnet.com/s/audio-identifier-0.7.1

Audio DiffMaker
Liberty Instruments, Inc. | www.libinst.com

Mediainfo 0.7.87
MediaInfo free software | mediaarea.net/en (Windows & Mac)

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NCH Software | www.nch.com.au/soundtap (Windows & Mac)

Spotify
Spotify AB | www.spotify.com

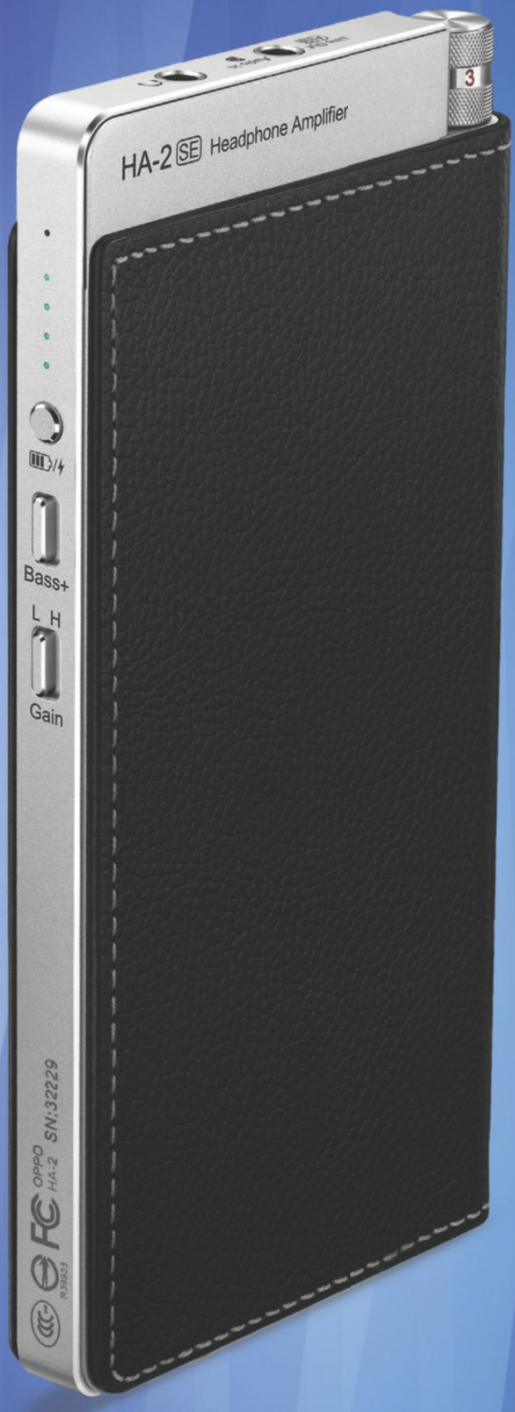
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	MusicScope			MLoudness Analyzer	
WAV File	"M"	"S"	"I"	"I"	Range
Track 1	-15.8 LU	-17.7 LU	-21 LU	-21 LU	6.4 LU
Track 2	-13.3 LU	-18.8 LU	-23.4 LU	-23.4 LU	9.5 LU
Track 3	-14.2 LU	-18.9 LU	-22.3 LU	-22.3 LU	5.9 LU
Track 4	-12.2 LU	16.5 LU	-22 LU	-22 LU	17.5 LU
Track 5	-12.9 LU	-18 LU	-21.5 LU	-21.5 LU	8.6 LU
Track 6	-16 LU	-19.2 LU	-23.1 LU	-23.1 LU	7 LU

Table 2: The loudness data provided in this table is from the first six tracks of a Spotify playlist "Sleep." The momentary "M," 3-second average "S," and integrated values "I" are from MusicScope. The MLoudness Analyzer found the same "I" values and range is "I" minus "M." I found "S" to be a useful loudness normalization parameter and adjusted the "S" values of tracks 1, 4, and 5 to -19 LU. A Loudness Unit (LU) is approximately 1 dB.

a large difference between the compared samples. In fact, the entire files (about 3 to 5 minutes in length) were so different they would not compare. I had to use just the first 60 seconds of each to get a comparison without an error message. The sample rate errors are also fairly large and this "bit rate jitter" may account for the significant difference between the files recorded at different times.

In a listening test, I noticed "glitches," which appear to be dropouts (see **Photo 4**), in the music at differing times in the supposedly identical files. Overall, this does not greatly affect the pleasure of listening to the music but it does make it apparent that it's not CD quality.

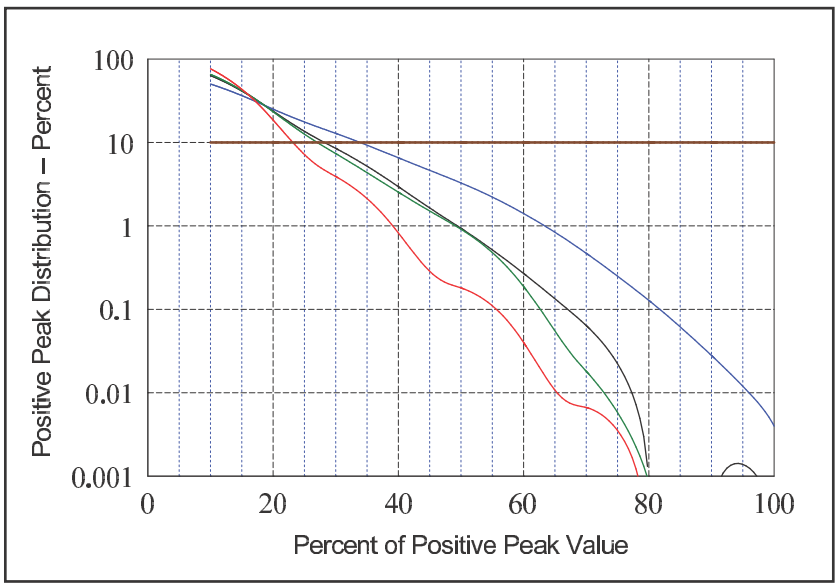


Figure 1: This graph shows generally good dynamics of the four tracks listed in **Table 1**. Paul Desmond's "Autumn Leaves" is represented by the blue line, Diana Krall's "Dancing in the Dark" is the black line; Dave Brubeck's "It's a Raggy Waltz" is the green line; and Willie Nelson's "Always on My Mind" is represented by the red line. The horizontal brown line is white noise, included for comparison. Its dynamic range is zero, which would illustrate 100% compression—not very musical.

Loudness

Streaming services have supposedly become more aware of maintaining equal loudness between tracks. Usually this is not a problem when the tracks are from the same album. But when a playlist includes tracks from different albums, sometimes many albums, it can become a concern.

Spotify includes a loudness normalization option that you can enable or disable in your account's Preferences screen. Just go to Settings/Advance Settings and, on the Playback menu you will see the option to "Set the same volume level for all songs"—activated by default.

I captured a Spotify playlist named "Sleep" that was advertised as: "Fall into a deep relaxing sleep with these soothing sounds." Loudness normalization would certainly be useful here! I used two software loudness analyzers: MusicScope and the free MLoudness Analyzer, to look at the first six playlist tracks—all from different albums.

My results are summarized in **Table 2**. Several values in the "S" column were close and I verified this measurement with a listening test. I adjusted the amplitudes of tracks 1, 4, and 5 to have "S" values close to -19 LU. The listening test convinced me that all six tracks seemed to sound as if they were at about the same level, so volume control "riding" wasn't necessary.

MusicScope and MLoudness Analyzer implement EBR-R128 (European Broadcast Unit) and ITU-R BS-1770 (International Telecommunication Unit) loudness measurement algorithms. In **Table 2** the "I" columns are the same and the Range value is MusicScope "I" minus "M."

Dynamic Range

Figure 1 shows a plot of the positive-peak dynamic range of the four tracks in **Table 1**. The figure was created using the dynamics program that I wrote and described in my article, "Signal Dynamics and Loudness," (*audioXpress*, May 2015). The horizontal brown line at 10% is white noise. It has no dynamic range. The closer a music track's dynamics come to a horizontal line, the greater the compression. Even though each music track is different, overall their dynamics look good.

Some loudness measuring programs are calling the peak-to-loudness ratio (PLR) dynamic range. This is incorrect. PLR is measure of the compression with smaller numbers being higher compression. Dynamic range is the ratio of the lowest audible sound to the highest, both measured either as peak or RMS.

In conclusion, Spotify has some rough spots. But, as I mentioned in the Sidebar, Ogg Vorbis is a cut above MP3. I expect the Spotify music quality will continue to improve as streaming technology becomes more mature.

Ogg Vorbis

Ogg Vorbis is a lossy audio compression format that is free, open, and unpatented. It was designed to replace the proprietary, patented audio formats such as MP3, VQF, and AAC. Its somewhat strange sounding name has this meaning: Ogg is the name of the Xiph.org container for audio, video, and metadata. Vorbis is the name of the specific audio codec.

The Ogg container is the program loaded by your audio editor or Digital Audio Workstation (DAW) to implement the Vorbis codec. The FLAC codec, and possibly others, can also be embedded in the Ogg container.

Although Vorbis is lossy and discards some data to accomplish its compression, it uses a variable bit rate algorithm. This means it is selective about which data is discarded. More data is retained where the audio complexity needs it and less is retained where it matters less. For a given file size, Vorbis is reputed to sound better than MP3.

The Vorbis and MP3 codecs are very different. Although there is some overlap, they each discard different parts of the audio. This means that converting Vorbis to MP3 or vice versa is not a good idea because you will likely produce a file that sounds worse than your original. A Vorbis to .wav or another lossless format is satisfactory because all of the original data will be retained, although it won't be "CD-quality." For more information, visit www.xiph.org.

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A New Generation ICEpower



In this new article series, *audioXpress* will explore the history, recent developments, and new-generation products from some of the most innovative amplifier platform companies serving all segments of the audio industry. We start with a visit to Danish amplifier powerhouse ICEpower and check out its most recent and advanced Class-D platform, ICEedge.

By
Ward Maas
(The Netherlands)

The history of ICEpower, interesting as it is, may not be known by everyone. It is interesting because it clarifies where the company stands today and provides a look at its plans for the future.

ICEpower, founded by Karsten Nielsen, has close ties to the Danish iconic company, Bang & Olufsen. In 1999, Nielson's doctoral work at the Technical University of Denmark drew the attention of the Bang & Olufsen management, resulting in a joint venture that created the foundation for a new business. At that time, Class-D technology was maturing and generating increasing interest in the audio world. The advantages of high efficiency, high power density amplifiers made possible by the new Class-D topology, were clear. (Nielsen's 1998 PhD thesis is a good introduction to Class-D amplification and is still available on the ICEpower website at www.icepower.dk/en/technology/publications.)

Symbiotic Relationship

Having a close relationship with a much larger company makes it possible to access their resources, design tools, technical experience, and business knowledge. A huge advantage for a new small company, it enabled ICEpower to quickly become one of the leading companies in the amplification world. Starting in high-end niche markets, ICEpower developed a range of premium quality audio power conversion engines used by some of the world's most prestigious brands, such as Bowers & Wilkins, Pioneer, Alpine, Samsung, Audi, and Bang & Olufsen's own products. Gradually, the company built a powerful OEM business, delivering platforms used in low power systems (e.g., mobile phones) to large scale pro audio line array systems.

In early 2016, realizing that the business could expand further on its own, a decision was made to separate from Bang & Olufsen. In April of 2016, a management buyout supported by Industry Development (Industri Udvikling), a Danish private equity company, allowed ICEpower to become an independent entity.

One of the first strategic moves of the newly formed organization was to merge with Audio Bricks

and acquire its holding company, Bolecano Holding AB, from Sweden. The founders of Audio Bricks, Patrik Boström and Lars Press Petersen were well-known to ICEpower. Boström had founded Anaview in Sweden in 2001, only two years after ICEpower was founded in Denmark. And, Petersen held the position as Innovation Manager at ICEpower from 2003 to 2007. Bringing both men on board created a strong ICEpower management team, together with Kjeld Lindegaard Andersen and Mads Emil Solgaard, which gave ICEpower considerable knowledge of Class-D amplifier development and cutting-edge ideas, unique among Scandinavian companies. **Photo 1** shows Boström and Solgaard at ICEpower's new temporary lab facilities.

ICEpower was once again able to act with the speed and the agility of a small company, while being able to leverage its unique market position and strong reputation to its advantage, combining a mature portfolio of technology and products in consumer, professional, automotive, and mobile audio applications, with real market knowledge. It also gained some of the most recent amplifier technology platforms.

ICEpower Innovations

ICEpower has developed low-power Class-D chips for mobile audio applications (e.g., phones and laptop computers), for which it developed more than 12 ASICs (see **Photo 2**). That development provided ICEpower with a solid foundation for in-house chip development, which it then manufactured for companies such as Samsung, Sanyo, and Toshiba. The first ICEpower Class-D chipset to be used for ICEpower's own amplifier products, the ICCx chipset, was developed 15 years ago (2002). Already a very advanced chipset, the ICCx has served ICEpower for many years, shipped in millions of amplifiers, and is still in use for many products.

Many years of accumulated insight lead to the recent development of a totally new chipset generation, the ICEedge, which consists of a dual-channel controller IC and two driver ICs. During the development of this chipset, which took seven years, a string of research projects were initiated leading to the filing of an additional six patents. The aim was to simultaneously provide truly outstanding high-end audio quality and flexibility. ICEpower also focused on protection features, giving the amplifiers containing these chips a nearly unbeatable product reliability at a reduced product cost. ICEpower regards the ICEedge chipset as the most advanced Class-D chipset in the world. With this technology platform, amplifiers with a range of 50 to 7,000 W can be built.

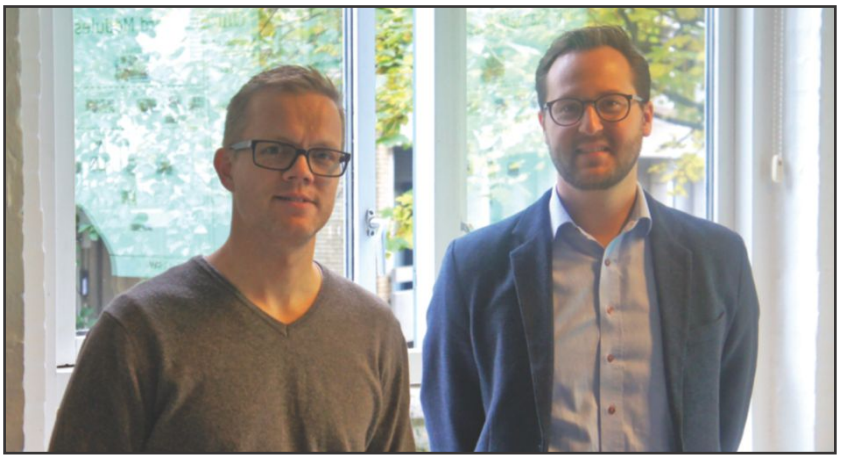


Photo 1: Patrik Boström and Mads Emil Solgaard welcomed audioXpress to their temporary lab facilities, after separating from Bang & Olufsen.

Instead of trying to put everything on one chip, the ICEdge chipset was deliberately split into a driver part and a controller part, enabling the use of a low-noise process for all the sensitive analog parts in the signal chain (e.g., high-quality audio op-amps and comparators). This way, extremely low noise levels, which normally are only achievable with discrete analog audio components, could be reached.

The gate drivers feature a high breakdown voltage, a high voltage, and current slew rates. The method—a silicon on insulator process—enabled a gate driver design capable of up to ±150 V supply voltage with a low propagation delay of just 110 ns.



Photo 2: ICEpower developed more than 12 ASICs for mobile audio applications (e.g., laptops) used by many of the large volume manufacturers.

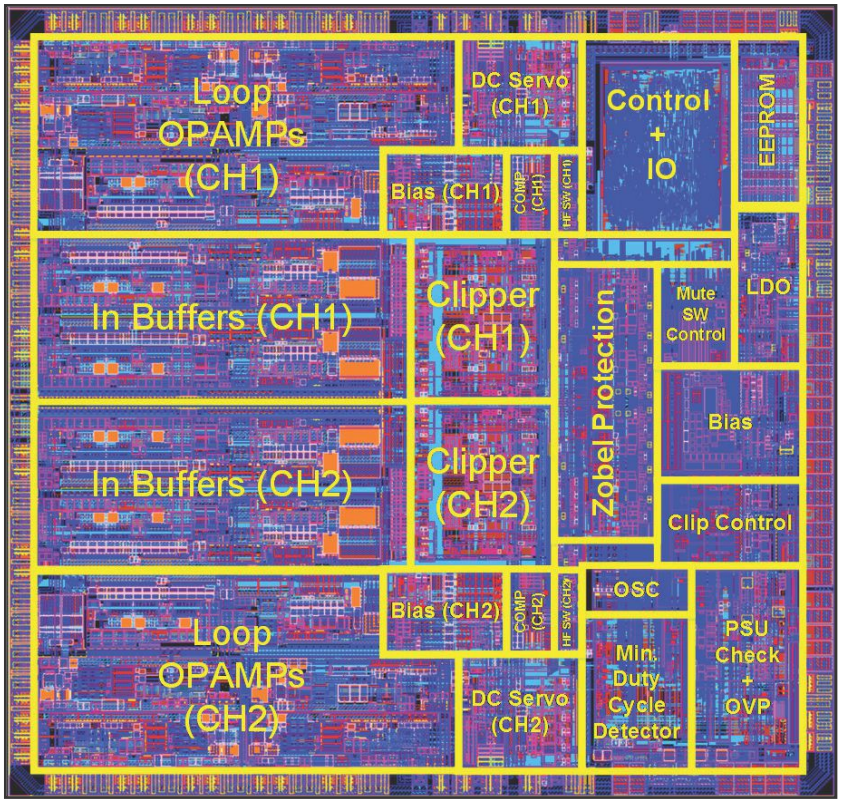


Photo 3: In the ICEedge controller, ICEpower used a dual-mono design, with the two channels being completely separated and with the physical layout being fully identical.

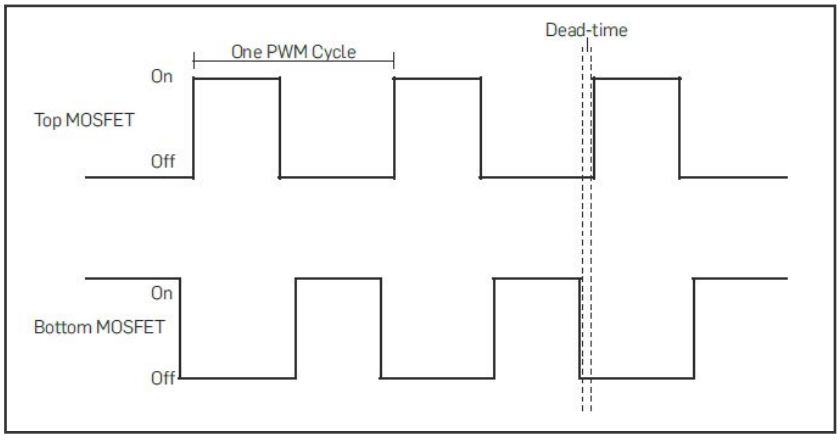


Figure 1: A small delay is introduced by the ICEedge Driver in the MOSFET gate drive timing, ensuring that one MOSFET is turned off before the opposite MOSFET is turned on. This prevents a momentary short circuit of the power supply and increases efficiency. This delay is shown in the figure as dead-time.

The ICEedge Controller is a fully integrated pulse-width modulation (PWM) modulator and a system controller for dual-channel self-oscillating Class-D switching amplifier applications (see **Photo 3**). The controller has a fast communication interface for two ICEedge drivers. All passive loop components are external, allowing flexibility, precision, stability, and linearity.

The controller includes 16 high-performance audio op-amps and two high-speed comparators, high impedance fully balanced audio inputs, and DC servos for a low DC offset voltage at amplifier outputs. In terms of protection features, the controller combines a precision clipper and supply voltage surveillance circuits, overcurrent protection, thermal management, over/undervoltage, a modulation index, and Zobel protection. The design allows controlled pop reduction during start up and shut down, simple pin control and diagnostic information, with no microcontroller required for full operation and it features an I^2C interface for control and system programming.

The ICEedge driver IC is a fully integrated gate driver for Class-D switching amplifier applications using discrete external MOSFETs. The design uses high voltage technology for amplifiers with up to ±150 V power supply voltage, which enables up to 7 kW into 4 Ω and features a high-speed (110 ns) 2.2/1.5 A driver for direct drive of MOSFETs. The drive capability can easily be extended with the addition of an external active drive circuit and the design allows for continuous dead-time compensation, ultra-fast cycle-by-cycle output current limit, and optimized dead-time compensation for both low- and high-side drive.

There's also a built-in high-speed measurement system for accurate MOSFET temperature measurement and amplifier protection, temperature diagnostic information including the possibility of early warning for fan control, and the ability to implement an active anti pop system during amplifier start up. The design also allows both single and split supply for the lowest system cost. A few features are very interesting and deserve a closer look.

Special Features

Continuous Dead-time Compensation (CDC). To get a high efficiency and low distortion it is necessary to have a switch behavior of the power MOSFETS as ideal as possible. In Class-D output stages, the two output MOSFETs can, if controlled incorrectly, short circuit the power supply at each output transition. This will cause increased losses and electro-magnetic interference (EMI) and also

possible amplifier failure. Unfortunately, a MOSFET's switching-on and switching-off behavior is not symmetrical. The result is the need for a rather large so called "dead-time," the time needed to ensure no cross conduction (i.e., both Mosfets conducting at the same time) occurs. (see **Figure 1**).

The solution is a small delay introduced by the ICEedge Driver in the MOSFET gate drive timing, ensuring that one MOSFET is turned off before the opposite MOSFET is turned on. This prevents a momentary shortcircuit of the power supply and increases efficiency. However, to keep distortion at the lowest possible level, very accurate timing is required.

Fortunately, the ICEedge Driver's dead-time setting is very accurate, leaving only minute variations in production and across applications. But since the dead-time for the total power stage also depends on the switching behavior and the timing of the power MOSFETs introduced by complex speaker loads, dead-time related distortion still exists, especially in high power amplifiers.

To compensate for this phenomenon, the ICEedge Driver provides an ICEpower proprietary CDC mechanism that adds small continuous compensating adjustments to the gate driver output timing derived from the previously measured PWM switch timing. In other words, based on the actual timing of the power stage MOSFETs, the ICEedge Driver will maintain the optimal timing of the full power stage and thus provide optimum low distortion properties.

*Switching on...*The driver chips have an on board pre-bias setting system avoiding switch-on pops. Before the amplifier is released the bootstrap capacitor for the upper MOSFET is pre-charged by a current source. This has to be done correctly, otherwise the PWM sequence would not be starting at 50% resulting in an audible pop at the start-up sequence (see **Figure 2**).

*And switching off...*A nice feature is the amplifier's controlled switch-off behavior. If you perform an abrupt shut-down of the amplifier, the PWM cycles will stop in the middle of one cycle. This most likely results in a pop noise. The solution used here is switch-off frequency sweep. When the amplifier is shut down, a high frequency sweep with a predefined switch stop time is activated. By sweeping the PWM to a high frequency, the remaining "pop-energy" gradually reduces to nothing and ensures a pop-free switch-off of the amplifier (see **Figure 3** and **Figure 4**).

Temperature... check! ICEedge measures the temperature in eight zones and gives an early warning before the temperature limit is reached.

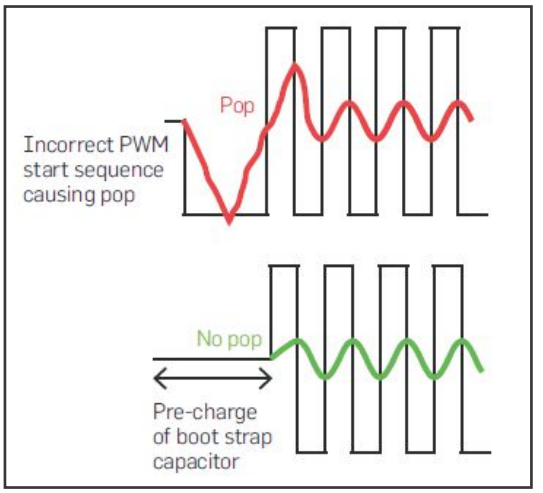


Figure 2: Some designs make no prevention against pop noise. This results in a pop at the beginning of the PWM cycles. ICEedge implements an advanced solution to eliminate pop noise.

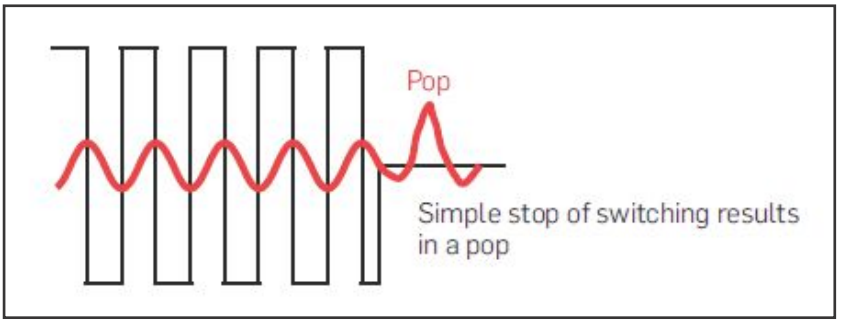


Figure 3: An abrupt shut-down of the amplifier causes the PWM cycles to stop in the middle of one cycle, likely resulting in a pop noise.

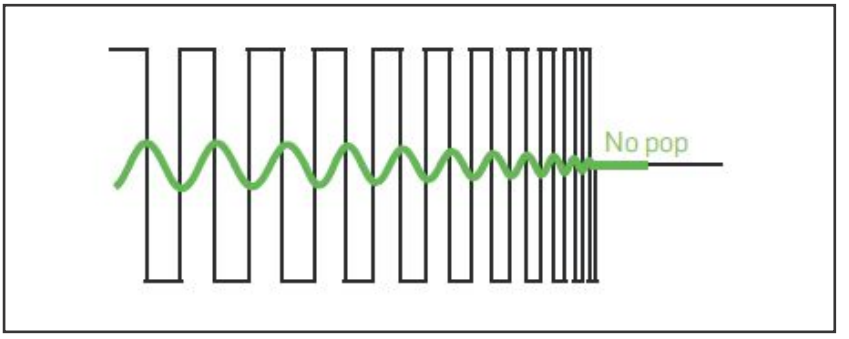


Figure 4: ICEedge eliminates pop noise.

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.

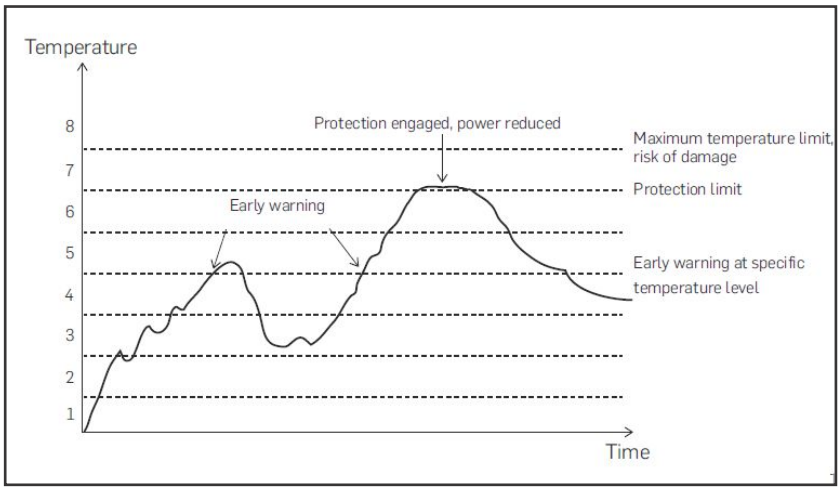
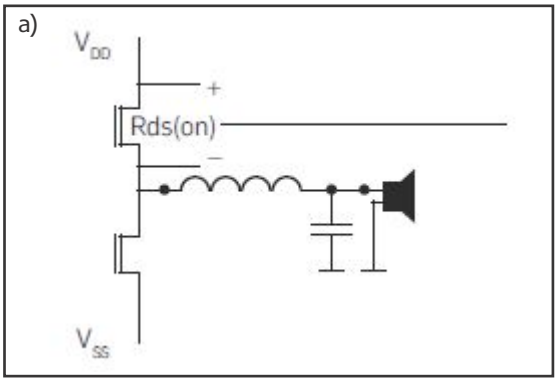
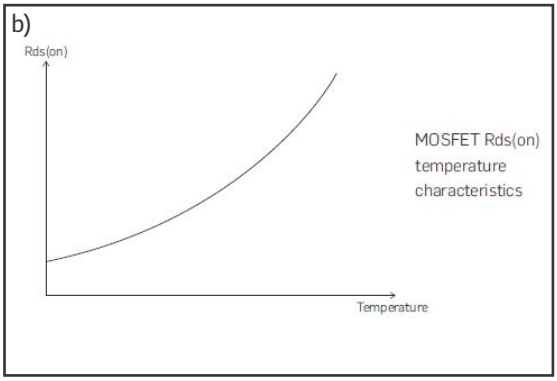


Figure 5: This is an early warning temperature capture.

Figure 6: ICEedge measures the current by monitoring the voltage drop across the MOSFET Rds(on) to avoid damage and keep the music playing (a). However, Rds(on) is highly temperature dependent (b).





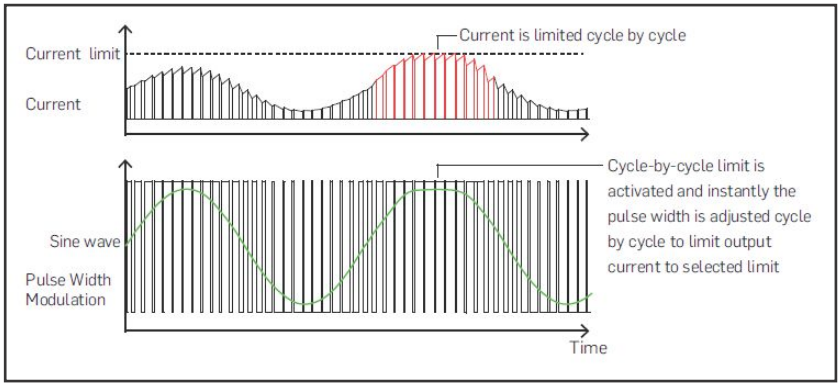


Figure 7: ICEedge has a Temperature-Compensated Cycle-by-Cycle Current Limit (TCCCL).

At a certain temperature, the system will send a warning about the rising temperature. This way, there is time to react and limit the power before any risk of shutdown or damage. Should the temperature keep rising, a protection system will activate and gradually turn down the power until the temperature has reached safe operating levels. Most important is that by organizing temperature control this way an amplifier will not break down from overheating, thus ensuring its continuous service (see **Figure 5**).

Of course, this depends greatly on where and how you measure temperature... Usually the MOSFET temperature measurement does not correspond to the actual die temperature and is delaying the actual temperature of the die itself as it is not measured there. There is the risk that an over-temperature warning comes too late and damage to the MOSFETS is already done.

ICEedge is capable of measuring the temperature directly at the MOSFET's drain terminal with no electrical isolation. This means that the actual temperature information is available with the most minimal delay possible, giving the shortest reaction time of the temperature control system.

Current Limiting. MOSFETs are rather sensitive to overcurrent. If the limit is exceeded, a MOSFET will be damaged. Continuously monitoring a MOSFET during operation is a good idea. In principle, that can be done rather easily by monitoring the voltage drop across the MOSFET Rds(on), which can be seen in **Figure 6a**. However, this Rds(on) is highly temperature dependent (see **Figure 6b**). The ICEedge Temperature-Compensated Cycle-by-Cycle Current Limit (TCCCL) measures the temperature and compensates for the changing Rds(on), which can be seen in **Figure 7**.

Monitoring the current cycle-by-cycle and the actual temperature of the MOSFET die enables a direct response if the current limit is reached by lowering the current to ensure that the amplifier will not be damaged. In this way, it is also possible to use better specified MOSFETs with lower current ratings or filter coils with more desirable specifications. Of course, this can also be used to save costs as you can integrate the parts at a lower price point without risk of damage.

The Possibilities

Marketwise, implementing ICEpower amplifiers provides a range of choices from existing and new platforms with different form factors and solutions for higher volumes, higher quality, and more power ratings. Current amplifier modules, from small to large sizes, allow a choice of single-channel, dual-

channel and multichannel solutions, with or without power supply.

ICEpower's AS Series includes a range of full-featured amplifiers from 100 to 1,200 W with universal mains and ErP/Energy Star compliant standby (see **Photo 4**). The ASX Series features power options from 25 to 700 W, ideal for more cost-focused products with less need for features. The ASP Series features ICEpower's most powerful Class-D modules. The recent 1200AS1 module features a power rating of 1 × 1,200 W in 4 Ω and the 1200AS2 includes two 1,200 W channels in 4 Ω (see **Photo 5**). The company has also introduced the new ICEtheater7, a one-board seven-channel high-quality amplifier, leveraging its newest Class D DualLoop3, topology (see **Photo 6**).

ICEpower also offers the newly developed, full featured 100AS1 and 100AS2 modules (see **Photo 7**). The modules, a one- and two-channel 100 W amplifier, enable high performing and high volume applications, which together with the 1200AS1/2 modules completes the requirements for 100 to 1,200 W at a highly competitive price point. These new models feature DualLoop3 Class D topology, universal mains, a standby functionality, a regulated power supply, PFC, monitoring, diagnostics, a comprehensive protection scheme, and several more features.

With the newly acquired Audio Bricks product range, ICEpower offers a second option that enables designers to incorporate an Audio Brick module into a manufacturer's own design.

Since the merger with Audio Bricks, ICEpower has decided to integrate its products and technology as ICEpower products, under the ICEbricks series.

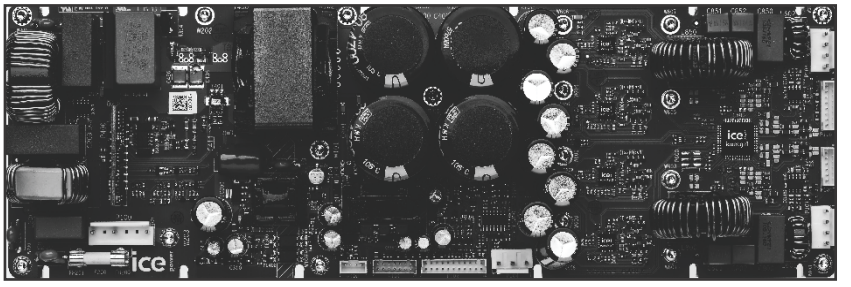


Photo 4: An example of an AS module from ICEpower, with a power output of 2x 1200 W in an ultra-compact and lightweight package.

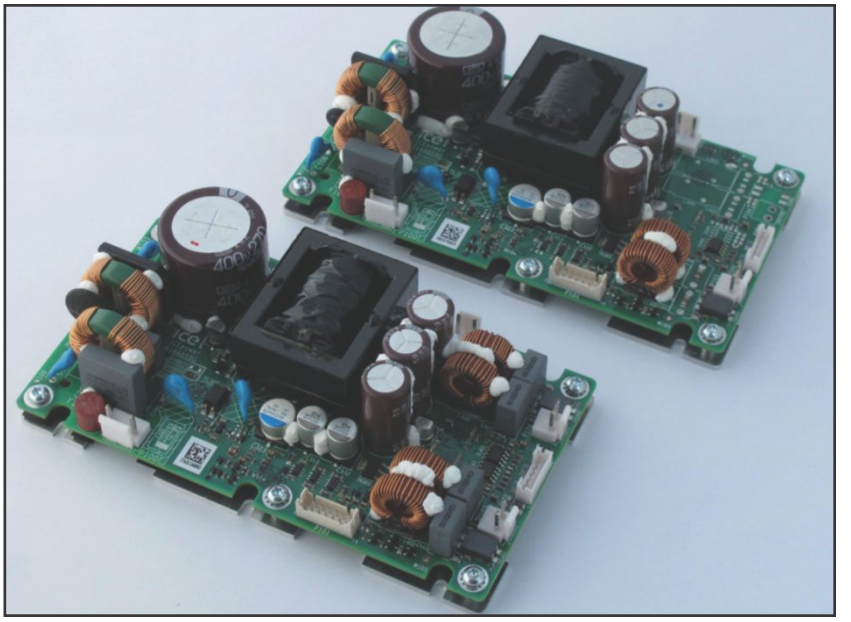


Photo 5: The recent 1200AS1 module features a power rating of 1 × 1,200 W in 4 Ω and the 1200AS2 includes two 1,200 W channels in 4 Ω.

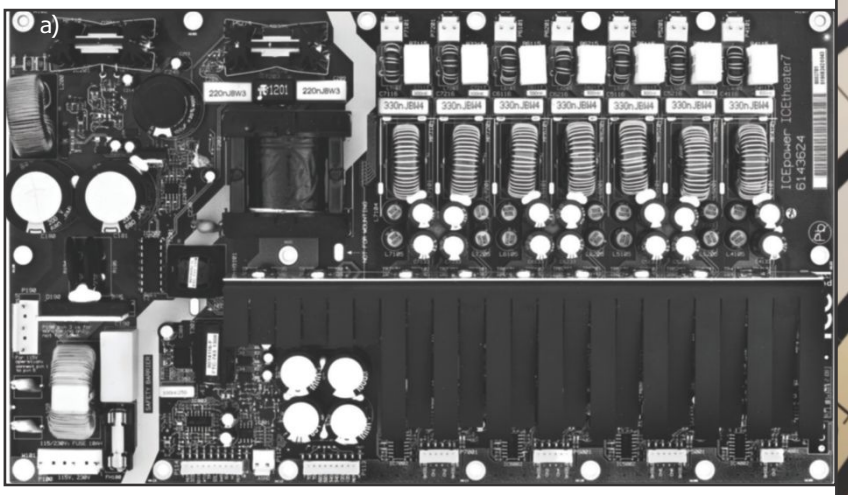




Photo 6: The new ICEtheater7, a seven-channel high-quality amplifier using the newest Class-D DualLoop3 topology (a) and a custom OEM 16 channel amplifier, using ICEmatch modules, which was on display at the NAMM Show 2017 (b).

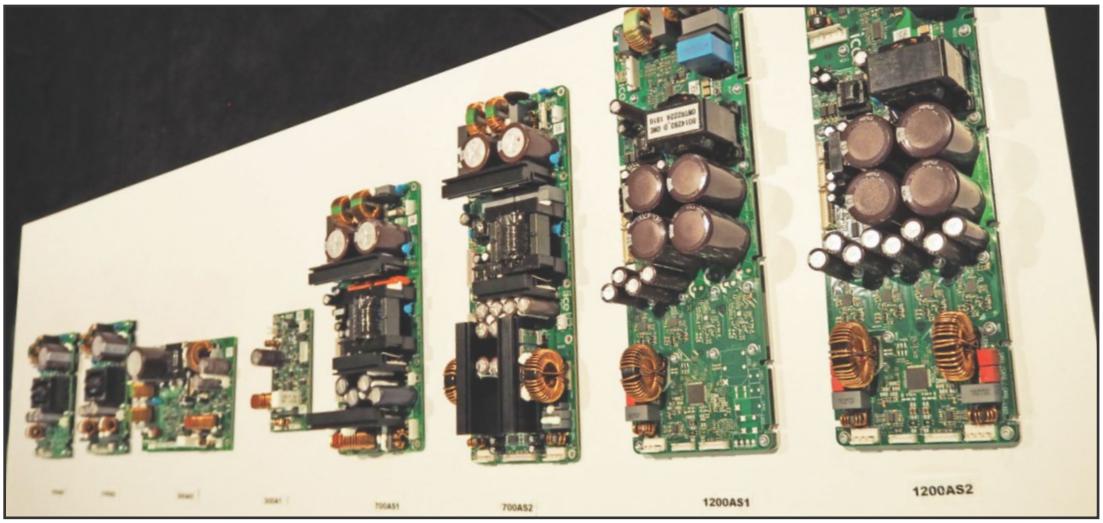


Photo 7: These are the A1 and AS1/2 modules, which are the newest generation of ICEpower modules and were on display at the NAMM Show 2017.



Photo 8: ICEpower has set up in temporary headquarters until it moves to a permanent location in Copenhagen, Denmark.

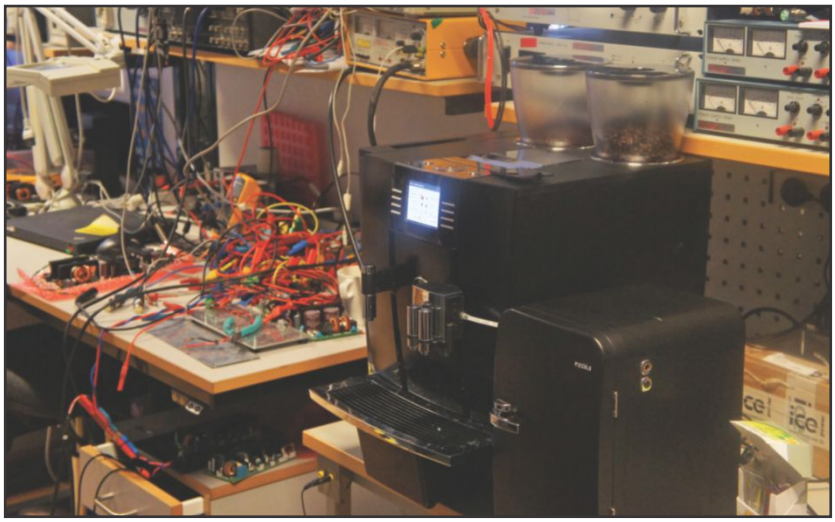


Photo 9: This new coffee machine now powers ICEpower's temporary premises.

We will explore the possibilities with ICEbricks in a future issue of *audioXpress*.

ICEpower also offers a third option for manufacturers to use the ICEedge chipset and design their own products. A prerequisite is that the manufacturer be willing to invest in its Class-D knowledge on a much larger scale.

How does this technology work in practice? In a future article in this series, *audioXpress* will take a closer look at the new ICEpower flagship: the 1200AS2 module. This is a 2x 1,200 W in 4 Ω amplifier with a built-in power supply that can be used as a pro audio workhorse or potentially in high-end amplifier applications.

Its separation from Bang & Olufsen has forced ICEpower to move to a (temporary) new building, which *audioXpress* visited. While the company prepares to move to a completely new location in Copenhagen, Denmark, we found a company deeply focused on continuous research with no lack of ideas and aspirations. The move means that a lot of test equipment has to be relocated and the current laboratory was showing a happy relocation atmosphere (see **Photo 8**). As some of our photos document, development and testing continues as usual.

One of the most successful tests was already the new coffee machine, which we found was the center of attention in the lab (see **Photo 9**). While the old one stayed behind in the division of joint property with Bang & Olufsen, the new one is already supporting ICEpower with new fresh ideas.

For more information about the ICEedge Class-D chipset visit www.ICEpower.dk.

A New Audio Amplification Powerhouse

On May 1, 2016, Keld Lindegaard Andersen, CEO, and Mads Emil Solgaard, VP of Sales & Marketing, together with the private equity fund, Industry Development (Industri Udvikling), acquired ICEpower A/S from Bang & Olufsen A/S. The ICEpower management team had 50% of the company and Industri Udvikling another 50%. At the time of the separation from Bang & Olufsen, the ICEpower management team and Industri Udvikling stated their intention was to develop ICEpower further and consolidate its position as a leader in Class-D audio power conversion.

In September 2016, Danish company ICEpower, now independent from Bang & Olufsen, announced the decision to merge with Audio Bricks, and to acquire its holding company, Bolecano Holding AB, from Sweden. While ICEpower is highly recognized in the market as one of the leading brands in audio amplification, Audio Bricks was a relatively new company, created in 2014 by Patrik Boström and Lars Press Petersen. Boström and Petersen have now become partners and shareholders of ICEpower, respectively holding the position of CTO of Amplification and CTO of Research and SMPS at ICEpower.

The two founders of Audio Bricks are both familiar with Class-D power electronics. Boström was one of the founders of Anaview AB, in Sweden, while Petersen was Innovation Manager at ICEpower from 2003 to 2007.

Boström has a BSc in Electrical Engineering and developed amplifiers and power supplies for OEM automotive, high end, consumer, home theater, and sound reinforcement applications. Boström started his electronics career at Ericsson in Stockholm designing switch mode power supplies for Telecom applications. In 2004, he founded Anaview, which became a well-known supplier of complete high-quality powered Class-D modules. Boström is the inventor of several patents within the field of Class D and loop design.

Petersen holds a PhD in Power Electronics from the Technical University of Denmark. In 2014, he started Audio Bricks with Boström. Petersen is an innovator and power supply specialist. Prior to that, Petersen worked for ICEpower from 2003 to 2008 and was deeply involved in the research of new topologies and system designs. Between 2008 and 2011 he headed a start-up business, Upcon Technology, which specialized in switch mode power supplies focusing on high efficient telecom and server power supplies. From 2012, he worked as a senior advisor and researcher at the Technical University of Denmark, in the field of switch mode electronics. Petersen has participated in more than 30 scientific papers and patent applications.

"We share the same ideas on development of quality sound by technology and combining technologies to reach our goals. Already in the early zeroes we worked together, became friends and discussed how we could join forces. The timing is right for all of us now," stated Keld Lindegaard Andersen, ICEpower's CEO.

For more information, visit ICEpower or Audio Bricks at www.ICEpower.dk and www.a-bricks.com.



The Audio Bricks modular audio amplification platform was introduced in 2015, at the 139th International Audio Engineering Society (AES) Convention in New York.



Even though Audio Bricks has continued to exist as a separate entity, ICEpower displayed its new ICEbricks designation for the Audio Bricks modules—Engine-400, AUX-plant, and Motherboard-1— at the NAMM Show 2017.

Finding a Better Bass Trap



In several previous Sound Control articles, we have discussed using bass traps to mitigate the effects of resonant modes in control rooms, listening rooms, and home theaters. We have also mentioned the use of tuned absorbers, often employing Helmholtz resonators, as well as non-resonant absorbers operating according to one or more of several principles. In this article we look closer at what makes a bass trap... better.

By
Richard Honeycutt
(United States)

Resonant absorbers using the Helmholtz principle are designed so that at the frequency to which they are tuned, they provide a high particle velocity in the air at the entry to the absorber. Typically, absorptive material is placed at this location, and the high-velocity air particles moving through the absorptive material lose much of their energy as friction converts the kinetic energy to heat.

Resonant Absorbers

It is important to note that the amount of absorption in a resonant absorber directly affects the Q, which is inversely related to the bandwidth of the absorption. Thus, it is possible to build a Helmholtz trap whose Q is so low that a curve of absorption vs. frequency would lead one to believe that it is a non-resonant absorber; yet it does operate on the principles of a resonant device.

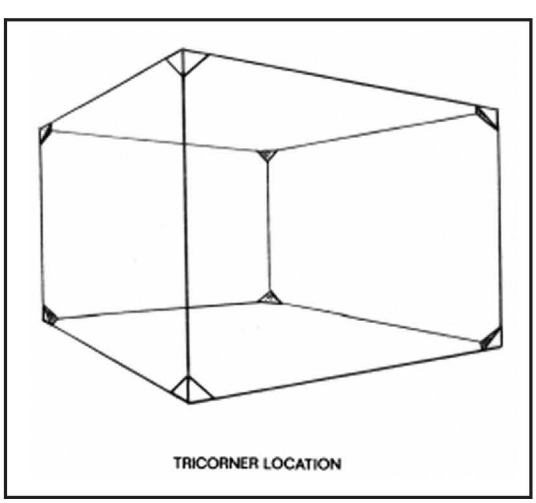


Figure 1: : All room modes have sound pressure maxima at the tri corners. (Courtesy of Art Noxon of Acoustic Sciences Corp)

A high Q Helmholtz bass trap can be created similarly to the way Acoustic Sciences Corp. makes its Tuned TubeTraps. The company creates a large cavity above the opening of the Helmholtz resonator so the resonator is able to "breathe" freely, which maintains its high Q response curve. The surface of the upper cavity is sealed off with a sound absorbing acoustical panel.

In this type of "resistor-capacitor" bass trap, air that flows through the resistive surface creates friction which absorbs energy and at the same time the air flow raises the back pressure inside the cavity, which slows down the influx of air into the cavity, hence weakening the absorption. At the tuned frequency, the pressure fluctuations inside the upper air cavity stimulate the resonator into resonance. The motion of air in the open mouth of the lower resonator cavity "cancels" the buildup of back pressure inside the cavity, so the entire sound absorbing surface of the sound trap has no back pressure and becomes extremely sound absorptive.

Another type of resonant absorber uses

diaphragmatic absorption in the form of thin wood or gypsum panels over a cavity containing material such as low-density fiberglass. These devices utilize a mass-spring resonance in which the mass is the mass of the area of thin panel between supports, and the spring is the compression of the air in the cavity or a separate mechanical spring. Diaphragmatic absorbers are usually most effective in the 125- and 250-Hz octave bands. An in-depth discussion of these and other types of bass traps is given in F. Alton Everest's *Master Handbook of Acoustics*.

Non-Resonant Absorbers

Non-resonant absorbers are typified by the fiberglass wall panels that are commonly found in auditoria, theaters, and cinemas. They function by friction between air molecules in the incident sound wave and fibers in the wall panel. The friction converts acoustical energy into heat. If there is an air cavity behind the sound panel, the panel can "breathe" more deeply and provide absorption at lower frequencies.

Other non-resonant absorbers include "limp-mass" absorbers. There are three types of limp-mass acoustical devices. The first is just a limp mass, such as the Noise S.T.O.P. vinyl barrier (www.acousticalsurfaces.com/noise_barrier/vinylbar.htm). These devices are used for noise isolation (acoustical insulation), but not for reverberation control.

The second is a limp mass over a fiberglass panel. It adds mass to the air movement in front of the panel, which improves its low-frequency absorption while backscattering the treble. Acting via the mass law these limp-mass bass traps reflect the treble and absorb the bass.

The third type is a damped limp-mass absorber, also placed over an air cavity, usually filled with fiborous. They convert acoustical energy into heat through internal friction occurring during the bending of the damped limp-mass sheet. Limp-mass bass traps often perform effectively at low frequencies where large thickness or deep air cavity wall panels would otherwise be needed.

Absorption

In room acoustics, speakers make sound, the room stores the sound, and the absorption disposes of it. How loud the reverberation is depends on how much absorption there is in the room. If absorption is strong, the room reverb noise level stays fairly low and the direct signal from the speaker can be enjoyed. If absorption is weak, the room reverb noise level grows loud and masks our ability to hear much of the direct signal from the speaker at all. In a typical untreated listening room, the "running reverb" is about 10 dB louder than the direct signal from the speaker. Since the running reverb is a chaotic version of the direct sound, it is a perfect sound masking noise floor. In a well-developed listening room the running noise floor might be 1 or 2 dB below the direct signal.

Of all room reverb storage mechanisms, the effect of room modes lasts the longest and is the loudest. These modes have particular frequencies

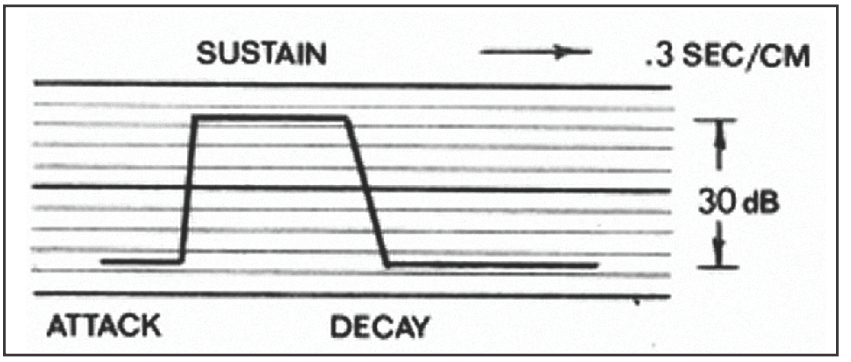


Figure 2: The envelope of a tone burst shows a typical steep decay. (Image courtesy of Art Noxon, Acoustic Sciences Corp.)

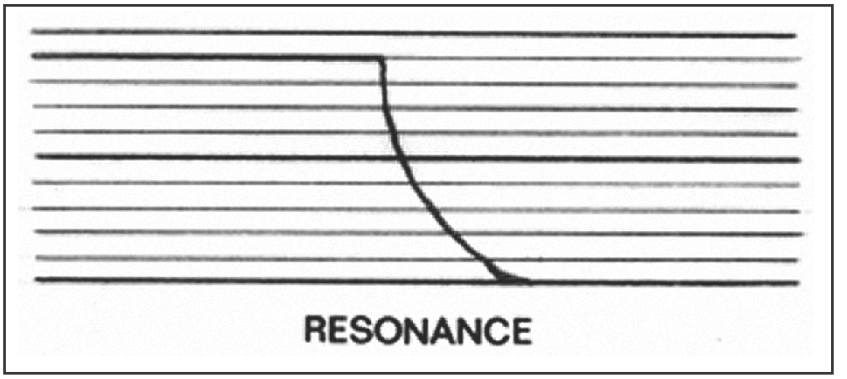


Figure 3: The room, and especially resonant modes, lengthen the decay. (Image courtesy of Art Noxon, Acoustic Sciences Corp.)

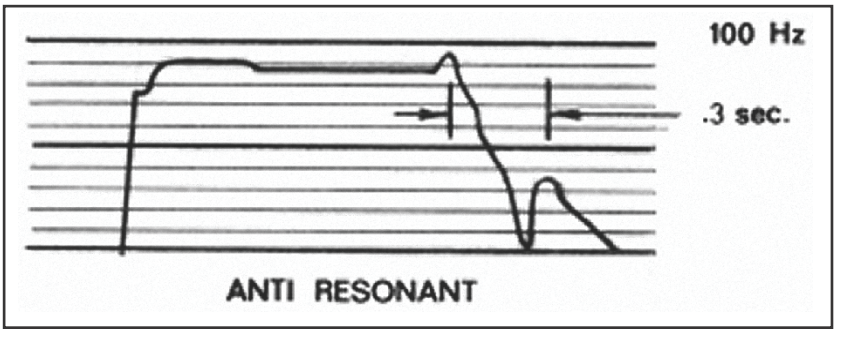


Figure 4: A sound burst, whose frequency is near, but not equal to, a modal frequency, produces a more distorted pulse.

based on room dimensions. We can use tuned traps, which absorb sound at only one frequency, or we can use broadband traps that absorb sound throughout a frequency range. In either case, we place these traps in the high pressure zones of the room resonances, which are typically walls, floors, or ceilings, and the corners between the walls, floors, and ceilings.

When a sound wave strikes a rigid boundary, the pressure doubles, which corresponds to a 6 dB increase in sound pressure. Wall traps are operating on a +6 dB pressure location. Corner traps are operating in a +12 dB location; and tri-corner traps, in a +18 dB location. Remember that the amount of sound power (milliwatts) a trap absorbs is proportional to how loud the sound is, so we like to put bass traps where bass pressures are the highest.

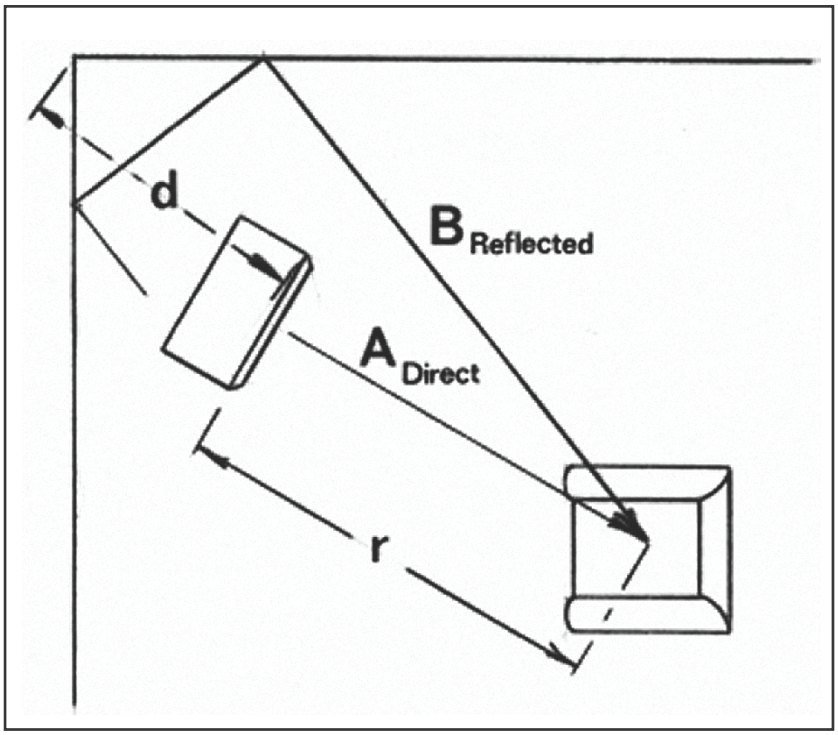


Figure 5: At frequencies near a mode, sound reflected from a corner behind a speaker can cause coloration of the sound. (Image courtesy of Art Noxon, Acoustic Sciences Corp.)



Photo 1: The TubeTrap is designed to control bass response. Three TubeTraps are shown in this photo. (Photo courtesy of Acoustic Sciences Corp.)

Broadband Traps

Broadband traps are convenient because the room does the "tuning" by delivering extra high pressure zones to the trap's location. The same trap can damp any number of modes as long as the mode has a pressure zone where the trap is located. Accordingly, the same trap can be used in different locations to catch mode pressure zones of different frequencies.

Because of the different operating principles, the application approaches for non-resonant and resonant absorbers are somewhat different. Non-resonant absorbers provide a total absorption (measured in Sabins) proportional to the area. For example, 4" thick Owens Corning #703 fiberglass mounted against a hard wall absorbs about 99% of the incident acoustic energy in the 125 Hz and higher octave bands.

To get enough absorption to achieve a reverberation time (RT) of 0.5 seconds in a 20′ × 20′ × 10′ control room, about 280 ft^2 of fiberglass would be needed, assuming about 10% average "residual" absorption due to other room materials and furnishings. While this much material can be applied to walls with little effect on the room's functionality, it is difficult to apply anywhere near this amount of fiberglass in the room corners, especially the tri-corners.

Resonant traps can decrease the sound pressure near their mouths at their tuning frequency for an air volume about 50% larger than that of the air inside the resonator. A resonator with a 4" diameter opening and a 20" diameter by 3′ long cylinder volume will resonate at about 40 Hz. The sound cancel effect is audible within a radius of about 18" but the resonator's effect is long reaching. It will noticeably level out the pressure peaks and valleys caused by standing waves, up to 6′ from the trap.

The tri-corners where two walls and the floor, or two walls and the ceiling intersect in a room are locations at which pressure maxima exist for all modes (see **Figure 1**). Since resonant absorbers function by using sound pressure to produce high velocity, leading to high-frictional energy losses, they are most effectively located in the tri-corners.

Non-resonant absorbers such as flat sound panels work by interaction between air-particle velocity at the absorber surface and the absorptive material's internal friction: the sound pressure per



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se does not enter the equation. Thus, there is no particular advantage to locating fiberglass panels in corners, as compared to other places in the room.

Room Decay

Room decay "stretches" the decay of the tone burst into an exponential decay. In acoustically developed control and listening rooms, this stretching is insignificant above the Schroeder frequency but can become significant at frequencies near a room mode. (The Schroeder frequency is the crossover frequency between diffuse reverb behavior in the room at higher frequencies and discrete room modes at lower frequencies. While this frequency can be precisely calculated for any room, the transition from modal to diffuse behavior is always gradual.)

There are two effects that can come into play. If the sound has the exact frequency of a mode, only increased decay (and its consequent degraded transient response) will occur. But if the sound's frequency is near—but not equal to–the frequency of a mode, there will be an initial decay followed by a reappearance of the sound at a level that may be within 10 dB of the original sound level. This second burst of sound will have the frequency of the mode, not the original sound frequency. Thus, the added "bass tail" is out of tune with the program sound, and room coloration has occurred. **Figures 2–4** illustrate this effect.

Low-frequency transients can encounter a second problem that is less commonly known. Consider a speaker located in a corner, reproducing a tone burst: a test tone that reasonably simulates the attack of a kick drum or tympani. (Other instruments producing very low-frequency sound tend to have more gradual attacks, or to have most of their acoustic energy in the harmonics, with the level of fundamental many decibels lower.)

As shown in **Figure 5**, the listener hears both direct sound from the speaker and sound from the speaker's back, reflected from the room corner. The interaction of the direct and reflected waves produces a resultant whose onset seems to be delayed by a few milliseconds. During the attack of the tone burst, as the direct and reflected waves add in a phasor fashion, a time-varying amplitude change results, as you can see in the upper left-hand corner of the wave envelope shown in **Figure 4**. Fourier decomposition of the resulting wave reveals an extraneous transient in the treble frequencies. While this transient may not be directly audible, its absence after proper acoustical treatment to reduce the reflected wave amplitude may well result in a

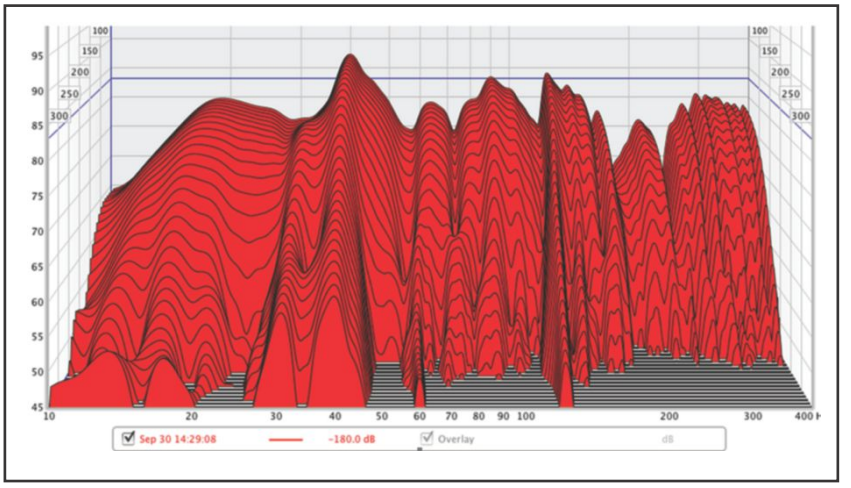


Figure 6: This waterfall plot shows the frequency and time response of the control room before the addition of TubeTraps. (Image courtesy of Art Noxon, Acoustic Sciences Corp.)

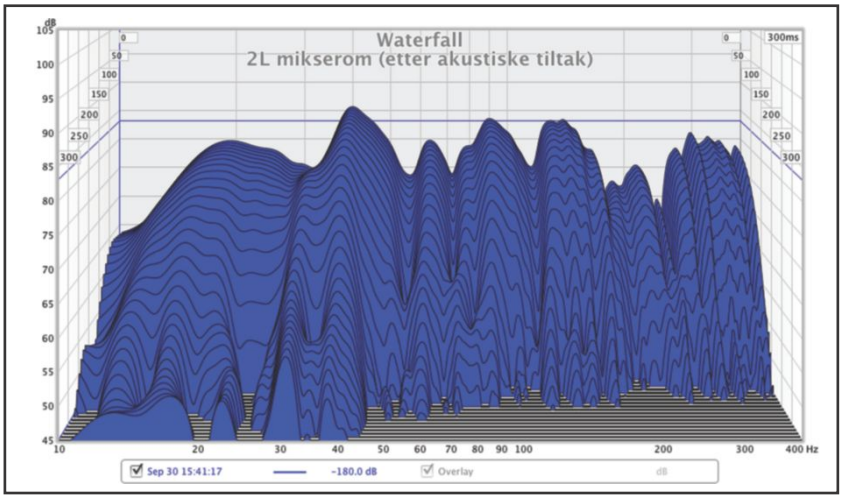


Figure 7: This waterfall plot shows the room response after a tuned 40-Hz resonator model TubeTrap was installed. (Courtesy of Art Noxon of Acoustic Sciences Corp.)

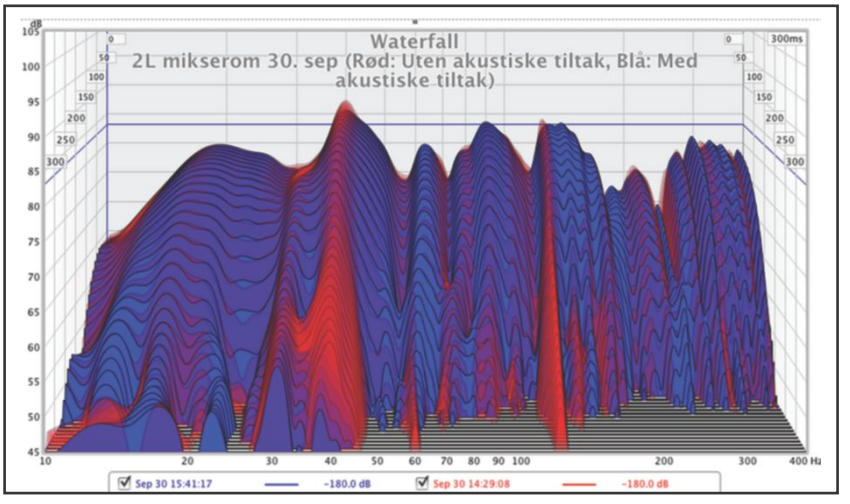


Figure 8: This composite waterfall shows the before (red) and after (blue) responses overlaid. (Courtesy of Art Noxon of Acoustic Sciences Corp.)



Photo 2: Here is one view of the studio after the addition of the TubeTraps. (Photo courtesy of Morten Lindberg)



Photo 3: This fisheye view of the treated studio, as seen from the control location, shows two stacked TubeTraps at left and two more at right. (Photo courtesy of Morten Lindberg)



Photo 4: This is the wall to the right-hand side of the control desk, with two 2-high stacks of TubeTraps visible. (Photo courtesy of Morten Lindberg)

noticeably cleaner sound. If bass traps are placed in the corners behind the speakers, the reflected wave is reduced in amplitude, and the sound at the onset of the burst has less coloration.

Bass Trap Effects

One example of a bass trap utilizing the Helmholtz principle as well as other absorption and reflection principles is the TubeTrap product line (see **Photo 1**), manufactured by Acoustic Sciences Corp. Typically, TubeTraps provide bass absorption between 55 Hz and 250 Hz, and can be custom designed and manufactured to eliminate specific frequencies in addition to the standard broadband bass absorption.

Measuring the in-room effects of bass traps is not as simple as it would seem. First, the normal test signals are pink noise or sine sweeps, and neither is like music in terms of transients. Second, adding bass traps only results in a 1-2 dB change in pink noise or swept-sine response measurements. However, the fine structure of the decay is cleaned up: the sharpness of variations in response is smoother.

One way to show this improvement is by the waterfall plots shown in **Figures 6–8**. These plots illustrate the improvement resulting from the addition of six 40 Hz TubeTraps and two large full bandwidth TubeTraps in the large, professionally built control room of Grammy Nominee Morten Lindberg of Oslo, Norway.

The response is illustrated using before and after waterfall plots. A waterfall plot is a graph of sound pressure level (SPL) (vertical axis) vs. frequency (horizontal axis), showing sequential time traces arranged on an "into-the-page" axis. **Figure 6** shows the response of the untreated room. **Figure 7** shows the room response after the installation of a tuned 40 Hz resonator model TubeTrap. **Figure 8** shows these two plots overlaid. Note the smoother response

in the 40-Hz region, and the elimination of ringing at 60 and 120 Hz, which had resulted from the lengthened decays caused by modes, stretching the decay at those frequencies.

The smoother response in the mid-bass after treatment results from the broadband nature of the TubeTrap tuned bass trap. It is also important to note the small change in SPL at 40 Hz, indicated by the roughly 2 dB difference between the red and blue early curves. This change by itself may have been undetectable. But, the improvement in transient response was indeed marked.

In the words of Morten Lindberg, of Lindberg Lyd, "[The] traps did exactly what they were supposed to, without drying up the overall response unnecessarily." The TubeTraps were supplied and set up by Alan Grindheim of Gjallarhorn Audio, TubeTrap distributor in Europe, who said: "The Resonator Traps eliminated the low tonal overhang that otherwise accompanied every LF [low-frequency] attack transient played in the room." These improvements result from the room's Q having been decreased by typically a factor of 4.

Photos 2–4 show the control room. The TubeTraps are the grey cylinders in the corners of the control room. (Note the red arrows shown in **Photo 2**.) The wooden half-cylinder vertical diffusers are split tree trunks.

Author's Note: I would like to thank Art Noxon, founder of Acoustic Sciences, and Morten Lindberg, of Lindberg Lyd AS, for their help in preparing this article.

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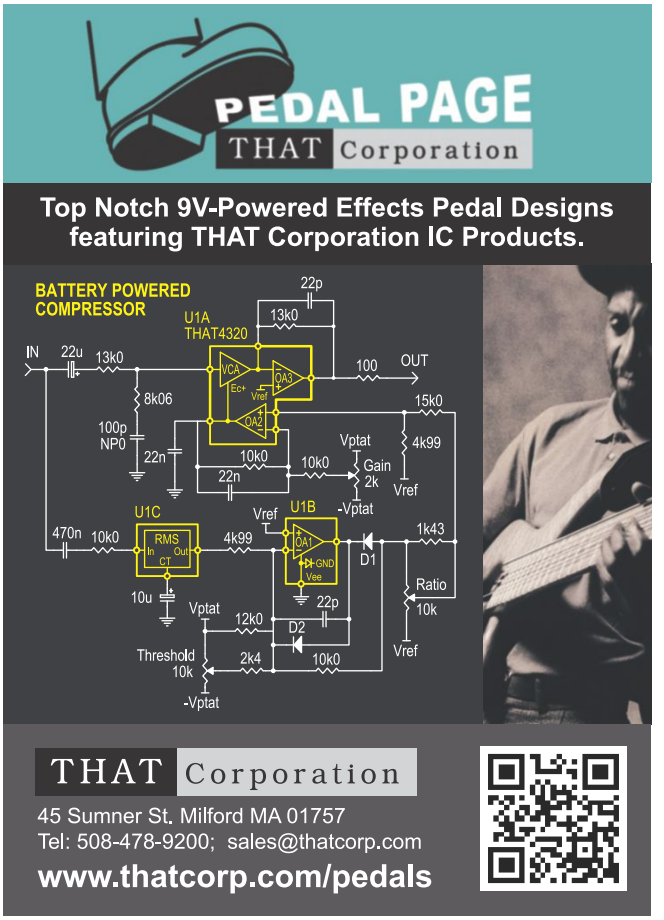
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CEntrance DACportable High-Resolution USB DAC and Headphone Amplifier

Pushing the Envelope in Personal Audio



CEntrance's self-powered wideband Class A DACportable is smaller than an average playing card.

Headquartered near Chicago, IL, CEntrance is known for its co-development and technologies licensed to many pro-audio and high-end consumer companies. In 2009, CEntrance introduced a portable product called DACport, a small, USB-powered digital-to-analog converter plus headphone amplifier. In 2015, it added DSD playback and sampling rates up to 384 kHz on the ultra-slim DACport HD. Recently, CEntrance added a rechargeable battery to its new DACportable, featuring the AK4490 D/A chip coupled with a Class-A headphone amplifier with digital volume control, which *audioXpress* received for review.

By
Oliver A. Masciarotte
(United States)

These days, my commute consists of a 45-minute drive to the western suburbs of Minneapolis, all the while listening to TIDAL or podcasts via Bluetooth from my phone. For many folks, their phone is their main source of entertainment, with streaming and local files providing the content. However, most phones are anything but high fidelity, and some manufacturers have seized the opportunity to address that shortcoming. One approach is an "audiophile" phone, which is not that crazy since many DAP portables are basically smartphones without the cellular radio.

A few manufacturers have taken a more purist approach. CEntrance, long known for its high-quality, reasonably priced electronics, have created the DACportable, a digital-to-analog converter that also gives you a wideband, low total harmonic distortion (THD) Class-A headphone amplifier. Thanks to its built–in power, it won't contribute to draining your phone or tablet either. Moreover, the battery power provides ultra-low noise, removing contamination from its asynchronous USB power rail.

The Technology

The Class-A circuitry is DC coupled, and is powered by a bipolar, ±9 VDC supply. This ensures excellent headroom for a consumer product. The direct coupling exposes any DC offset in the source file, though offset is rare these days for two reasons. Modern ADCs have eliminated the problem at the source, and DAWs (when properly configured) reveal any offset during post–production where it can be removed. The advantage of DC coupling is solid low-end amplitude and phase response all the way down to 0 Hz.

As with all proper DAC designs, CEntrance guarantees a bit-perfect audio stream. Michael Goodman, the man responsible for all things CEntrance, told me the company is particularly proud of its USB implementation, which it has "licensed widely in both the pro and CE audio industries." By widely, he means more than 48 audio firms, from consumer electronics stalwarts PS Audio, Bel Canto, Playback Designs, and Accuphase, to a range of pro and MI giants including Ernie Ball, Tascam, Crane Song and Mackie. Since CEntrance is an official third-party design house for XMOS, DACportable employs one of its microcontrollers.

Goodman mentioned that, "As a consulting company, we've enabled dozens of audio brands with USB technology. It was natural for us to choose XMOS for this design. We make our own Windows drivers and write our own code for XMOS chips." If you were to compare the audio bit stream entering the DACportable's USB receiver, from the audio playback application to the bit stream right at the DAC inside the device, you would see the exact same data, with nothing altered or lost.

The product is assembled in the US, is RoHS compliant, and supports all the modern rates, at 24 fixed or 32 floats: 44.1 to 384 kHz, along with baseband DSD and DSD2 (5.6448 or 6.144 Mbps). On one side of the unit is a 20 dB gain switch to better match sensitivity between low impedance in-ear monitors (IEMs) and 600 Ω studio cans. The product consists of a 98 mm x 41 mm x 21 mm black metal rectangle, with one rounded corner that contains a small, continuous volume wheel.

When asked if the variable gain logarithmic control is analog or digital, Goodman recounted his journey with gain control implementations. "Our older products, such as the venerable DACport Classic, used analog gain potentiometers for volume adjustment. While that's the most straightforward method, the pot eventually becomes dirty and starts to develop an unwanted, scratchy sound. This would happen after several years of use, but we still felt that it wasn't a good customer experience. So, when developing DACportable, we made a decision to switch over to a digitally controlled potentiometer. The thumb wheel pot in the DACportable doesn't pass audio. Instead, it creates a control signal for the 32 bit microprocessor, which in turn sends commands to adjust the volume in the digital domain. We use…256 steps and zero crossing gain adjustment to make sure that the volume control is smooth and linear. It actually feels better than a regular pot. And the benefits of digital control are clear—your right and left (channel gain) always remains perfectly balanced, even down to the lowest volume settings, which is not possible with analog



Michael Goodman, founder of CEntrance, can be often found managing the company's booth at trade shows.



In addition to DXD and DSD2 support, the DACportable has two gain settings to match cans of any sensitivity.



The DACportable is made to travel, with recessed controls and a size that mates well with modern smartphones and portable players.



The DACportable is shown with its affordable yet capable little brother, the DACport HD, and a worthy opponent, Sony's PHA-3.

pots, and control is not subject to (wiper) noise, which are inherent with mechanical pots."

Along with the 20 dB gain switch are two EQ switches. All three are recessed to avoid accidental changes during transport or use. The EQ, a two-position flat vs. 3 dB of boost, is voiced, according to Goodman "…to be both musical and subtle. This is not a typical boom box 'smile curve.' The LF EQ option features a specially developed, custom boost curve to add punch without mud. The HF EQ is designed "…to add just the right amount of 'sheen' at the top range of the audio spectrum, to accentuate the airy qualities of strings and wind instruments."

DACportable shares a lot of technology with its less expensive new little brother, the DACport HD. Both products feature the same sample rate support and high output drive capabilities. Actually, the DACport HD was the first to offer the new "Amp Extreme," a headphone amplifier design with the highest output power, 775 mW into 32 Ω, and the lowest output impedance yet (1 Ω). Compared to the DACport HD, DACportable adds an eight-hour, rechargeable battery and the aforementioned two-band EQ from CEntrance's award-winning HiFi-M8. As we all know, power is the Achilles heel of mobile computing. Hence, the built-in battery, along with welcome compatibility with iPhone and Android.

The rechargeable battery uses lithium-polymer technology, the newest formulation with the highest energy density available for CEntrance products. The battery is sized at 3,000 mAh and, according to Goodman, it will confidently give you eight hours of play time. Using my Audeze EL-8 and LCD-3 cans and playing at what I consider reasonable levels, I routinely got 10 hours.

"We give our customers versatility in terms of charging. The unit can be charged from a USB port or a small 0.5 A iPhone charger cube, but also from the larger, iPad-style 2.5 A charger. With the latter, the charging time can be as quick as four hours and you are ready to roll again," said Goodman.

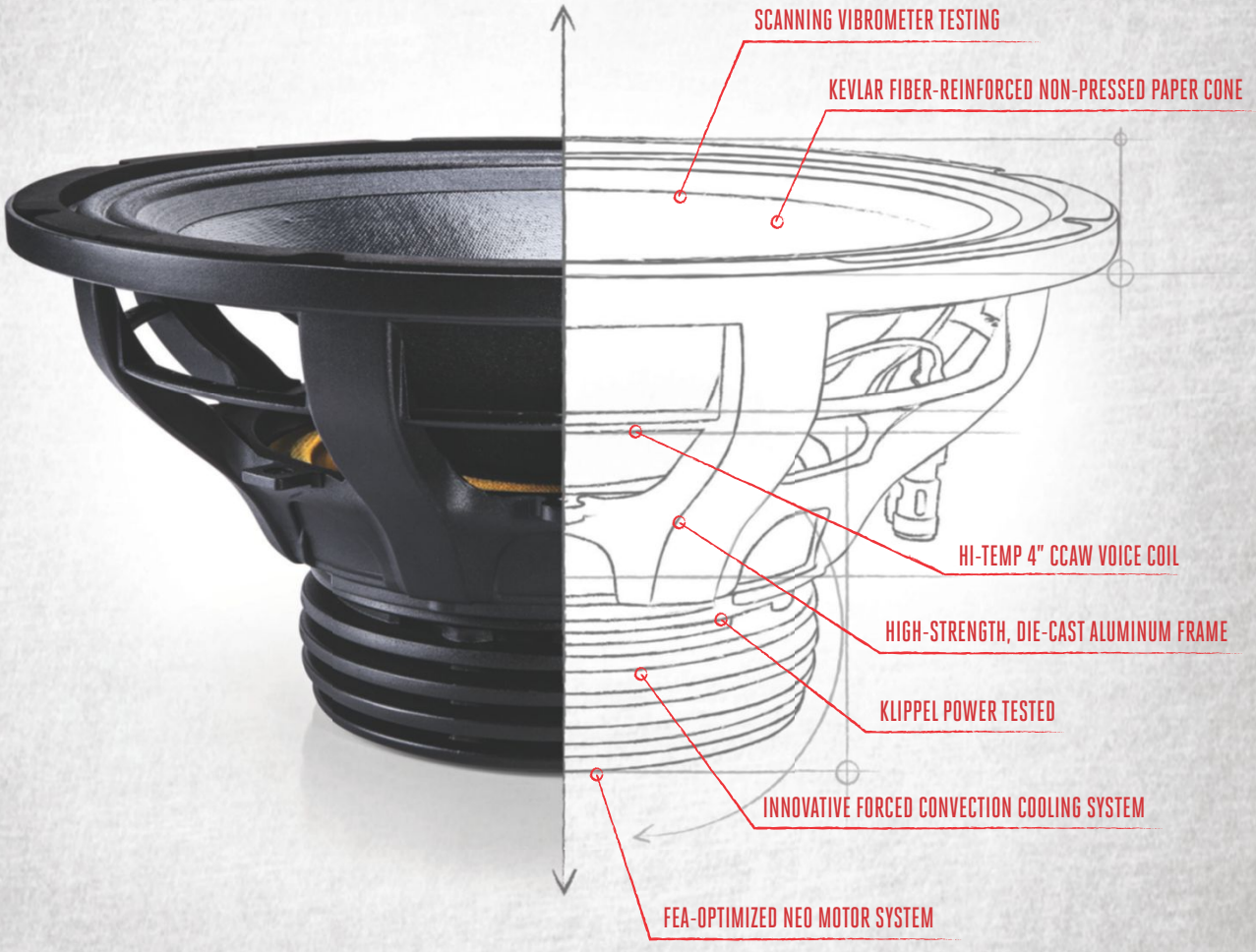
The Design

The design brief for the DACportable, along with new challenges the designs team came up against, can be summarized as miniaturization. A surface-mount assembly, with a double-sided PCB, made the overall package smaller. The PCB itself features four layers of copper for extra grounding and shielding, and is passivated with gold plating to reduce corrosion and accommodate fine pitch components, such as the microprocessor that comes in a ball grid array BGA package for higher density, better heat conduction, and lower interconnect inductance. Goodman told me it was a process of refinement.

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What about MQA?

Whenever the topic of DACs comes up these day, it seems a certain topic will soon follow and that's Master Quality Authenticated (MQA). To be precise, does/will this or that DAC support MQA decoding? Prompted by TIDAL's recent announcement that MQA streaming support is now active and curious about DACportable's future upgradability, I posed that question to CEntrance Managing Director Michael Goodman, who said "We believe that MQA is an exciting new development in the audio industry and are watching market adoption. As an engineering firm, we don't see hurdles to the technical implementation. As soon as the market momentum is there, we will be able to support it."

"We already had a lot of experience with battery charging technology, D/A conversion, amplification, and USB interfacing so those parts just needed to be made much smaller in order to fit a device that can easily (fit comfortably) inside your pocket or a laptop bag. Making everything fit on the tight PCB required multiple rounds of prototyping, tweaking and adjustments for best usability and absolute pristine sound. That took months of work and now we are shipping what is revision 5.1 of the PCB," explains Goodman.

As with all current microcontroller-based products, there's a lot of software under the hood for the best user experience. The latest firmware adds a timed power down feature, which turns the device off after several minutes of inactivity. This is convenient if you've forgotten to turn it off. CEntrance implemented the feature after experiencing a dead prototype when it was most needed. The unit waits patiently and, if nothing is playing, it will turn itself off to preserve charge. There is nothing more frustrating than a product that expires overnight because you forgot to power it down.

High performance requires very clean power, and batteries can provide that when properly managed. When I asked how many regulators are hiding inside the DACportable, I was surprised to learn there are five: one for the rapid battery charger, another for the 3.3 VDC digital supply, two more for the bipolar analog supply mentioned earlier, and a separate ultra-clean 5 VDC supply for the DAC itself.

Goodman stressed that, "All supplies are custom, of course." Those five don't include a special battery gauge chip that measures battery voltage with "single percent precision." This IC silently optimizes battery life as the battery ages over years of use, again to provide the best user experience. All of the power supplies are sequenced by the microprocessor, so only the subsections needed are powered up and in the sequence that optimizes performance.

In addition to the classy Class-A analog path and careful power massaging, another contributor to the outstanding sound is the clocking. Two custom, mil-spec clocks provide much tighter temporal tolerances, along with a wider immunity to temperature variation. The DACportable exhibits 1 ps of jitter, an order of magnitude below the threshold of golden-eared audibility.

Manufacturer's Specifications

Digital Specs
Resolution: 32 bit (Also supports 24 bit)
Sample Rate: 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.2 kHz, 192 kHz, 352.4 kHz, and 384 kHz
DSD Support: DSD64 and DSD128
Clocking: Two custom, mil-spec clocks, 1 ppm precision, 1 ps jitter
Compatibility: Mac, Windows, Linux, and Android, iOS.

Analog Specs
Output Impedance: 1 Ω
Output power, 32 Ω: 775 mW (total)
Supported Headphones: 16 to 600 Ω
Maximum Output Level: 1 kHz: 4.1 VRMS/ +14.5 dBu

Audio Output
Gold-plated 3.5 mm headphone jack
AC Adapter: Standard 5 V adapter with USB "micro B" plug
DC Powering/Charging: +5 V (2.1 A) from external adapter, such as iPad charger
Internal supplies: Battery-isolated, triple-filtered ±9 V, analog power supply rails
Battery Specs: Rechargeable Li-Polymer, 8 hours play time
Chassis Material: Scratch-resistant, lightweight anodized aluminum

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centrance.com

Listening

First, I had to match the gain between the DACportable and the other two micro DACs I had on hand: the admirable and functionally similar Sony PHA-3 and the slightly elderly Geek Out 1000 (GO1K) from LH Labs. Both the PHA-3 and the DACportable have external manual gain controls, while the Geek Out lacks a gain control, taking its gain setting from the host. So, I set hardware gain controls at maximum, and played a brown noise track out of TIDAL to build a table of gain compensations for each device. The GO1K had the least gain, the PHA-3 had the most, while the DACportable was in between. All of them could easily drive any of my cans, from my closed back work headphones, the 32 Ω AIAIAI TMA-2-S01, to my high sensitivity, 25 Ω Westone W60 reference universal IEMs. For this evaluation, I settled on my open-back circumaurals, the 110 Ω Audeze LCD-3.

Because the DACportable has an unbalanced output, I couldn't use any of my balanced wire sets, which I grab for both my Pono Player and my Sony

PHA-3, so I started with my Audeze LCD-3's factory cable. On power up, the DACportable emitted a brief, low amplitude spit of pinkish noise. Since I tend to use highly adjustable software EQ, initially I kept the hardware's EQ disabled.

The PHA-3 was my reference portable DAC/HPA until the DACportable showed up. Both models are subjectively flat, uncolored, and quiet. That said, in most all regards, the DACportable exhibited better low amplitude detail—think room ambiance and reverb tails—along with extended presence and air that plucked strings (e.g., acoustic guitars and harpsichords) showcase. Low-frequency grunt and weight are delivered in spades with the DACportable. Kick drums simultaneously have more impact, while the snap of the beater is cleaner than what the PHA-3 offers. The soundstage is wider and more detailed with the CEntrance critter. One aspect that I enjoyed with the PHA-3 was its ever so slight mid–forwardness. As I said, the PHA-3 reads "flat," but its slightly darker voice brings midrange more into the spotlight. For modern, aggressively colored productions (especially popular music and dance) that is not a bad thing as the hard edges are smoothed a bit.

As I mentioned, the original GO1K has been superseded by the Geek Out V2. Since I had only the original on hand, I thought I'd try it. Back in the day, the GO1K summarily trounced the original Dragonfly, in every sound quality metric you'd care to consider. In its time, it was no slouch. The GO1K has two anti–image filter choices. For these tests, I chose its "Time Coherence Mode," which uses a minimum phase digital filter and time optimization algorithm that removes all pre-ring(ing) from the original signal and realigns the impulse response. Despite that, the elder DAC/HPA of my three came in a distant third in sound quality.

For less than $350, the DACportable sets a new standard in portability, versatility, and convenience. Yes, there are less expensive microDACs available but, with their inferior fidelity and lack of support for all modern sample rates, why bother? In the end, it maintains the solid reputation that's made CEntrance a leader in digital audio, and they're still on course.

Goodman summed it up with, "We wanted to make the smallest, most powerful, most usable portable DAC/Amp that would work with the widest array of sources, from computers to portable devices, and I think we have achieved this goal. We have famous producers and mastering engineers using DACportable for work, and it's nice to know that music is being made with it, not just enjoyed."

To complement this review Stuart Yaniger performed a series of measurement tests on the CEntrance DACportable to determine if the device is as good as we think it is.

About the Author

Oliver A. Masciarotte has spent more 30 years immersed in the tech space, working on facilitation, optimization, marketing, and product development for clients worldwide. As an author and speaker, he enjoys informing folks about technological best practices. More information is available at seneschal.net and othermunday.com.

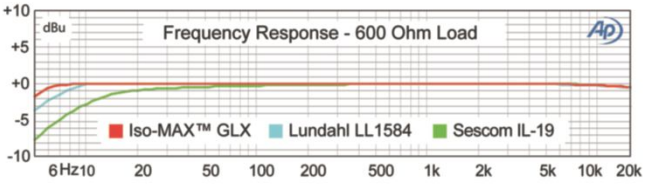
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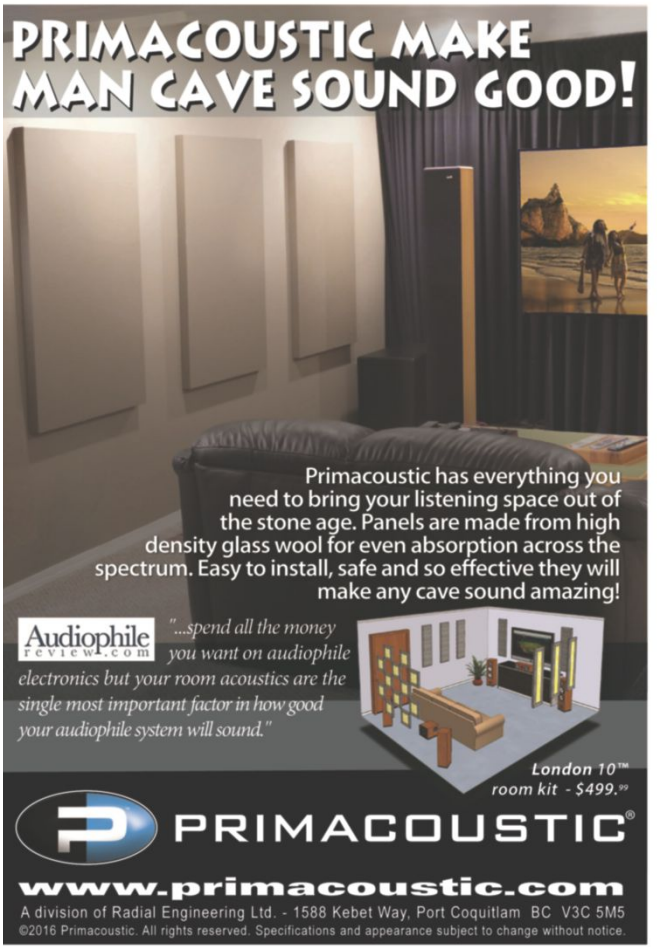
Ground loop isolator comparison	Iso-Max GLX	Lundahl LL1584	Sescom IL-19
Freq Response	10Hz ~ 30kHz	10Hz ~ 30kHz	40Hz ~ 30kHz
Distortion @ 20Hz:	.001%	.01%	.1%
Distortion @ +4dB:	.005%	.06%	.3%
Phase shift @ 20Hz:	0°	0°	20°
Ground lift switch:	Yes	No	No
Construction:	Metal	Metal	Plastic
Warranty:	3 Years	1 Year	1 year
MAP Price (estimated street)	$69	$130	$45



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CEntrance DACportable Measurements

By
Stuart Yaniger
(United States)

For the tests, I used an Audio Precision APx515 analyzer connected to an HP DC5700s desktop computer running Windows 7 SP1. The dummy loads for the headphone amplifier were made from 2 W non-inductive carbon composition resistors.



Photo 1: The DACportable is compared to the iPhone 6 Plus.

Trials and Tribulations

To begin, let me say that the measurement phase of this review was a bit more of an adventure than I expected. The driver download indicated as "released" on the website and compatible with Windows 7 would not allow the DACportable to be correctly installed. After some correspondence with the helpful folks at CEntrance, I instead downloaded and installed the beta release driver indicated as compatible for Windows 10; this enabled the DACportable to be properly installed.

The first unit I received gave inconsistent and often very poor results when tested, and failed after a few days. CEntrance promptly replaced it with a second unit: All of the data shown here were acquired from this second unit (see **Photo 1** and **Photo 2**).

The driver, called Universal Driver (UD), is quite picky about its playmates. When the DACportable is selected by the Audio Precision APx515 analyzer in ASIO mode, the Audio Precision software crashed spectacularly—something it did not do with any other DAC/driver combination I have used. This meant that the Audio Precision software could not be used as an ASIO source for the DACportable/UD ASIO combination. I tried using ARTA's function generator as a source, and it also immediately crashed. When my 24 bit .wav files were played from foobar, they would only run as 16 bit. And using an M-Audio Delta 192 S/PDIF input with the DACportable output caused spectacular glitching.

As a workaround, I installed the DACportable on a second computer (a Lenovo T520 laptop with Windows 7 64-bit), then I used ARTA to generate .wav files of test signals on the laptop. Next, I played the test signals using J River Media center, which had no problem with the CEntrance ASIO driver. The output of the DACportable was then connected to the APx515's analog inputs, with the APx515 connected to the Hewlett-Packard lab computer. This was a clumsy setup (three pieces of software and two computers), but it worked—other than an inability to do frequency or voltage sweep measurements controlled by the APx515 or with ARTA. The moral here is: You may want to check the compatibility of your software with the UD. The folks at CEntrance have promised to take a look at this, and if we can resolve it, I'll post an update and sweep measurements (distortion vs. frequency, distortion vs. level, and crosstalk vs. frequency) on the audioXpress website.

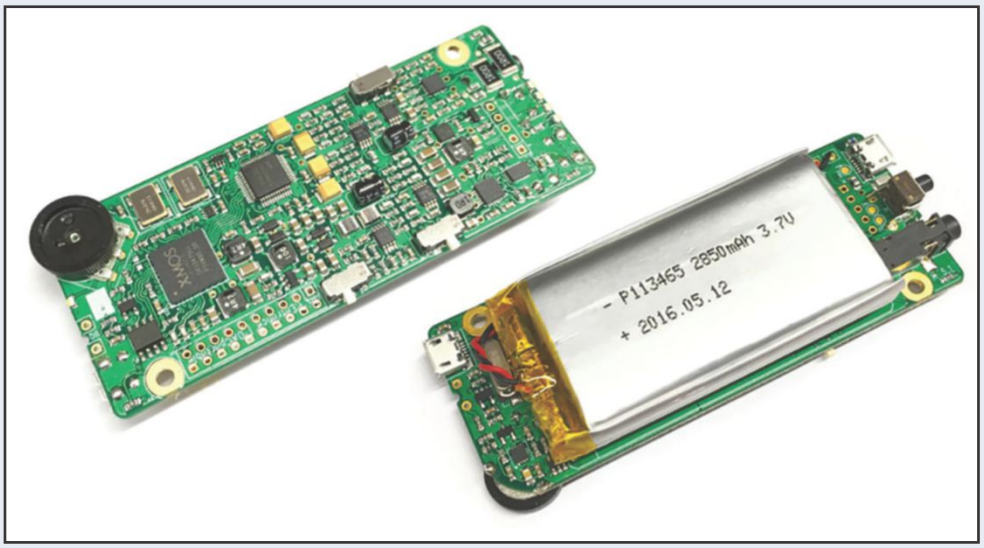


Photo 2: Take a look inside the DACportable.

Measurements

Moving on to the measurements, the DACportable has two gain settings (HI

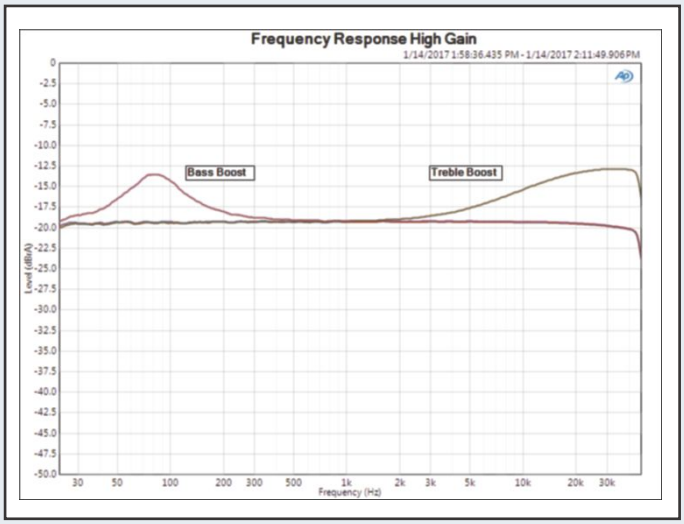


Figure 1: The frequency response of the DACportable can be varied from flat to treble boost to bass boost, depending on the switch settings.

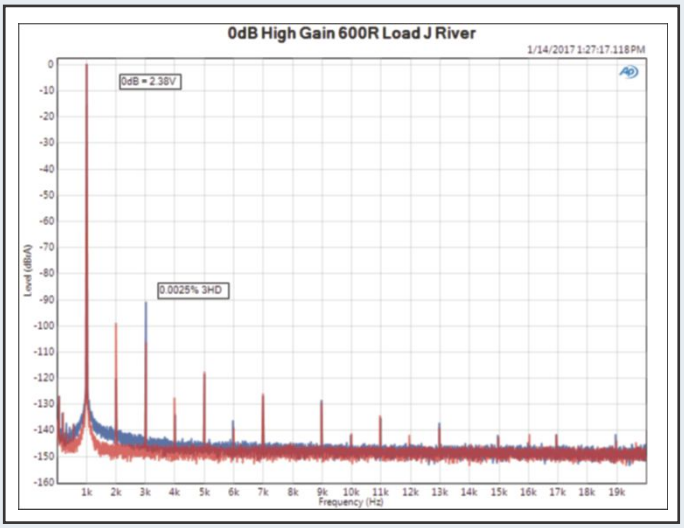


Figure 2: The spectrum for a full scale 1 kHz sine wave with a 600 Ω load (2.38 V out) shows very low distortion.

and LO). With a full scale input (0 dB) 1 kHz sine wave, maximum output into either a 600 Ω (600R) or 32 Ω (32R) load was 2.38 VRMS and 0.271 V, respectively. The CEntrance website gives 4.1 V as the maximum output specification, which is not met here. Nonetheless, the output is more than sufficient to drive the vast majority of headphones to ear-shattering levels.

Source impedance at 1 kHz was 1.1 Ω, with a rise to 1.3 Ω at 10 kHz. This is sufficiently low that the frequency response with a headphone load will not significantly deviate from flat. Crosstalk at both 1 kHz and 10 kHz measured -56 dB. This is likely limited by the common return lead used for three wire (L, R, common) headphones rather than stray capacitive coupling, since the latter would show increasing crosstalk at higher frequencies.

The DACportable also has two tone controls comprising bass and treble boost switches. **Figure 1** shows the frequency responses with the boosts turned off and with the bass and treble boosts turned on, using a 600 Ω load and HI gain. These responses were acquired using white noise and power averaging at 96 kHz sample rate and 24 bit samples. As can be seen, the frequency response is quite flat with a slight roll-off at the frequency extremes, and the tone boosts give a very moderate (6 dB) enhancement. Changing the load to 32 Ω or the gain to LO have no effect on the frequency response.

Figure 2 shows the distortion spectrum at 1 kHz and full output with a 600 Ω load. One channel is significantly worse than the other for third and fifth harmonics (indicating a symmetrical nonlinearity), but still clocks in at under 0.003% distortion. The spreading of the spectral skirts is also a little worse for that channel, but the artifacts are all under -130 dB from full scale, which is something smaller than negligible.

When the DACportable was loaded with 32 Ω, the distortion increased a bit (see **Figure 3**), but the worst harmonic (the third) is still 75 dB down from the fundamental (under 0.02%), an acceptable

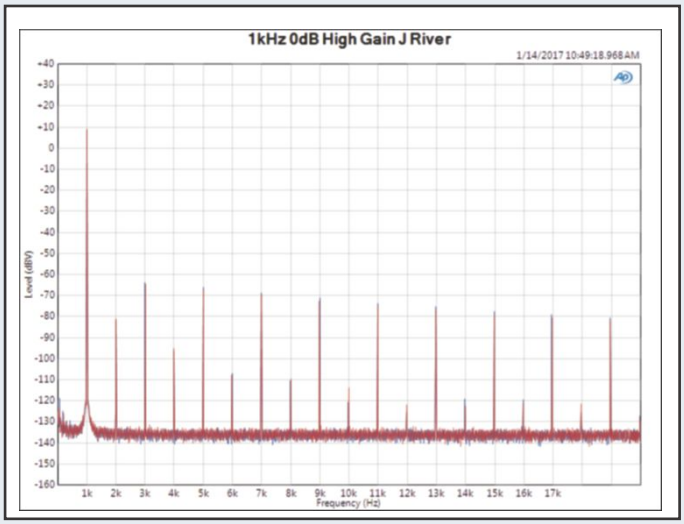


Figure 3: The spectrum for a full scale 1 kHz sine wave with a 32 Ω load (2.38 V out) shows increased distortion, mostly odd-order.

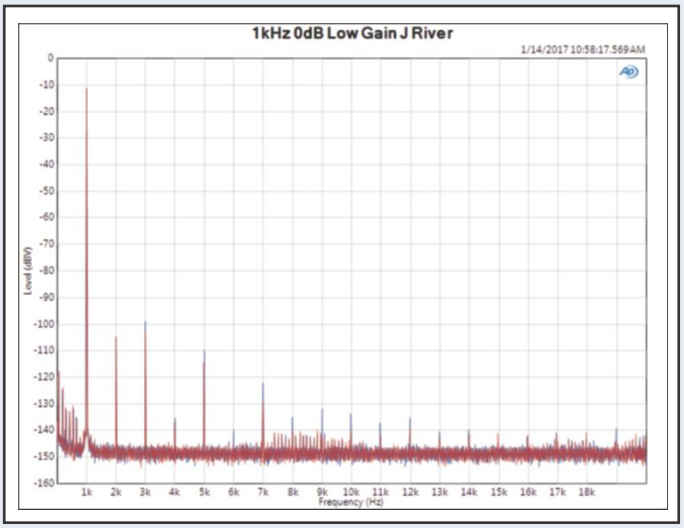


Figure 4: Reducing the DACportable output to 0.271 V (32 Ω load) significantly reduces distortion.

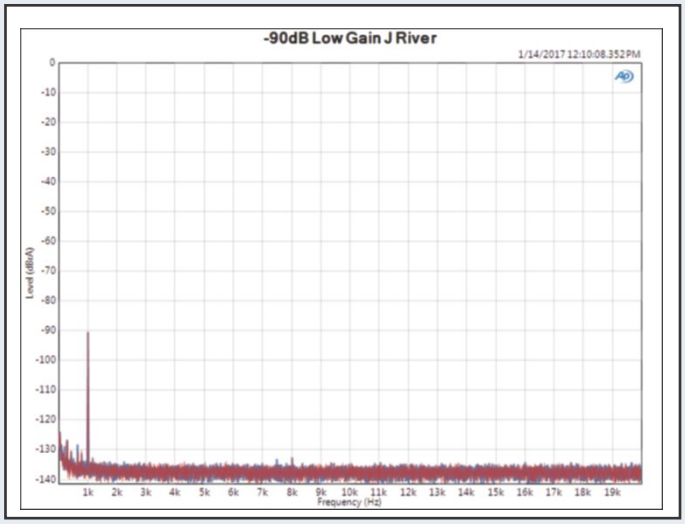


Figure 5: The spectrum for a sine wave at -90 dB appears perfectly clean, confirming excellent low-level linearity.

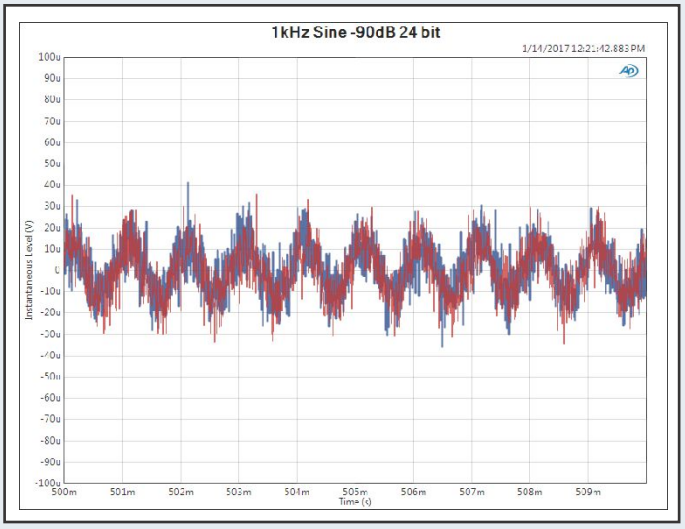


Figure 6: The waveform of a 1 kHz sine wave at -90 dB shows no indication of low-level errors or non-linearity.

result. **Figure 4** shows the distortion spectrum for the same full scale signal and 32 Ω load, but with the DACportable in LO gain mode. The distortion is significantly lower than with the DACportable in HI gain mode (0.004% for the third harmonic), suggesting that the distortion is dominated by the analog output stages.

Low-level performance was measured using a -90 dB 1 kHz sine wave, the spectrum of which is shown in **Figure 5** for the DACportable set to LO gain and loaded with 32 Ω. This is perfectly clean, with an impressively low noise floor. This excellent result is reflected in the oscilloscope plot of the output, shown in **Figure 6**. You can see the sine wave overlaid with noise, with no signs of low level nonlinearity or crossover distortion.

Figure 7 shows the intermodulation distortion (IM) spectra of the DACportable using a 1:1 19/20 kHz dual sine wave for the DACportable set to LO gain and full output with a 32 Ω load. Total IM was about 0.02%, again a very good number for this load. Increasing the load to 600 Ω and using the HI gain setting yielded the spectrum shown in **Figure 8**, which indicates IM to be less than 0.002% in the worse of the two channels at full output, an excellent result.

For multi-tone measurements, I used a test signal which is a series of tones spaced one-half octave apart. Misbehavior of electronics or excessive jitter will cause spurious tones to be created between the tones. **Figure 9** shows the DACportable's output spectrum at HI gain and full output into a 600 Ω load. It is impeccable, showing no spurious signals above -135 dBV. Dropping the load to 32 Ω, the results look a bit messier at full output (see **Figure 10**), but all the spurious signal lies at -110 dB and below so is unlikely to be of any audible consequence.

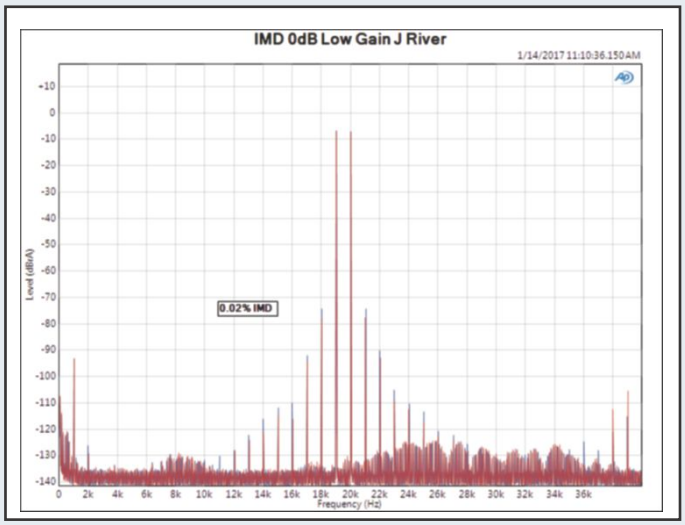


Figure 7: The DACportable's 19/20 kHz IM distortion is quite low (0.02%) even at full output and a 32 Ω load.

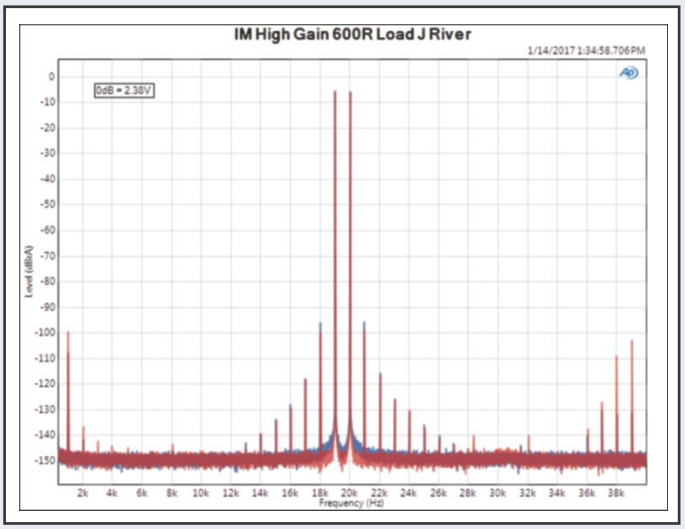


Figure 8: Increasing the load to 600 Ω reduces the full-scale IM distortion to 0.002%.

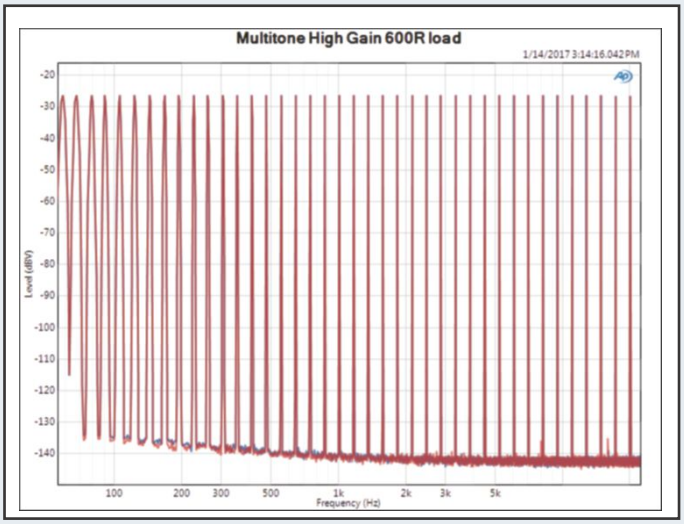


Figure 9: A multi-tone test with half-octave spacing at full output and a 600 Ω load shows a low noise floor and an almost complete absence of spuriae.

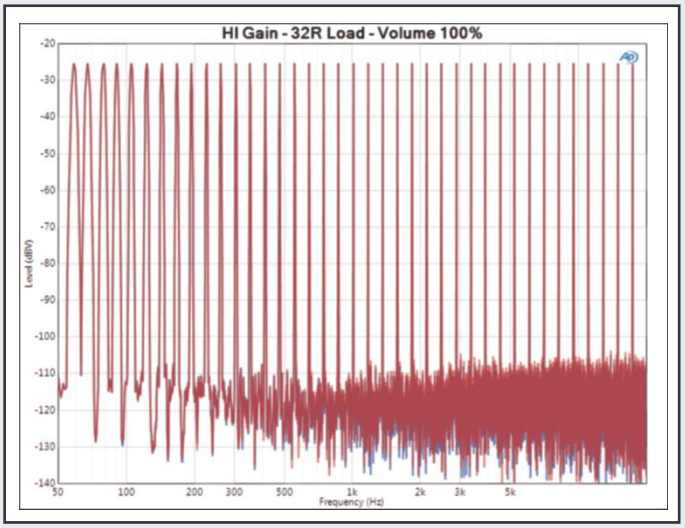


Figure 10: Changing the load to 32 Ω increases the distortion and the noise of the multi-tone signal spectrum, though they remain relatively low.

Square waves are diagnostic of the anti-imaging filters used in a DAC. **Figure 11** shows the DACportable's output for a 1 kHz square wave at a 96 kHz sample rate. The leading-edge ringing is typical of the Gibbs phenomenon seen with brick-wall filters. The square wave is otherwise clean and unremarkable. Likewise, **Figure 12** shows an undithered square wave at a -90 dB level, and like the -90 dB sine wave shown previously, is clean and undistorted, but overlaid with the small amount of noise from the analog circuits. This is a fine result.

Conclusions

The DACportable turned in a first-rate set of measurements, with nothing to indicate anything less than complete audible transparency. If used with compatible software, its performance leaves little to be desired, especially with higher impedance headphones or headphones with normal efficiency. Oliver Masciarotte's good impressions discussed in his article, "CEntrance DACportable High-Resolution USB DAC and Headphone Amplifier," are borne out by the measured performance. ax

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.

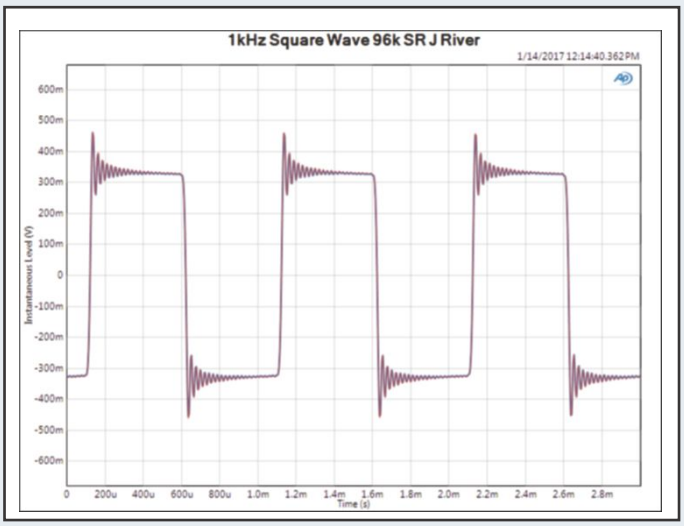


Figure 11: A full-scale square wave shows ringing on the leading edge, a characteristic of brick-wall filters.

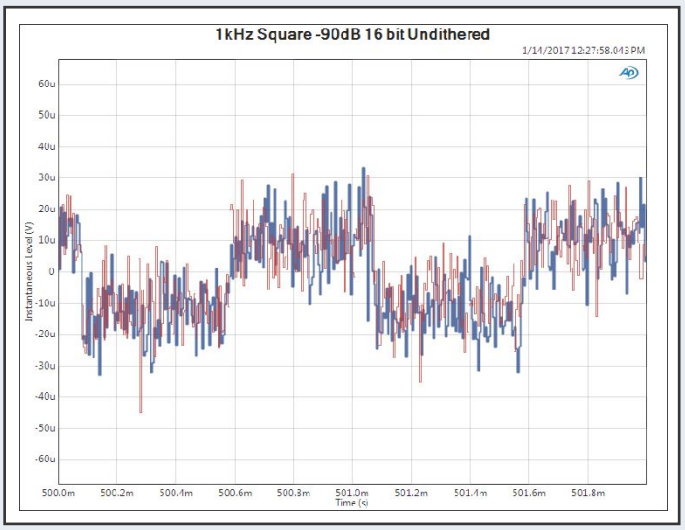


Figure 12: A square wave at -90 dB demonstrates the DACportable's low-level linearity.

The Hafler Revival

A Review of Hafler's PH50 and PH60 Phono Preamps and the PH44 Moving Coil Step-Up Transformer

To understand more about the Hafler revival, we delve into some of Hafler's history and the companies associated with it, including Dynaco, Radial Engineering, and Jensen Transformers. In this review we examine a few of its most recent phono offerings.



Photo 1: Front view of the Hafler PH50 phono preamp. The preamp is designed for high-output magnetic cartridges and has 35 dB of gain. The switchable low-cut filter features a 12 dB/octave rolloff that is continuously variable from approximately 20 Hz to 500 Hz.

By
Gary Galo
(United States)

Radial Engineering, Ltd., is a professional audio company based in Vancouver, BC, Canada. Its products are sold under several brand names, including Radial, Tonebone, Primacoustic, Reamp, and Zebracase.

Radial was conceived by Peter Janis and Domenico Pingitore in 1992, and they began as Canadian importers for several well-known audio brands, including Mogami cable, Jensen Transformers, and Sonex acoustical products. After Pingitore left Radial, Janis decided to get into the manufacturing business, introducing Radial's first direct box in 1996. Radial's history can be found on its website (www.radialeng.com/history.php).

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Hafler PH50 Phono Preamp: $449.99
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Jensen Transformers was founded by Deane Jensen in 1974. Jensen recognized that the mediocre sound quality of many mixing consoles and other audio products was due to badly designed transformers that did not work well with active circuitry. His innovations in transformer design included double-Faraday shielding for superior noise rejection and improved quality-control testing of core materials.

Jensen soon established his company as one of the leading manufacturers of high-performance audio transformers. After his untimely death in 1989, the company was taken over by his associate and friend Bill Whitlock. Technical excellence and innovation continued under Whitlock's masterful guidance, maintaining Jensen's preeminence in the transformer field to this day.

Radial has been closely associated with Jensen since the early 1990s. The company designed many products around Jensen transformers, and eventually became Jensen's biggest customer. Upon Whitlock's retirement in 2014 at age 70, Jensen Transformers became a division of Radial Engineering. Fortunately, Jensen has continued to benefit from Whitlock's expertise, retaining him as a technical consultant.

Hafler History

David Hafler founded The Dyna Company in 1955 with business partner Ed Laurent, offering a 50 W mono power amplifier—the Mark II—as their first product. As with all subsequent Dynaco products the Mark II was available as a kit or factory-assembled. This amplifier was followed in 1956 by the Mark III and in 1959, with the arrival of the stereo LP—the legendary Stereo 70. Dynaco sold more than 300,000 Stereo 70 power amplifiers over the life of that product. Other highly regarded tube products included the PAS-2 and the PAS-3X stereo preamps, and the FM-3 tuner. Dynaco launched its first solid-state products in 1966—the PAT-4 stereo preamp and Stereo 120 power amplifier. Pared-down versions of those two products formed the basis for the SCA-80 integrated amplifier. Other Dynaco solid-state offerings included the Stereo 150, the 400, and the 416 power amps, the PAT-5 and the PAT5 BiFET preamps, and the FM-5 tuner.

Dynaco introduced the Dynaquad four-channel adapter in the early 1970s, which simulated ambiance information in a pair of rear loudspeakers, by simply inserting a 10 Ω resistor between the front and rear-channel grounds. No additional amplification was needed, and Dynaco incorporated this pseudo-quad system into the SCA-80. Dynaco's loudspeakers—the A25, the A35, and the A50—were designed and built in Denmark using SEAS drivers. The loudspeakers were remarkable for their natural, low-coloration sound. Many critical listeners found the A-25 to be equal or superior to Acoustic Research's considerably more expensive AR-3a, which at the time was considered a benchmark among home high-fidelity loudspeakers.

Hafler sold Dynaco to Tyco, Inc., in 1966, but he continued to work with the company until 1974. After a few years with Ortofon, he founded The David Hafler Company in 1977, returning to the kit business with the DH-101 stereo preamplifier. This was followed by the DH-200 and the DH-500 MOSFET power amplifiers, and the DH-110 preamp. As with their Dynaco ancestors, these reliable and cost-effective products were enormously popular with kit builders, and inspired several modification articles in *audioXpress*' predecessor, *The Audio Amateur*.

David Hafler retired and sold his company in 1987 to Rockford Corp. Under Rockford's management, the Hafler brand moved toward the pro-audio market rather than consumer products. Radial began its connection with Hafler as the Canadian distributor for Hafler amplifiers, ultimately purchasing the company in 2014. Radial now plans to reintroduce some of the classic Hafler and Dynaco amplifiers. Due on the market shortly is a completely updated tube amplifier based on the Dynaco Stereo 70. Information on all new Hafler and Dynaco products can be found on the Hafler website (www.hafler.com).



Photo 2: The rear panel of the PH50 phono preamp has gold-plated PC-mount RCA jacks that are used for input and output connections. DC power is supplied by an external wall-wart transformer. The rear panel of the PH60 preamp is identical.

Hafler Phono Preamps

Since its acquisition by Radial, Hafler has introduced four power amplifiers (ranging from two to 12 channels), two headphone amplifiers (one tube, one solid state), two RIAA phono preamps, and two moving coil step-up transformers. The step-up devices incorporate Jensen transformers designed specifically for this purpose.

The PH50 phono preamp has 35 dB of gain and is designed for moving magnet cartridges with outputs in the 3-5 mV range (see **Photos 1–3**). It will also work fine with moving iron cartridges with similar output levels (e.g., the Grados). The PH60 has 70 dB of gain and was designed for low-output, moving coil cartridges (see **Photo 4** and **Photo 5**). Cartridges of this type generally have output levels well under 1 mV, with most falling in the 0.3 to 0.5 mV range.

Both preamps are based on the Texas Instruments (TI) LM833-N dual IC op-amp. The

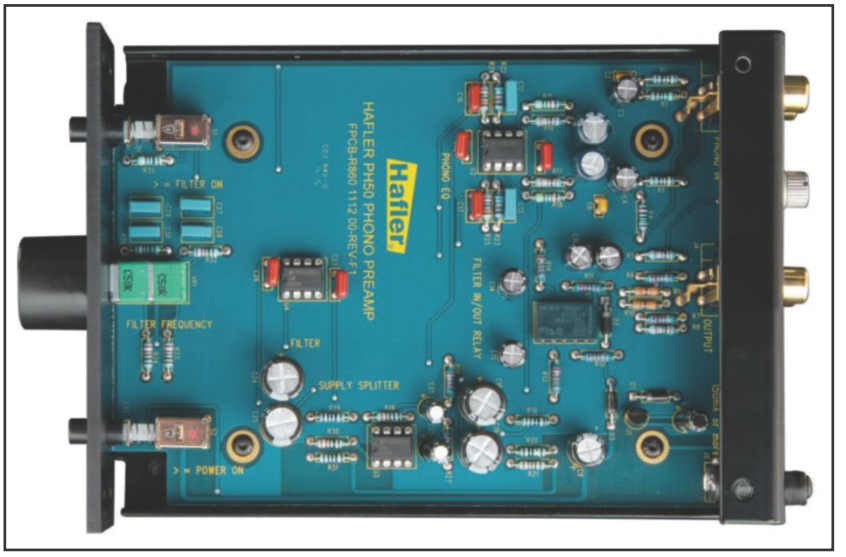


Photo 3: An inside view of the PH50 phono preamp shows three LM833-N dual op-amps, which are used for the feedback RIAA phono preamp (top right), the rumble filter (middle left), and the power supply polarity splitter (bottom center).

LM833-N features a 4.5 nV/√Hz low input noise voltage, a slew rate of 7 V/µS, and total harmonic distortion (THD) of 0.002%. This device has been designed with a special emphasis on audio applications and has the same noise, distortion, and slew rate specs as the older LM833. The main differences appear to be in the manufacturing process and the "typical" gain-bandwidth product, which is 16 MHz for the LM833 and 15 MHz for the LM833N (minimum for both is specified at 10 MHz). According to TI's datasheet for the LM833-N, "This dual amplifier IC utilizes new circuit and processing techniques to deliver low noise, high speed, and wide bandwidth without increasing external components or decreasing stability."

Page 14 of the TI LM833-N datasheet contains an RIAA phono preamp circuit. A look inside the PH50 reveals that Hafler has used the TI circuit in this preamp. The only differences are the values of the input and output coupling capacitors, which TI set at 33 µf and 10 µF, respectively. Hafler changed each to 22 µF. The circuit, as modified by Hafler, is shown in **Figure 1**.

The RIAA feedback circuit is the familiar "Network A" (Lipshitz nomenclature). Since the LM833-N is unity-gain stable, no amplifier stability resistor is needed in the feedback circuit. This also puts the high-frequency rise in response well above 20 kHz, so no output R/C low-pass filter is needed for correction. The 390 Ω resistor and the 100 µF capacitor set the low-frequency -3 dB point at about 4 Hz. The capacitor also ensures unity gain at DC, which will maximize headroom.



Photo 4: This is the front panel of the Hafler PH60 moving coil phono preamp, which features 70 dB of gain to accommodate even very-low-output cartridges. The switchable rumble filter has a 6 dB/octave rolloff with a -3 dB point of 90 Hz.



Photo 5: An inside view of the PH60 moving coil phono preamp shows eight 2N4403 transistors that are used in parallel to provide the gain required for the low-output cartridges (bottom center). The preamp uses the same LM833-N, feedback-based RIAA preamp circuit as the PH50 (middle center).

RIAA Accuracy

I'm always interested in running simulations on RIAA circuits, but feedback-based circuits with composite networks require a SPICE macromodel for the specific op-amp used. Unfortunately, TI doesn't have a macromodel for the LM833-N, but one for the LM833 is available for download. Since the gain-bandwidth product is nearly identical in these two devices, I decided to use the LM833 macromodel (I simulated this circuit with several audio-grade op-amps and found that the response in the upper two octaves varied by no more than 0.06 dB).

Figure 2 shows the simulation results, which demonstrates that the circuit is capable of RIAA accuracy ±0.3 dB from 20 Hz to 20 kHz. Actual performance will depend on component tolerances.

I measured the RIAA accuracy of my review sample using an Owon AG-1012F signal generator, my actively-buffered Jung-Lipshitz precision inverse RIAA network, and a Hewlett-Packard/Agilent 34401A digital multimeter, giving all equipment a two-hour warm up period. The Agilent meter has a decibel function with adjustable resolution of three to five decimal places. I outlined a procedure for measuring RIAA response in a sidebar to my article "Adcom's GFP-565 Preamplifier (Part 4)" (audioXpress, February 2004). Although my signal generator and meter are different, the procedure is essentially unchanged.

Figure 3 shows the measurement results. The measurement curve is very close to the simulation, with the left channel ±0.4 dB and the right channel ±0.36 dB, 20 Hz to 20 kHz. Both channels are around 0.3 dB down at 20 Hz, due to the 100 µF capacitor in the feedback network. From 30 Hz to 20 kHz, the left channel accuracy is ±0.3 dB and the right is ±0.28 dB.

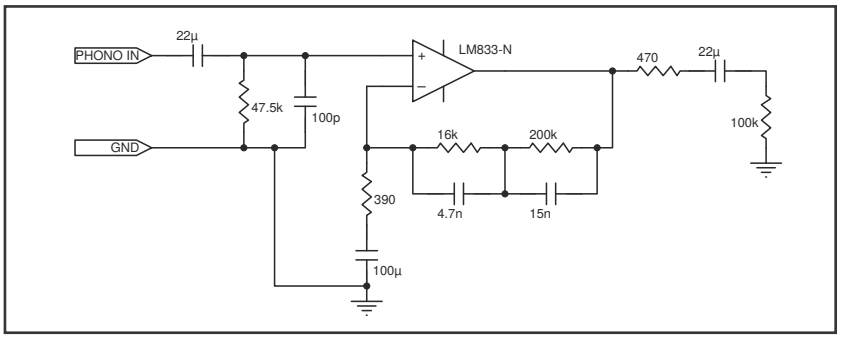


Figure 1: This is the TI-National phono preamp from the LM833-N datasheet, p. 14, as modified by Hafler. Hafler changed the input and output coupling capacitors to 22 µF.

Although perfectionist audio enthusiasts such as myself would like to see RIAA accuracy with ±0.1 dB, achieving that requires spending a great deal more money and invariably involves series or parallel combinations of off-the-shelf resistor values and parallel capacitors—often hand-selecting the parts for close tolerances. Even the renowned John Curl-designed Parasound JC-3+, priced at $2,995, claims ±0.2 dB, and a survey of under $500 phono preamps shows most are in the ±0.3 to ±0.5 dB range. The Hafler PH50 is certainly competitive with other phono preamps in this price range.

One feature of the PH50 is the variable rumble filter. This is a switchable, 12 dB/octave filter, based on another LM833-N op-amp, with a low-frequency cutoff that can be varied with a potentiometer on the front panel. The front-panel filter I/O switch activates a small relay mounted close to the output jacks.

Strangely, Hafler doesn't specify the low-frequency cutoff range. Measurements on my sample show it to be -3 dB at 23 Hz in the counter-clockwise position, 107 Hz in the center, rising to 490 Hz fully clockwise (the nominal values are probably 20 Hz, 100 Hz, and 500 Hz). This flexibility is much more sensible than the now-discarded (and discredited) International Electrotechnical Commission (IEC) amendment to the RIAA playback curve, which added a 7950 µS time constant. The IEC amendment put a fixed -3 dB rolloff at 20 Hz, which was invariably audible on systems with extended low-frequency response. The Hafler approach enables user discretion in dealing with rumble and record warp problems.

The PH50 is supplied with a wall wart external power supply rated at 15 VDC out at 400 mA. Wisely, Hafler has avoided the cheap, switching-mode wall warts that are becoming increasingly common with consumer audio equipment. Hafler uses a real 60 Hz power transformer, and keeping it external prevents hum from compromising the noise performance of the preamp. (The PH50's 14 AWG steel case also helps. This same case is used for all Hafler phono





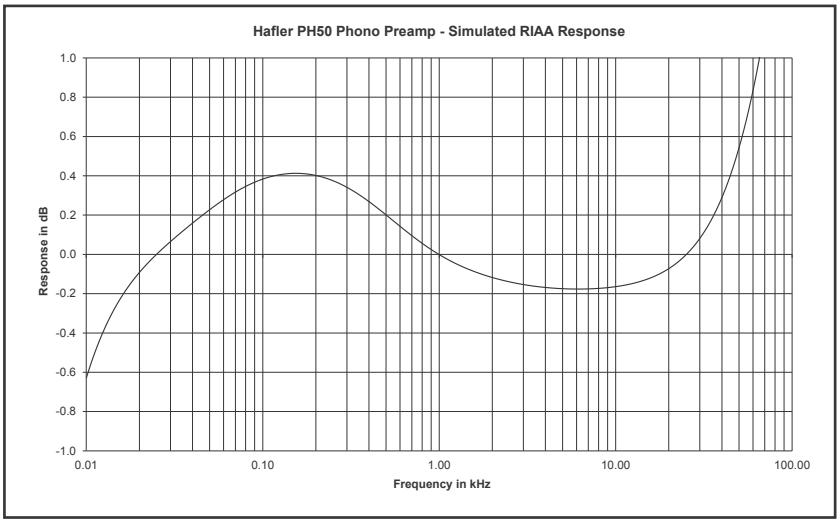


Figure 2: Computer simulation of the PH50 phono preamp circuit shows the RIAA accuracy to be better than ±0.3 dB, 20 Hz to 20 kHz.

products.) Under the load presented by the PH50, the transformer puts out 20 VDC. Inside the PH50 chassis, this is regulated down to +15 VDC with 78L15 three-terminal IC regulator. A third LM833-N is used as a supply splitter, delivering ±7 VDC to the phono preamp and rumble filter op-amps. The PC board in the PH50 is a conventional through-hole type, with plated through-holes. Hafler describes the PC board as being "military spec for greater durability" and it also notes that a "full surface ground plane reduces susceptibility to radio frequency interference."

There are no surface-mount components. All passive parts are leaded, and all resistors are metal-film types. The three op-amps are 8-pin DIP packages. The older products made by Dynaco and Hafler have long been favorites of modifiers. I'm sure the use of through-hole parts in their new phono preamps will make them attractive to DIY readers. Just remember that any modifications will likely void your warranty!

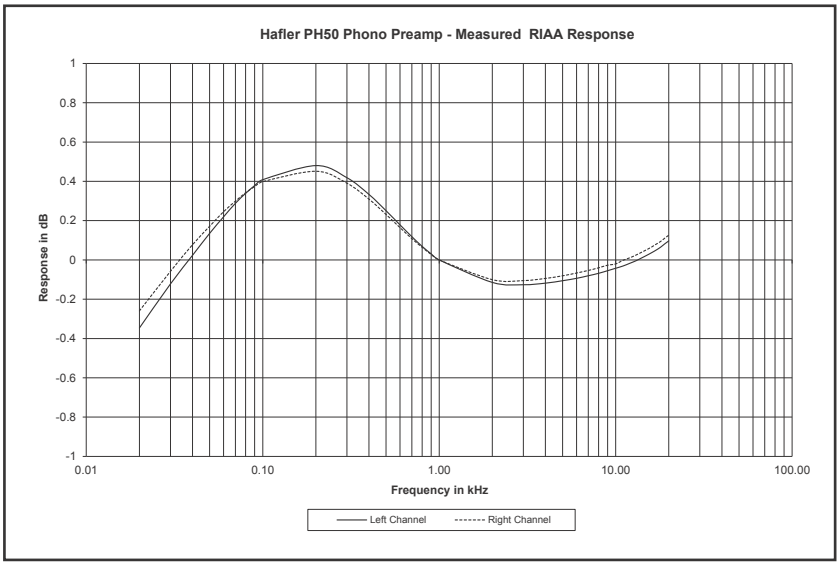


Figure 3: Measured RIAA accuracy of the PH50 phono preamp shows the response is very close to the simulation, ±0.4 dB left and ±0.36 dB right, 20 Hz to 20 kHz.

Vinyl Listening

I auditioned the PH50 using a Systemdek IIX turntable fitted with an Audio Technica AT-110E moving magnet cartridge. The AT110E is no longer part of Audio Technica's North American product line, but it can be purchased from LP Gear for less than $70. It's a remarkable cartridge for the money, offering performance that belies its modest price. With an output of 4.5 mV, the AT110E proved an ideal match for the 35 dB of gain provided by the PH50. The PH50 is exceptionally quiet—literally a silent partner during my listening sessions. At full preamp gain, with my ear up to the speaker (no music, of course!), there's just a trace of hiss. It offered clean, detailed sound with reasonably good soundstage information and dynamics. Compared to higher-end phono preamps, the soundstage is a bit narrow and the depth a bit short, but in this price range I have no complaints.

I also tried the PH50 with the Grado Signature XTZII that I normally use in my Grado Signature Tonearm, mounted on my custom AR-based turntable (see Resources for articles about the AR-based turntable). This turntable is fitted with George Merrill's GEM Dandy Rubber Cork Compound Platter Mat (which may be the best $60 you'll spend on your turntable). The moving-iron XTZII has low output for this type of cartridge—about 1.7 mV. Although I had to increase my preamp gain to compensate, the PH50's low noise floor was never audible during my listening.

Most importantly, the resolution of the PH50 is good enough to reveal the main differences between these cartridges, with the high-end Grado cartridge offering greatly increased detail, lower distortion, and better spatial characteristics. Incidentally, both turntables were powered by my low-distortion electronic speed control (see Resources for the original article describing this project), which I completely overhauled several years ago. There are a lot of choices in the under $500 phono preamp world, but the Hafler PH50 is a competitive product, and the unique rumble filter will make it attractive for many users.

Moving Coil Preamp

The PH60 moving coil preamp uses the same LM833-N-based RIAA circuit as the PH50 and the same type of construction, including the PC board and leaded components. The design is

transformerless. The high-gain front-end to step up the low output from moving coil cartridges is discrete, employing eight 2N4403 transistors in parallel. This is a well-known approach to moving coil step-up, combining high-gain with very low noise. The 2N4403 is a general-purpose, bi-polar PNP transistor with a current-gain-bandwidth product (fT) of 200 MHz. Since the paralleled transistors are operated single-ended, this gain stage is, by definition, pure Class A. The PH60 offers three choices for cartridge loading—50 Ω, 100 Ω, and 200 Ω—set with jumpers on the PC board. The factory setting is 100 Ω, which is the most common load requirement for moving coil cartridges. There are spaces on the PC board for adding a fourth load resistor if the three supplied resistors don't match your cartridge. Since the PC board accommodates leaded components, this will be an easy addition in the field.

The PH60 comes with the same wall-wart transformer as the PH50, supplying -18 VDC under the load presented by the preamp. (Polarity from the outboard supply is simply a matter of the preamp's DC supply connector orientation.) Since the discrete front-end is operated from a single-ended power supply, Hafler opted to power the LM833-N op-amp the same way. A 7912 three-terminal negative regulator steps the -18 VDC input down to -12 V to power the front-end and the op-amp. Electrolytic coupling capacitors are used at the preamp's input, output, as well as between the front-end and the LM833-N RIAA stage. The PH60 has a fixed, switchable passive rumble filter inserted between the front-end and the RIAA stage. Again, Hafler doesn't specify the rumble filter characteristic—I measured a 6 dB/octave roll-off with a -3 dB point of 90 Hz, which is higher than I would have expected.

I auditioned the PH60 using a Denon DL-103R moving coil cartridge mounted in my Grado arm. The performance of the PH60 was impressive, especially considering its reasonable price. The output of the Denon cartridge is 0.3 mV—rather low even for a low-output moving coil (Denon specifies 0.25 mV, but the calibration sheet that came with mine indicated 0.3 mV).

Many moving coil RIAA preamps are designed with a 1 kHz gain of 60 dB, but the Hafler's 70 dB of gain was definitely beneficial for this cartridge. The PH60 delivered plenty of output to my preamp, without any audible noise at any normal listening level. At full preamp gain the broadband hiss is a bit higher than the PH50, but still extremely low. There's no trace of hum. The preamp also revealed all of the essential qualities that moving coil advocates admire in these cartridges, including an airy, open treble region, excellent inner detail, and a large, precisely-localized soundstage. The DL-103R is slightly elevated in the treble compared to my more neutral Grado XTZII, and the PH60 simply did what it was told in this regard. At no time was the treble harsh or fatiguing. This preamp is good for long-term listening. Building a clean, quiet preamp to match the gain requirements of very-low output cartridges is a challenge for any

Manufacturer's Specifications

Hafler PH50 Phono Preamp
Audio circuit type: Active operational amplifier
Frequency response: RIAA curve
Total harmonic distortion (THD+N): 0.008%
Dynamic range: 91 dB
Input impedance: 47 kΩ
Maximum input: -21 dBu
Gain: 35 dB at 1 kHz
Clip level output: +12 dBu
Output impedance: 400 Ω
Equivalent input noise: -79 dBu
Noise floor: -79 dBu
Intermodulation distortion: 0.04%
Controls: Low-pass cutoff
Switches: Low-pass ON, power
Low-cut filter: 20 Hz to 500 Hz
Filter Slope: 12 dB/Octave
Size: 4.5" (w) × 6" (d) × 1.75"(h) (114 mm × 152 mm × 44.5 mm)
Power: 15 V, 400 mA external supply
Warranty: 3 year transferable

Hafler PH60 Moving-Coil Phono Preamp
Circuit: Active discrete transistor & operational amplifier
Inputs: RCA (left/right)
Output: RCA (left/right)
Impedance: May be set by user via internal jumpers—25 Ω, 50 Ω, and 100 Ω and custom
Gain: 70 dB at 1 kHz
Size: 4.5" (w) × 6" (d) × 1.75" (h) (114 mm × 152 mm × 44.5 mm)
Power: 15 V, 400 mA external supply
Warranty: 3 year transferable

Hafler PH34 Moving-Coil Step-up Transformer
Circuit: Passive transformer based design
Inputs: Gold-plated-RCA
Output: Gold-plated-RCA
Impedance: Optimized for 3 Ω to 5 Ω cartridges
Gain: 31 dB
Size: 4.5" (w) × 6" (d) × 1.75" (h) (114 mm × 152 mm × 44.5 mm)
Power: Not required
Warranty: 3 year transferable

Hafler PH44 Moving-Coil Step-up Transformer
Circuit: Passive transformer based design
Inputs: Gold plated RCA
Output: Gold plated RCA
Impedance: Optimized for 45 Ω cartridges
Gain: 20 dB
Size: 4.5" (w) × 6" (d) x 1.75" (h) (114 mm × 152 mm × 44.5 mm)
Power: Not required
Warranty: 3 year transferable



Photo 6: This is a front view of the Hafler PH44 moving coil step-up transformer. Jensen JT-44K-DX transformers provide 20 dB of gain and are optimized for higher-impedance, low-output moving-coil cartridges.

designer, especially in this price range. Hafler has certainly succeeded with the PH60.

Moving Coil Transformer

Some moving coil cartridge enthusiasts prefer a step-up transformer to an active front-end. In this case, the step-up transformer feeds a regular moving magnet phono preamp. A transformer is a passive device, and won't introduce any additional broad-band hiss, but at such low signal levels there's always a potential for hum.

Dave Hill, General Manager of Jensen transformers, provided some history behind the Hafler moving coil step-up products. Deane Jensen's research on moving coil step-up transformers began in the late 1970s, resulting in Jensen products that are still in production today. Jensen worked with an audio accessories company called Audio Interface, owned by Fujio Suganuma, based in Van Nuys, CA.



Photo 7: The inside view of the PH44 shows the pair of Jensen JT-44K-DX step-up transformers with double MU-metal shields, along with the steel chassis, ensure low noise. Input and output connectors are high-quality, chassis-mount, gold-plated RCA jacks.

"Numa-San" helped Deane with listening tests of the prototypes, which led to the creation of the Jensen JT-34K-DX and the JT-44K-DX moving coil transformers. Audio Interface became the first company to produce a high-performance, passive moving coil step-up device, the l CST-80, which could be purchased with either of the Jensen transformers. Hill notes that Deane Jensen was particularly proud of those transformers. The Audio Interface CST-80 is still highly regarded—they fetch real money on the used market when they turn up.

With Jensen now a division of Radial, it's only logical that Hafler would make a line of moving coil step-up devices using Jensen transformers. There are currently two Hafler step-up transformers, the PH34 and the PH44, named for the Jensen JT-34K-DX and JT-44-DX transformers they contain. The PH34 is designed for very-low-impedance cartridges and has 31 dB of gain and an input impedance of 31.5 Ω. The PH44 has 20 dB of gain and an input impedance of 430 Ω (see **Photo 6** and **Photo 7**). In the datasheets for these transformers, Jensen recommends an output Zobel network to ensure clean square-wave response with no overshoot or ringing – they're included in the Hafler products. The series R/C network is placed in parallel with the transformer secondary, 13 kΩ + 100 pF in the PH34 and 6.8 kΩ + 200 pF in the PH44. DIY builders can also purchase the Jensen transformers separately, and mount them in their own cases or preamps, adding the recommended R/C networks.

The PH44 was supplied for this review, since it's the best match for my Denon DL-103R. Although Denon recommends a 100 Ω load for this cartridge with an active front-end, they specify a 40 Ω load with a transformer. I asked Hill about this issue. Some transformer manufacturers suggest adjusting the primary input impedance by adding a resistor across the secondary, effectively in parallel with the 47 kΩ input impedance of a typical moving magnet phono preamp. The lower secondary loading will be reflected back to the primary, in accordance with the impedance ratio of the two windings.

Jensen strongly believes that this is the wrong approach, especially with an output R/C network already in parallel with the secondary. Lowering the 47 kΩ preamp load resistance will alter the square-wave response, degrading the sound, and also lower the transformer output level due to the additional loading. Jensen's approach is to put an appropriate resistor in parallel with the transformer primary, which won't affect the transformer's gain or square-wave response.

Hill recently helped a DIY customer find the best loading for the DL-103R cartridge. The customer

found that Denon's 40 Ω recommendation made the cartridge sound "way too heavy" and that a 100 Ω load made it sound "dark and veiled." The customer thought that the transformer's 430 Ω load was just right.

For readers who want to experiment with off-the-shelf resistor values, a parallel resistance of 130 Ω will lower the Jensen JT-44K-DX transformer's primary impedance to 100 Ω. A 47.5 Ω resistor will bring it down to 42 Ω. There's plenty of room to work inside the Hafler case, so soldering a resistor across each input jack is an easy task. My findings were different. To my ears, 430 Ω sounded too bright and thin, with a bit of hardness in the treble. I found 100 Ω to be better, but still too bright. I finally settled on 42 Ω, which provides a tonal balance very close to the PH60 active preamp with 100 Ω loading.

One benefit of adding a parallel resistance across the input is that it gives the cartridge a more uniform load with frequency. I used my BK Precision 879B LCR meter to measure the PH44's primary impedance, with a 47 kΩ load across the secondary. The 879B can measure impedance at four frequencies, and under the load presented by the transformer, its sine-wave output is just under 300 mVRMS. I checked the transformer secondary on my oscilloscope to make sure the transformer wasn't saturating at any of the test frequencies. I measured 375 Ω at 100 Hz, 433 Ω at 1 kHz, and 343 Ω at 10 kHz. The variation in impedance is because inductive reactance of the transformer primary coil varies with frequency (this will be true

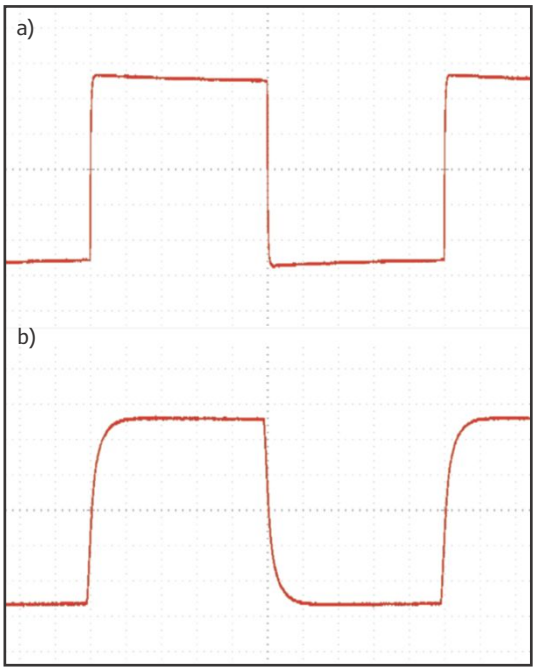


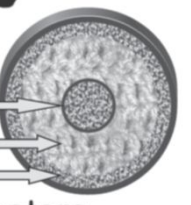
Figure 4: This shows a 1 kHz (a) and 10 kHz (b) square wave response of the PH44 moving coil step-up transformer. Generator output impedance was set to 25 Ω, with a 47 kΩ resistive load across the transformer secondary, as specified by Jensen Transformers. The PH44 contains a Zobel R/C network across each transformer secondary to eliminate ringing or overshoot.

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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 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 and 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, a frequent presenter at ARSC conferences, and has authored numerous articles and reviews in the *ARSC Journal*. Gary 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.

of any transformer). After adding a 130 Ω resistor across the primary, the readings were 97 Ω, 100 Ω, and 98 Ω at those same frequencies. With 47.5 kΩ across the primary, the readings became 43.2 Ω, 42.8 Ω, and 42.7 Ω. With the purely-resistive portion of the load now dominant, the impedance "curve" is very uniform. This should be beneficial to the performance of the cartridge.

Opinions on the Internet about loading for the DL-103R are all over the map, from as low at Denon's recommended 40 Ω to as high as 1 kΩ. A few users note that 47 kΩ is the load Denon claims they use to generate the individual frequency response curves that come with each cartridge. This is true, but it can't be right. Denon makes its own AU-300LC moving coil step-up transformer for the DL-103 series of cartridges, with a primary impedance of 40 Ω. If Denon used its own transformer to make the measurement, it couldn't possibly have loaded the cartridge with 47 kΩ. I believe that 47 kΩ probably refers to the loading of the transformer secondary, which is the load presented by virtually any moving magnet phono preamp it's likely to be used with.

The diversity of opinion about cartridge loading may be related to the associated equipment. I use wide-band, solid-state equipment, including a Benchmark DAC2-HGC DAC/preamp, my own custom built preamp, and a pair of Benchmark AHB2 power amps. Tube equipment often has a softer treble region, and less well-defined bass, which may explain the preference for higher-impedance loading by some users. There's clearly some "black art" involved here.

Transformer Performance

All of my past experiences with Jensen transformers have been extremely positive, and the PH44 is no exception. **Figure 4** shows the transformer's square-wave response at 1 kHz and at 10 kHz. The performance is excellent, and the absence of ringing or overshoot shows the effectiveness of the output R/C network. I auditioned the PH44 with my own custom-built moving magnet phono preamp (see Resources). The combination yielded extremely clean, detailed sound with a large, deep, well-defined soundstage. With my preamp gain set to maximum there's not a trace of hum, and no more broadband hiss than that of the moving magnet preamp that it feeds.

But, getting the best performance from this transformer is dependent on proper cartridge loading. The Jensen transformers are sonically transparent and will reveal even small changes to their operating environment or that of the cartridge itself. If I were to make one suggestion to Hafler, it would be to add a small PC board at the input, enabling the user to easily change load resistors with jumpers, as they have in the PH60 preamp.

The PH44 is a winner. It's refined enough to be used with the finest moving magnet phono preamps, and I recommend them to any moving coil user who prefers a step-up transformer to an active front-end. All of the Hafler products discussed in this article are made in Canada and feature robust construction. The long-term reliability should be excellent.

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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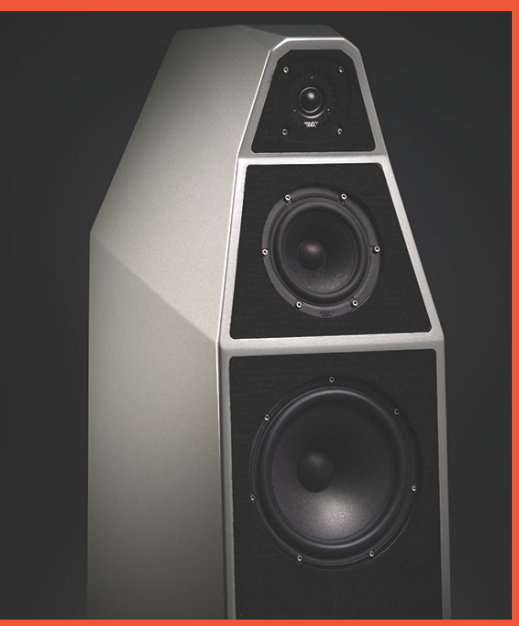
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LP Gear | www.lpgear.com







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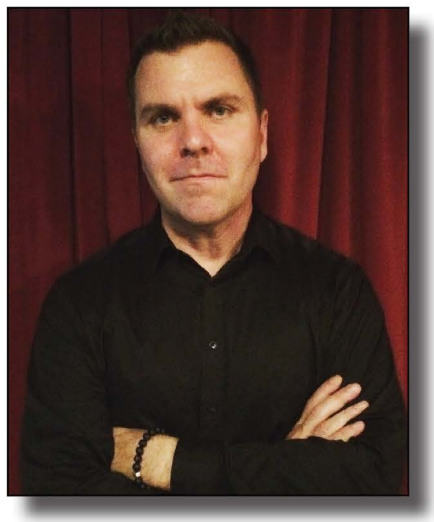






Prepping the Next Generation of Audio Engineers

An Interview with Drew Waters of SPARS



Drew Waters' love of higher education and the music/entertainment industry makes him a natural fit for the Society of Professional Audio Recording Services (SPARS).

By
Shannon Becker
(United States)

SHANNON BECKER: Tell us a little about yourself.

DREW WATERS: My undergraduate degree is in Sociology, and my Masters is in Jazz and Contemporary Media from the Eastman School of Music (Rochester, NY). I have a dual Ph.D. in Performance and Composition from New York University.



SPARS mentors work with universities and colleges to bring mentors to students and host speed-mentoring sessions at industry events, including the Audio Engineering Society (AES) conventions.

I was a professor for the State University of New York (SUNY) system, then at California State University (CSU) Monterey Bay. I left that position for a job at Capitol Records—VP of Studio Operations, Archives and Strategy. Recently I've been Dean of Academic Affairs for two different universities in California.

SHANNON: How did you first become interested in the audio industry?

DREW: I started playing electric bass as a teenager and doing gigs after years of reluctantly studying piano. Soon after, I bought a four-track Tascam cassette recorder, which started it all. I was a freelance bassist for years. Spending time in studios and working with engineers inspired me to learn how to make better recordings and I slowly acquire gear I needed (wanted). Educational resources back then included person-to-person sharing of information, if you happened to be in a large metropolitan area, and outdated books.

SHANNON: What is your primary area of interest?

DREW: I'm interested in the intersection between higher education and the music/entertainment industry. I have a studio in Los Angeles, CA, where I record and perform. More recently, I've

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Several SPARS members attend the 2016 AES convention in Los Angeles, CA, to meet with mentees and encourage aspiring audio engineers.

started shooting digital video as well as 8 mm and 16 mm film.

SHANNON: How has the recording industry changed in recent years? What do you see as the driving force behind the changes?

DREW: Digital audio has been the driving force for most changes in the audio industry audio for the past 40 years. Recent changes include the emerging industries of virtual reality (VR) and gaming. They offer enormous opportunities for both new and experienced audio engineers.



Scott Adamson talks to several mentees in a Live Sound session.

SHANNON: How did you get involved with the Society of Professional Audio Recording Services (SPARS, www.spars.com) and why?

DREW: SPARS approached me when I was VP at Capitol Records. I was a board member for a while, then president. It was a good fit as I have experience in both the music industry and higher education. If we can save a mentee—a student, a recent graduate or a person interested in the various opportunities within the audio industry—months or years of frustration by facilitating a meeting to gain insight from an established engineer or producer (mentor), our mission is accomplished.

SHANNON: As the current president of SPARS, can you please provide some background about the organization.

DREW: SPARS began as a Trade Group—an idea formed by a group of studio owners meeting at The Pacific Dining Car in downtown Los Angeles—during the Audio Engineering Society (AES) Convention in the fall of 1979. The goals were to provide a way for studios (initially, elite studios as there was a $2,000 fee to join once accepted) to have a forum toward striving for excellency in the craft, evaluate gear, best practices, and educate members. Those studios included: A&R Studios (NYC), Atlantic Studios (NYC), Criteria Recording Company (Miami, FL), Flyways-Header Recording (Hollywood, CA), Group IV Recording (Hollywood, CA), House of Music (New Jersey), Howard M. Schwartz Recording (NYC), Canteen Recording (Burbank, CA), Larrabee Sound (Hollywood, CA), Media Sound (NYC), Record Plant (Los Angeles, CA), Regent Sound Studios (NYC), Sigma Sound Studios (Philadelphia, PA), Sound Mixers (NYC), and Studio 55 (Chicago, IL).

Later, the Manufacturer Interface Program began and extended SPARS membership to manufacturers to create an effective dialogue with studio owners and operators from across the country. SPARS members could speak directly to manufacturers to discuss new product development, service, and pricing and manufacturers could assess the market place directly with stakeholders. Those manufacturers included Ampex, MCI, 3M, Neve, Otari, and SSL

SHANNON: How has SPARS role in the industry evolved over the years?

DREW: There is an age-old inverse function between technological advancements and price points. The better the technology gets, the cheaper it is. In

this case, consumers were slowly emancipated from the steep cost of analog gear and began to record themselves at home, to simplify the evolution. Manufacturers began to make gear for these types of users, and their stakeholders went from a handful to thousands. Less tape was used and smaller footprint studios required smaller gear. Pro-sumer level gear became ubiquitous. All of this combined reshaped SPARS and the relevancy it once had within the original membership.

In 2013, SPARS became a 501(c)3 (educational nonprofit organization) with the enormous efforts of Paul Christianson and Kirk Imamura. SPARS strives to provide educational resources and best practices, mentoring is an extension of that, and embraces a wider group of people.

SHANNON: Can you explain more about its mentoring and education initiative?

DREW: In working with both individuals and educational institutions we feel we can have the greatest impact on a prospective engineer's trajectory. We work with universities and colleges to bring mentors to students as well as speed-mentoring sessions at industry events. We strive to let people know that being an engineer is not only, and should not be, limited to triangulating one's self between two speakers in front of a large-format console.

There is a wide world of experiences in gaming, live, VR, film, and video. We select mentors who are experienced, encouraging, innovative, and reflect the contemporary industry. Recent mentors include Niko Bolas, Tony Schultz, Jeri Palumbo, Jeff Greenberg, Mark Rubel, Barry Rudolph, Dave Hampton, Joel Hamilton, Eric Johnson, Leslie Mona-Mathus, and Leslie Anne Jones

SHANNON: What does SPARS have planned for 2017?

DREW: We are undergoing an overhaul of the entire spars.com website as well as re-assessing our social media strategies with Paul Christianson, Kirk Imamura, and Sherri Tantleff. And, we will continue to provide speed-mentoring at the next AES convention in the fall of 2017.



DIM-100—A Device for Interfacing and Measuring with Sound Cards (Part 2)

Assembly and Testing



In the second part of this article, I will discuss the DIM-100's assembly and the test measurement results I obtained once it was completed. When you finish your own DIM-100, you will have a device to interface and measure your audio equipment.

By
George Ntanavaras
(Greece)

The DIM-100 is a great device that will prove useful for taking any audio measurements. To ensure the build quality, I designed the PCBs using the Eagle Layout editor Demo version, which can be downloaded for free from the Cadsoft website (https://cadsoft.io). It is a fully operational version, with the only limitation being the maximum dimensions of the PCB, which was another reason why I split the device into three PCBs.

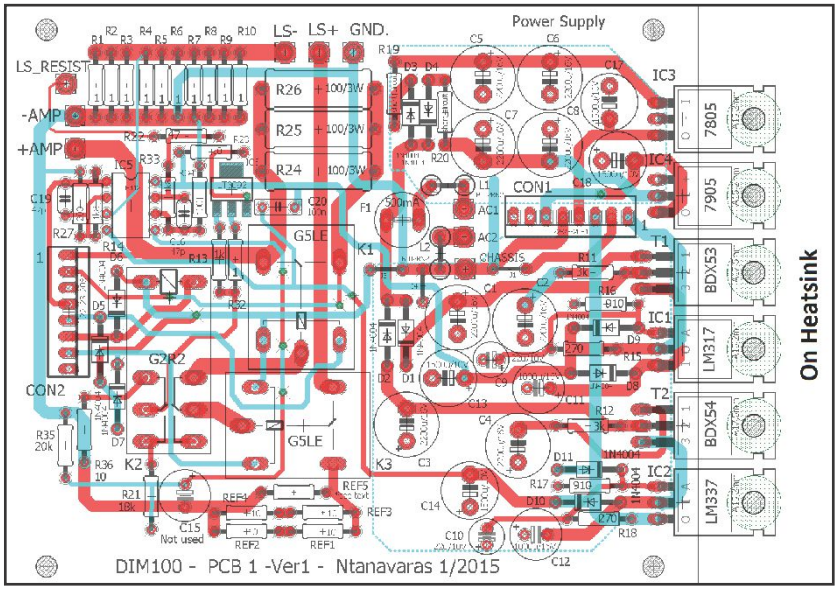


Figure 1: I began with the guide assembly for PCB1.

Device Assembly

For the DIM-100, I used high-quality PCBs manufactured from FR4 composite material with a 1.6 mm thickness and 35 µm copper solder on both sides and silkscreen on the top side. The guide assemblies are detailed for PCB1 (see **Figure 1**), for PCB2 (see **Figure 2**), and for PCB3 (see **Figure 3**).

The Parts List—provided in the Supplementary Material section of the audioXpress website—gives a detailed list for every PCB, with some additional components that can be used in the assembly of the DIM-100's front panel.

An aluminum enclosure houses the PCBs and the device's connectors. This offered good RFI/EMI protection for the DIM-100. The enclosure's external dimensions are: 175 mm (width) × 66.6 mm (depth) × 275 mm (height).

Using the "Front Panel Designer" program (www.schaeffer-ag.de), I also designed a new front panel for the DIM-100. I sent the output file for manufacturing and when I received it, I replaced the enclosure's original panel with the new one. The result was a more professional-looking device that was also easy to operate. Additionally, the assembly was much simpler.

Figure 4 shows the assembly drawing I followed to build the DIM-100. **Photo 1** shows an internal view of the assembled device. To begin, I placed PCB1 on the bottom of the metal box and screwed the device's four regulators and two transistors to the bottom side of the box, which I used as a heatsink. All of them were electrically isolated from the box with mica insulators. This was necessary because the device's maximum consumption is high at about 360 mA and heatsinking was required to keep the temperature of the regulators and the transistors low.

I attached the other two PCBs (PCB2 and PCB3, which is actually the RMS voltmeter) to the front panel with their switches. At the bottom of the enclosure, I placed a terminal strip that I used for the four coaxial cables that connect the device with the sound card. I permanently soldered these cables to the terminal strip.

The connections between the PCBs were easily made using the three cables with connectors CON1, CON2, and CON3 at their sides. I have provided a description of the pins for all the connectors used in the device.

CON4 and CON5 are used to connect all the device's inputs and outputs. I used a 2 mm² wire to connect the ground pins of all the input and output connectors together. This wire was connected to the AGND pin of PCB2 near connector CON5 and to the chassis.

Testing and Calibrating the RMS Voltmeter

After I finished assembling the DIM-100, I tested and calibrated the RMS voltmeter because I wanted to use it to test the device's other circuits. To calibrate it, I used my TTi 1604 True RMS Multimeter as the reference voltmeter. It provides 4.75 digits for measuring true RMS AC voltages with an accuracy of 0.5% for the frequencies between 45 and 400 Hz. From 400 Hz to 5 kHz, the accuracy drops to 1% and as the level and the frequency are increased, the accuracy gets worse.

I set my sine wave generator at 220 Hz, which is in the middle of the best accuracy frequency range and at 120 mVRMS. I set the switches of the multimeter to the "V" and "0 dB/200 mV" positions,

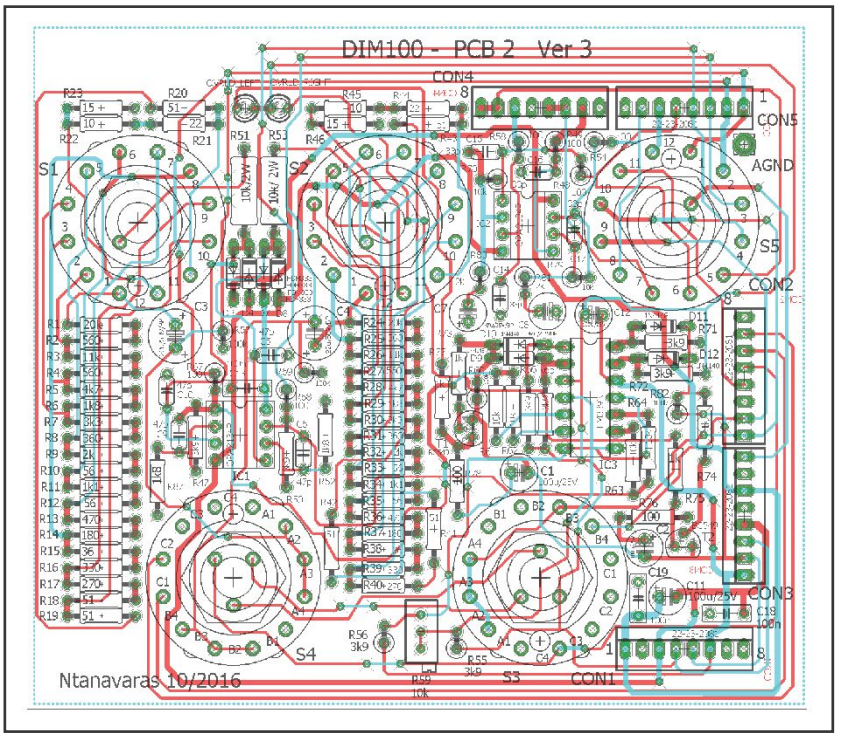


Figure 2: This is the guide assembly for PCB2.

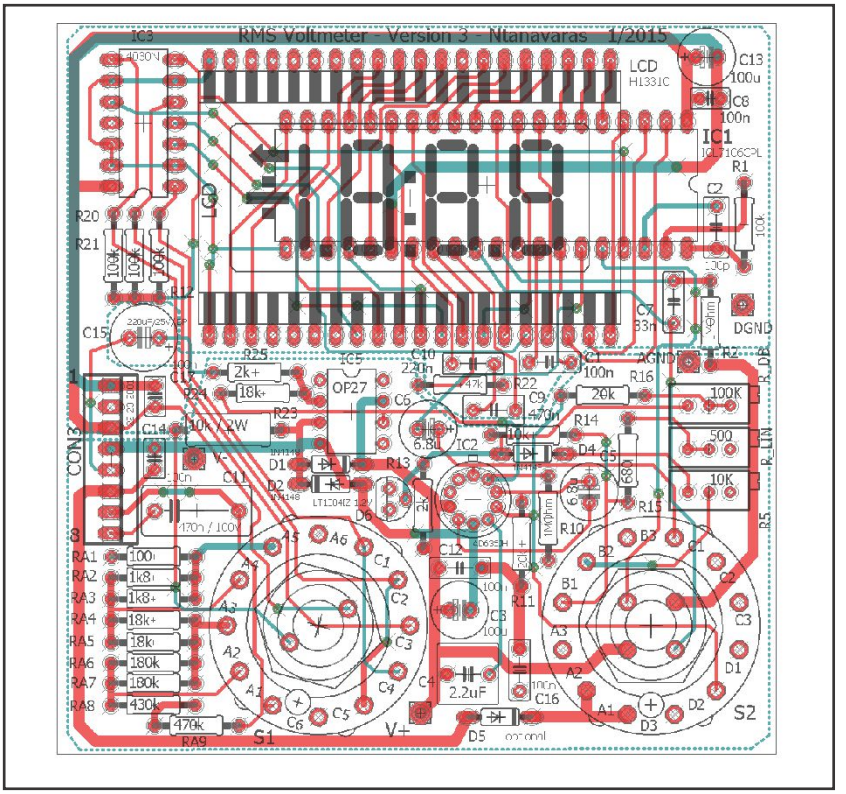


Figure 3: Here is the guide assembly for PCB3, the RMS voltmeter.

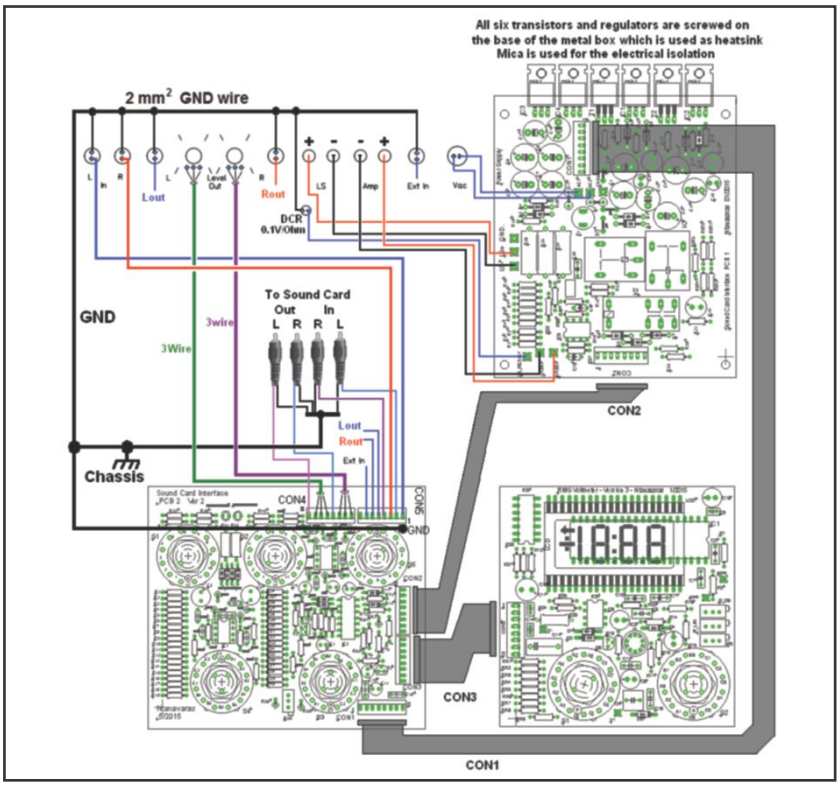


Figure 4: The drawing details how to finish assembling the DIM-100.

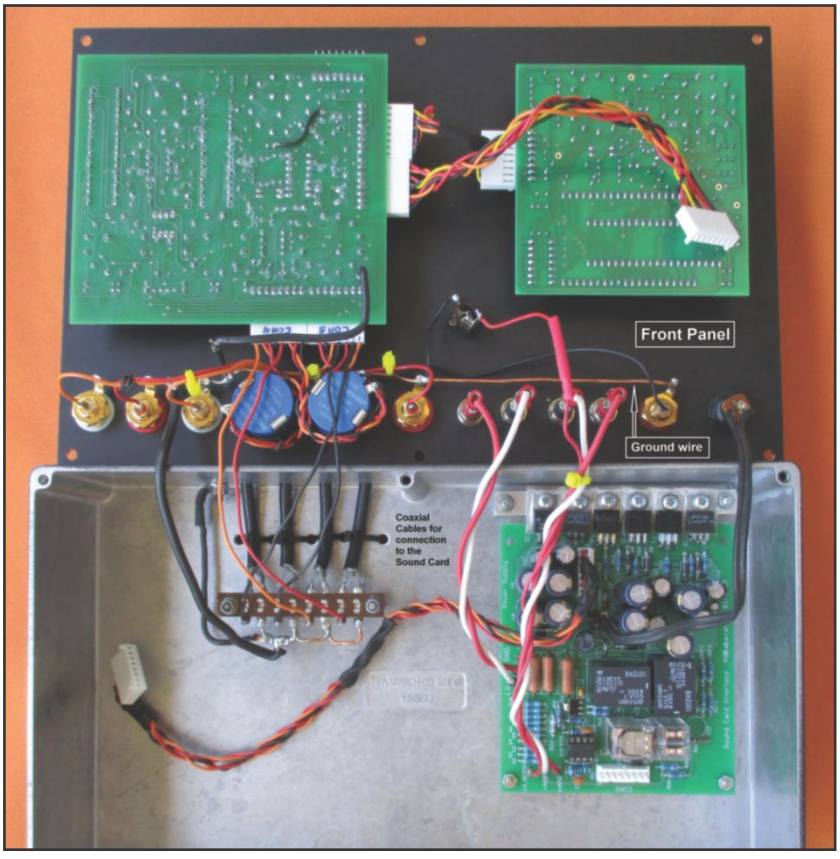


Photo 1: Here is an internal view of my DIM-100.

and I adjusted the trimmer R_LIN (500 Ω) until the voltmeter reading was exactly the same as the reference meter reading.

Next, I adjusted the generator's output level to 12 mV/1.2 V/6 V and the multimeter's position to 20 mV/2 V/20 V. I also checked that the reference multimeter reading and the RMS multimeter reading were the same.

Connector Pin Descriptions

(Note: CON1-1 means Connector 1, pin 1, etc.)

CONNECTOR CON1

CON1-1 : V analog (+)
CON1-2 : Analog Ground
CON1-3 : V analog (-)
CON1-4 : V digital (+)
CON1-5 : Digital Ground
CON1-6 : V digital (-)
CON1-7 : Not used
CON1-8 : Not used

CONNECTOR CON2

CON2-1 : Vamp (Power amplifier voltage input to the device)
CON2-2 : Vls (loudspeaker voltage)
CON2-3 : Not used
CON2-4 : LS current (loudspeaker current)
CON2-5 : Relay 1 (Relay 1 drive voltage)
CON2-6 : Relay 2 (Relay 2 drive voltage)
CON2-7 : Relay 3 (Relay 3 drive voltage)
CON2-8 : Not used

CONNECTOR CON3

CON3-1 : V digital (-)
CON3-2 : Digital ground
CON3-3 : V digital (+)
CON3-4 : V analog (-)
CON3-5 : Analog ground
CON3-6 : V analog (+)
CON3-7 : Voltmeter AC+DC input
CON3-8 : Voltmeter AC input

CONNECTOR CON4

CON4-1 : Right volume potentiometer GND
CON4-2 : Right volume potentiometer middle
CON4-3 : Right volume potentiometer top
CON4-4 : RSCout (Right channel output of sound card)
CON4-5 : Left volume potentiometer GND
CON4-6 : Left volume potentiometer middle
CON4-7 : Left volume potentiometer top
CON4-8 : LSCout (Left channel output of sound card)

CONNECTOR CON5

CON5-1 : Lin (Left input of the device)
CON5-2 : LSCin (Left input of the sound card)
CON5-3 : Rin (Right input of the device)
CON5-4 : RSCin (Right input of the sound card)
CON5-5 : Out L (Left channel output of the device)
CON5-6 : Ext In (External input for the voltmeter)
CON5-7 : Out R (Right channel output of the device)
CON5-8 : Not used

To calibrate the multimeter in the decibel range, I set it to the "dB" and "0 dB/200 mV" positions. I adjusted the level of the generator to 775 mV. (It should be exactly 774.6 mVRMS, if your reference multimeter can measure with such accuracy.) Then, I adjusted the trimmer R_DB (100k) until the meter read exactly 0.0 dB. The level of the generator was reduced by 40 dB to 7.7 mV (exactly 7.74 mVRMS). I then adjusted the trimmer R5 (10 kΩ) until the display read -40.0 dB. Once I finished calibrating the voltmeter, I tested the remaining circuits.

Testing the DIM-100

LTspice is a high-performance SPICE simulator, schematic capture, and waveform viewer offered for free from Linear Technology (www.linear.com). I used this program to check the accuracy of the input attenuator. I entered the circuit exactly as shown in the schematic listed as Figure 3 in the first part of this article "DIM-100—A Device for Interfacing and Measuring with Sound Cards (Part 1): Begin with the Design," (March 2017). The circuit has a sinus signal of 1 kHz/1 VRMS as an input, and LTspice calculated the values for each one of the 12 positions of the S1 and S2 switches (see **Table 1**). The decibel indications of the multimeter for values above 0 dB are not accurate, so I haven't included any values for the top three values.

I used the DIM-100's voltmeter and my signal generator to verify the values shown in **Table 1** for every position of the switches S1 and S2, while the S5 switch was set at the LSCin and RSCin positions. The measurements were almost identical to those shown in **Table 1**, confirming that the attenuator had been accurately calibrated.

When S1 and S2 switches were set under the -40 dB position, the voltage levels are very low. So I re-calculated them with the LTspice, and increased the input signal to 3500 mVrms. The new values are shown in **Table 2**.

Next, I checked the device's output section. I used my signal generator as a sound card output and I connected my oscilloscope to the device's L output in the front panel. I checked that the volume potentiometer operated smoothly and the waveforms of different frequencies were without any distortion. I also measured the DIM-100's maximum output at 3.5 VRMS (10 V_{PP}). This is the maximum level with the op-amps and the voltage supply levels that I used.

S1, S2 Positions	Output to the Sound Card Multimeter in mVRMS	Output to the Sound Card Multimeter in decibels
+10 dB	3165.8 mVRMS	*
+5 dB	1780 mVRMS	*
0 dB	1001 mVRMS	*
-5 dB	563.3 mVRMS	-2.8 dB
-10 dB	316.62 mVRMS	-7.8 dB
-15 dB	178.06 mVRMS	-12.8 dB
-20 dB	100.15 mVRMS	-17.8 dB
-25 dB	56.34 mVRMS	-22.8 dB
-30 dB	31.68 mVRMS	-27.8 dB
-40 dB	10.04 mVRMS	-37.7 dB
-50 dB	3.16 mVRMS	-47.8 dB
-60 dB	1.01 mVRMS	-57.7 dB

Table 1: I used the LTspice program to calculate the values for each of the 12 positions of the S1 and S2 switches.

Then, I continued to test the other circuits. I connected my generator to an external power amplifier and the amplifier's output to the device's "- Amp +" sockets. I also connected a test load of 8 Ω/100 W to the device's "+ LS -" sockets.

The generator was set at 220 Hz and I adjusted the level, until the amplifier's output (measured with my reference voltmeter) was 1,500 mV. I set the device to the "Impedance" and "LS" mode of operation and the "Voltmeter" switch to the "Vamp" position. I verified that the DIM-100's voltmeter also read 1,500 mV. Then, I set the "Voltmeter" switch, first to the "Vls" position where the multimeter read 294 mV and then to the "LScur" position where the multimeter read 36.2 mA.

To verify that the values were correct, I calculated the theoretical values from the diagram shown in **Figure 5**. There, I show the DIM-100's internal connections between the input sockets for the power amplifier, the series resistors 33.3 Ω

L or R Input (mVRMS)	S1, S2 Positions	Output to the Sound Card Multimeter in mVRMS	Output to the Sound Card Multimeter in decibels
3500 mVRMS	-40 dB	35.13 mVRMS	-26.9 dB
3500 mVRMS	-50 dB	11.08 mVRMS	-36.9 dB
3500 mVRMS	-60 dB	3.54 mVRMS	-46.8 dB

Table 2: I used LTspice to recalculate the values with the input signal increased to 3500 mVRMS.

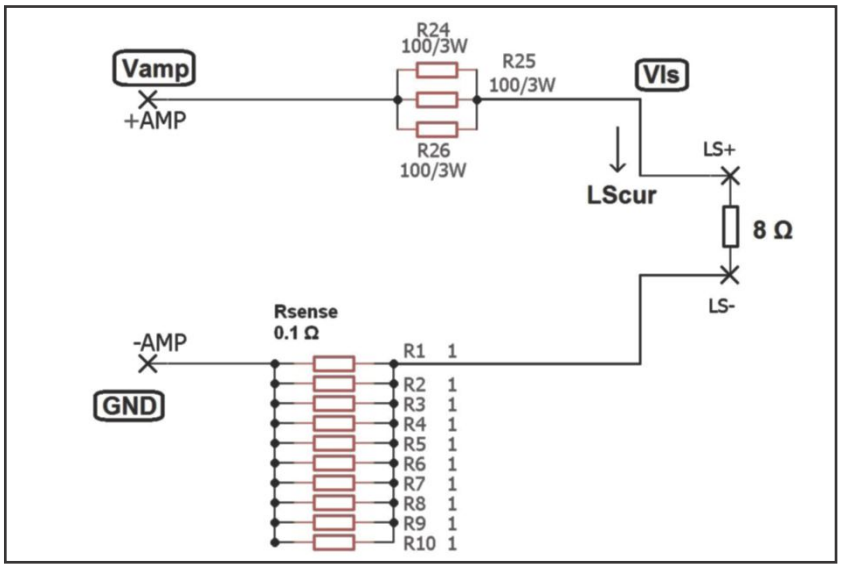


Figure 5: This drawing shows the connection between the power amplifier and the loudspeaker.

Frequency	THD (sound card only)	THD (sound card and DIM-100)
400 Hz	0.0077%	0.0077%
2400 Hz	0.0072%	0.0067%
8400 Hz	0.0066%	0.0062%

Table 3: I tested the DIM-100 at different frequencies to ensure that it did not add any additional distortion.

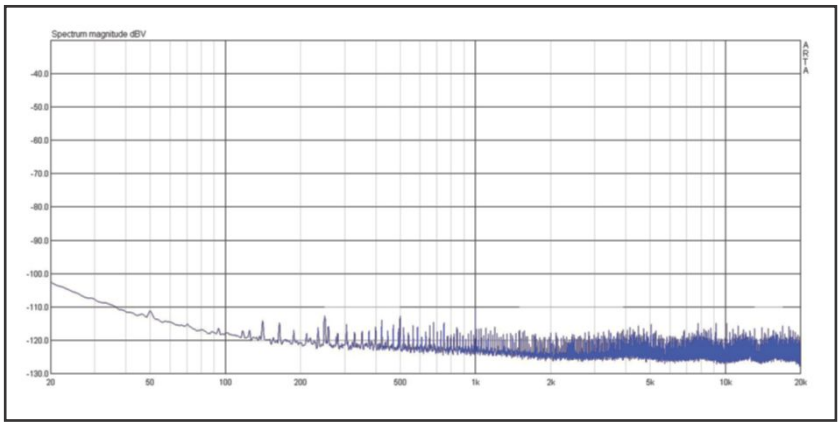


Figure 6: This is the device's output noise spectrum with the input short circuited.

used for the impedance measurements, the 0.1 Ω resistor that is used for the current measurement, the 8 Ω load, and the points of the measurements.

Looking at this circuit, the current is calculated:

$$LScur = \frac{1500}{33.3+8+0.1} = 36.23mA$$

which is almost the same as the measurement we mentioned earlier.

The voltage between Vls and the GND, includes the 8 Ω load and the 0.1 Ω sense resistor and can be calculated as follows:

$$Vls = \frac{8.1 \times 1500}{33.3+8.1} = 293.48mV$$

Next, I tested the circuit that measures the DC resistance of the loudspeaker driver's voice coil by setting the switches to the "Impedance" and "DCR" positions. I connected an external DC voltmeter at the front panel's "DCR 0.1 V/Ohm" output to measure the voltage, which should be 800 mV, corresponding to 8 Ω. The circuit shown in **Figure 5** indicates there will be some inaccuracies, because the measurement additionally includes in series the device's 0.1 Ω sense resistor, the internal wiring (relay contacts, PCB tracks), and the external wiring (cables and connector contacts). All these could be separately measured and then subtracted from the measurement if high accuracy is required. Another source of error is the offset voltage of the low offset IC5-LF412 op-amp. I had four pieces available and I found that one of them that had almost zero offset.

To adjust the CLIP voltage, I connected the device to the sound card and connected the device's Left output directly to the device's Left input.

Then, I set the ARTA software to the spectrum analyzer function with the signal generator producing an 1 kHz sine wave. The analyzer has a nice feature that indicates the input level in decibels. I adjusted the trimmer R59 so that the CLIP LED illuminated when the analyzer indicated -0.5 dB. The voltage to the line input of my sound card was measured at 0.43 VRMS. I also tested the Right channel, which demonstrated a similar behavior.

It is important that the device does not add any distortion or noise to the measurements, otherwise, it will not be very useful. As a reference, I measured the distortion of my sound card by directly connecting the input to the output. Then, I inserted the device into the loop and measured the distortion again. **Table 3** shows the results, which were similar, with no additional distortion, although for frequencies above 2 kHz, the distortion was a little lower when I inserted the device. Probably,

the driving capabilities of my sound card's output circuit was not as good as that of the device for this range of frequencies.

Figure 6 shows the DIM-100's low total noise with its input grounded in combination with my sound card, as measured by the ARTA.

Operation of the Device

Figure 7 shows the DIM-100's typical connection to perform a loudspeaker measurement. I used an external power amplifier to drive the loudspeaker and a microphone with its external preamplifier for the measurements.

I measured the frequency response with the switch S3 in the position "Response" where the device's Left input is internally connected to the power amplifier's output as the reference signal. The "Bypass" position can be selected and the Left input should be externally connected to the reference signal of the measurement.

The calibration procedure, as needed by the ARTA software, can be performed by turning the switch to the "Calibrate" position, while the impedance measurement can be performed without needing to change the cabling by simply switching to the "Impedance" position.

Conclusion

The DIM-100 is a useful device for interfacing and measuring all the signals to and from a sound card. It offers a lot of capabilities and it is useful for people who spend a lot of time measuring audio equipment, especially loudspeakers.

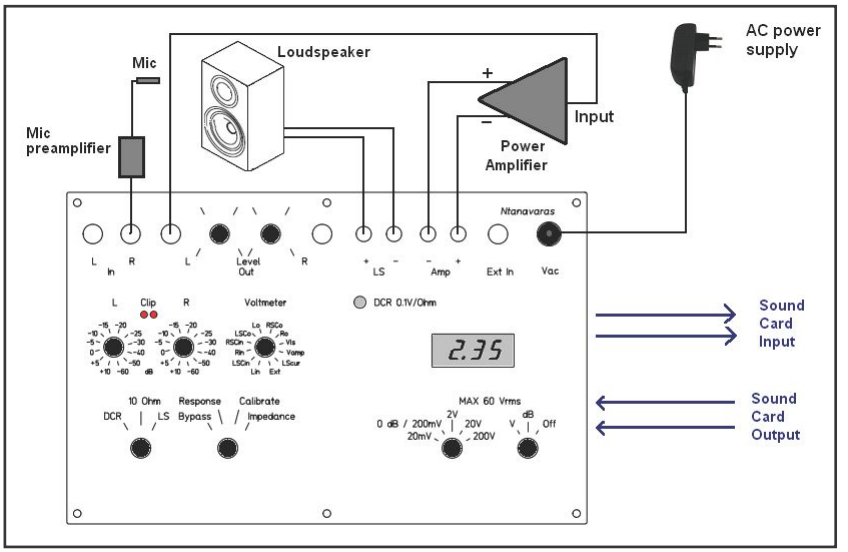


Figure 7: This is the setup I would recommend using to obtain good measurements with the DIM-100.

About the Author

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

Project Files

To download the Parts List, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html.

Sources

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

Eagle software program
Eagle PCB Design | https://cadsoft.io

Ltspice software program
Linear Technology | www.linear.com/designtools/software/#LTspice

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



Use 6C33 Tubes to Build a Push-Pull AB Amplifier



Photo 1: This powerful amplifier was easy to build and meets all the hi-fi requirements.

The 6C33 tube originates from Russian military hardware. It is built as a current control tube and is the control element in power supplies. The fact that the tube is designed for military applications can also be seen in its technical data. After the collapse of the Iron Curtain, many tube lovers discovered this tube type and conducted experiments with it. This article features a power amplifier that is in line with hi-fi requirements, powerful, and easy to build.

By
Gerhard Haas
(Germany)

Original article published in German in *Elektor Roehren Sonderheft 7*, May 2011.

I designed this push-pull AB amplifier using 6C33 tubes (see **Photo 1**). Just by looking at the 6C33, you can tell that this vacuum tube is built for heavy-duty use. **Photo 2** shows the electrodes, the supporting elements, and the glass bulb, which features a high material thickness in contrast to the otherwise conventional audio tubes. And the tube's datasheet confirms its military background.

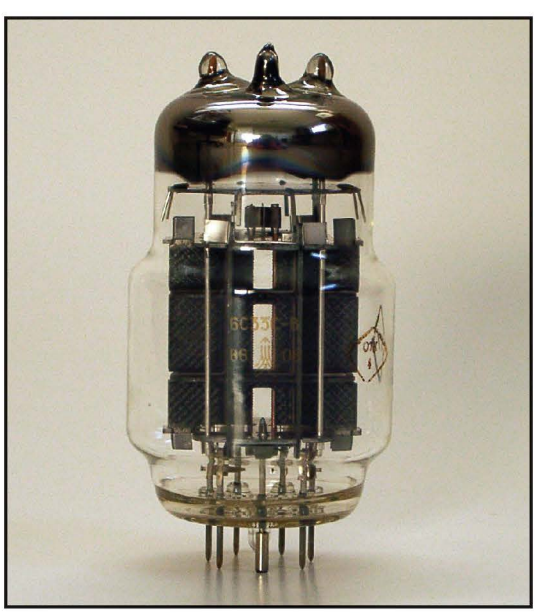


Photo 2: The 6C33 vacuum tube was originally used by the Russian military.

The 6C33 Tube

Depending on the model, continuous vibrations from 4 to 6 g are allowed. The "g" stands for the acceleration due to gravity measuring 9.81 m/s^2. In case of pulsed shocks, the vibrations may measure between 35 and 150 g, depending on the model. Such a high degree of resistance to vibration and shock is perfectly appropriate for use in fighter jets, tanks, and warships. However, these characteristics are less important for hi-fi purposes. What most tube lovers find fascinating is the 6C33 is a triode with a very high current output. And, powerful hi-fi power amplifiers can be built with it, if used correctly. Furthermore, the 6C33 can operate at relatively low operating voltages.

The Russian datasheets complicate the correct use since much of the data is specified on a "from up to" basis. For instance, the heating current at 12.6 V is specified as 2.8 A up to 3.6 A. It would not be wrong if the power supply was designed for the maximum possible current. However, in the case of tubes that are close to the lower limit, the power supply is oversized by about 22%. The tube's transconductance (μ) at an average 40 mA/V is very high, the idle amplification of μ = 4 is very low. Likewise, a grid current of 5 μA must be expected. According to the datasheet, this necessitates a grid bleeder that may not exceed 200 kΩ.

On Internet forums and some printed publications, you can find circuits using the 6C33 that feature significantly higher grid bleeders. In this case, either the development has not been carried out thoroughly and/or something has been cobbled together, which ultimately means the device will not properly function for long. When applying the correct circuit design, the active component is forced to its operating limit. Likewise, clever dimensioning makes it possible to use almost any kind of tube, except for a few outliers, without limitations. With all the active components, you should remember that tolerances of ±20% up to ±30% are quite common.

Another peculiarity of the 6C33 is that it actually consists of two triodes that are connected in parallel. It is possible to activate one or both systems using the heating voltage. A heating voltage of 6.3 V is provided for each triode. To take full advantage of the tube, the filaments should be connected in series to 12.6 V and both systems should be used. The 6C41 is the smaller version. It features half a 6C33 and is heated with 6.3 V.

Alternate Tube Choices

Because the grid bleeder must be of low impedance for the 6C33 and the preamplifier and the driver stage are supposed to supply amplification, it is not possible to use low-current double triodes (e.g., the ECC 83). Low-impedance circuit technology also requires large coupling capacitors. All this must be operated by appropriate high-current driver stages. This ensures that the frequency response does not significantly drop at the upper and at the lower end inside the amplifier. Therefore, with this design it is important to have the correct low-impedance circuit technology and build the amplifier using triode technology.

The EL 84 T is a high-current driver stage in a compact design. The EL 84 T is a variant of the EL 84 . Both tubes are similar. The difference is that the suppressor grid of the EL 84 T is lead out separately, which offers additional options. If the EL 84 T is not available, the EL 84 can also be used, however, in the latter case it will work as a quasi-triode.

The ECC 832 is a useful input and phase inverting tube for this application. This double triode is built as preamplifier and a phase inverter for push-pull amplifiers. The ECC 832 consists of half an ECC 83 with high gain and low anode current and half an ECC 82 with low gain and high anode current, which provides phase inversion. In this way, just one tube is needed, solving two problems at once. The result is a compact and manageable set of tubes.

In the 1950s and 1960s, many devices were developed and built according to this design principle. An important aspect is that all the tubes used in this amplifier design as well as their sockets are still manufactured and will likely be available in the future!

Similar to the 300 B, most of the voltage gain for the 6C33 has to be generated in the preamplifier and the driver stage. The 300 B, for example, only has an open-loop gain factor of about 4. To fully drive the 6C33 and avoid premature limiting of the driver stage, the latter is supplied with a higher operating voltage than the power tubes. Due to the

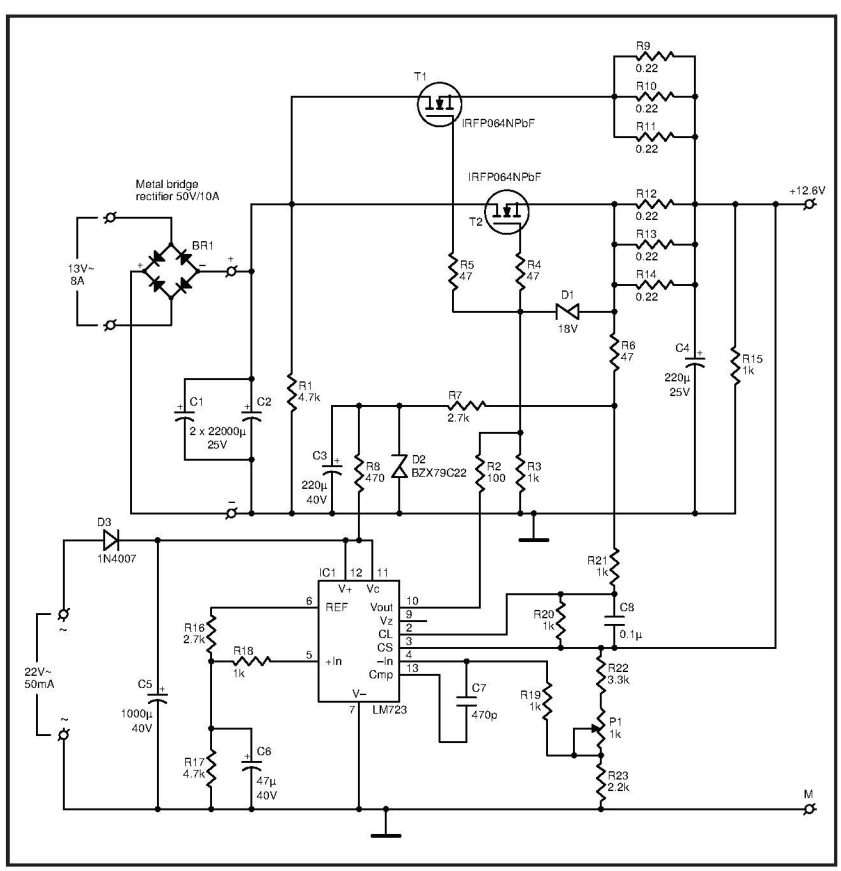


Figure 1: The heating circuit is a low-drop voltage controller.

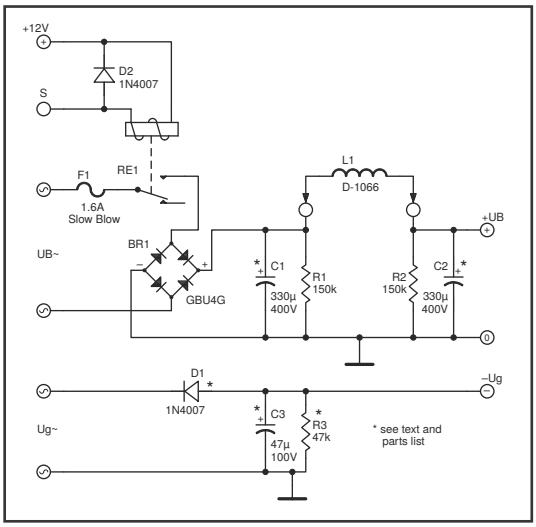


Figure 2: This is the CLC filter that I used with the 6C33 tube.

6C33's low operating voltage, the preamplifier and the driver stage only require moderate voltages. The advantage is that electrolytic capacitors can be used with common operating voltages without any cascading. The capacitances can be generously dimensioned, which is also beneficial for hum suppression as well as for clear reproduction of powerful basses.

When using the 6C33 tubes, be sure not to use the maximum permissible anode voltage. Safe operation is guaranteed for an anode voltage of about 200 V or slightly below. As can be seen in the build instructions, the operating voltage may be somewhat higher because there is a voltage drop at the transformer's copper resistance so that the upper anode voltage limit for a safe operation is maintained.

Even if you find circuits presented in Internet forums or other publications that work with higher operating voltages, it is better to opt for a lower operating voltage. As appealing as a higher voltage might be to get a higher power output, it is better to do without a few watts for increased reliability. Because the 6C33's manufacturing quality can be unpredictable, these tubes tend to fail at a higher operating voltages.

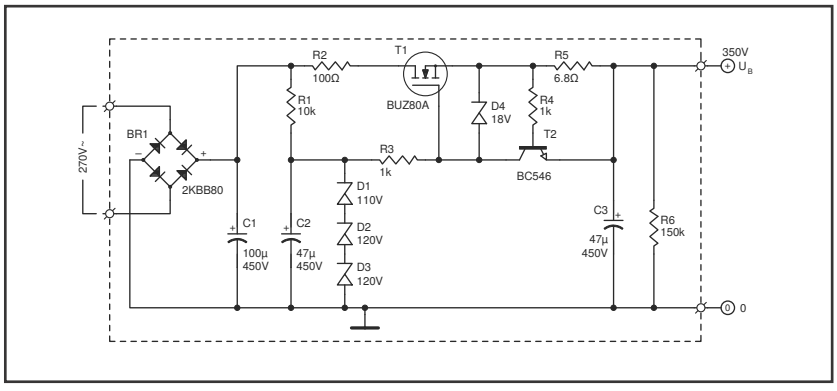


Figure 3: This circuit stabilizes the preamplifier and the driver stage.

Circuit Design

The circuit design presented here takes all these factors into consideration. Likewise, the controlled DC heating contributes to a good overall result. The tubes are always heated with constant voltage and a well-filtered current, which ensures good operating conditions.

Figure 1 shows the heating circuit, which is a low-drop voltage controller that generates little heat because of a relatively low power dissipation. However, the transistors T1 and T2 need to be mounted to a heatsink or to the housing for heat removal purposes. Likewise, the rectifier Rct1, which is not located on the circuit board, must be cooled. The design distributes the high voltage across two circuit boards.

Figure 2 shows a CLC filter used with the 6C33. On the circuit board, I have provided a relay that is operated by the heating voltage of 12.6 V. It can also be used to switch off the anode voltage of the power tubes in stand-by mode. The circuit board also has provisions for generating a negative bias by separate rectification and charging of an additional electrolytic capacitor, which is meant for the negative grid bias. Since these components are already present on the power amplifier board, D1, C3, and R3 can be dropped. The high voltage for the preamplifier and driver stage is stabilized by the circuit shown in **Figure 3**, which uses Zener diodes and a field-effect transistor (FET).

Figure 4 shows the entire amplifier circuit. The area marked with a dotted frame is located on one circuit board. Since there aren't any PCB sockets for the 6C33, the sockets must be directly screwed to the chassis and connected with short cables. Metal spacers must be installed between the chassis and the socket to ensure a better cooling air flow. Plastic spacers would melt because the 6C33 generates a lot of heat. The heating capacity at about 40 W and the anode power dissipation at more than 20 W per each tube should not be underestimated. The heat must be effectively dissipated! Due to the excessive heat generated by the 6C33, a mono design is recommended. The expense is greater, but thanks to the excellent channel separation of the monoblocks, the sound is considerably improved. It also lowers the device's temperature, which increases the life of the components.

The amplifier input features one XLR plug connection and a 1:1 input transformer for PCB mounting to be installed on a breadboard. This permits a floating balanced signal connection that

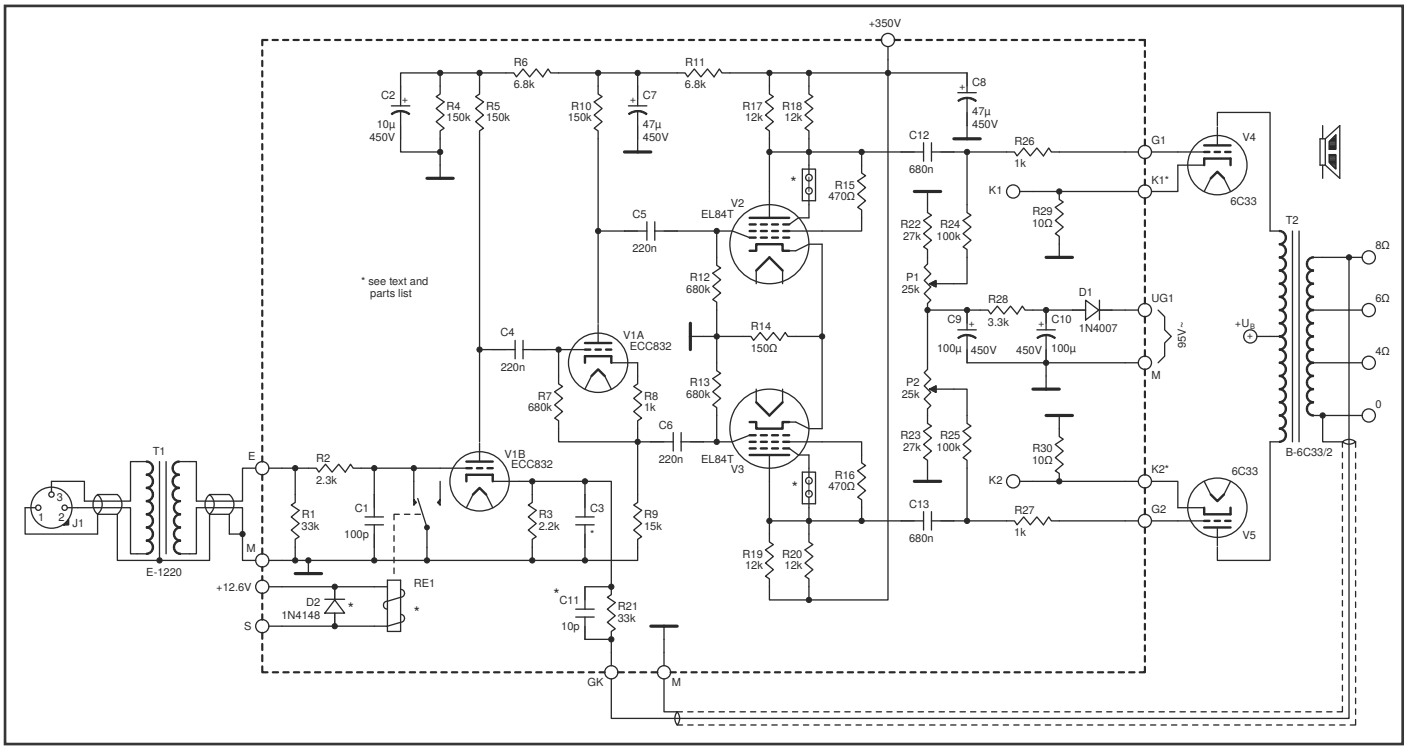


Figure 4: Here is the entire amplifier circuit.

safely avoids ground loops. The input lines need to be connected with two-core shielded cable as shown in **Figure 4**. This guarantees the most interference-free signal transmission. No coupling capacitor is needed between the transformer and the amplifier input, therefore, no sound modification can occur at this important point. The direct current path for the input tube grid would be closed by the transformer. However, it is critical that transformers are correctly terminated and loaded. Otherwise, it often leads to significant frequency response deviations. With 33 kΩ, the E-1220 type transformer is correctly terminated.

The preamplifier is connected free-of-ground via the XLR plug connection. R2 and C1 form a low-pass that keeps radio frequencies away from the amplifier input. The input is shorted over the relay contact. This allows for muting (e.g., during the warm-up phase), when switching off the amplifier or in stand-by operation. If a voltage is applied to the relay coil, which is taken from the DC heating, the input is enabled. If this function is not needed, RE1 and D2 are dropped.

ECC 832's Two Systems

As I stated, the input tube ECC 832 consists of two different systems. Half the ECC 83 provides the required amplification and half the ECC 82 is used for phase inversion. Since this system has high current levels, it can be wired with low resistances,

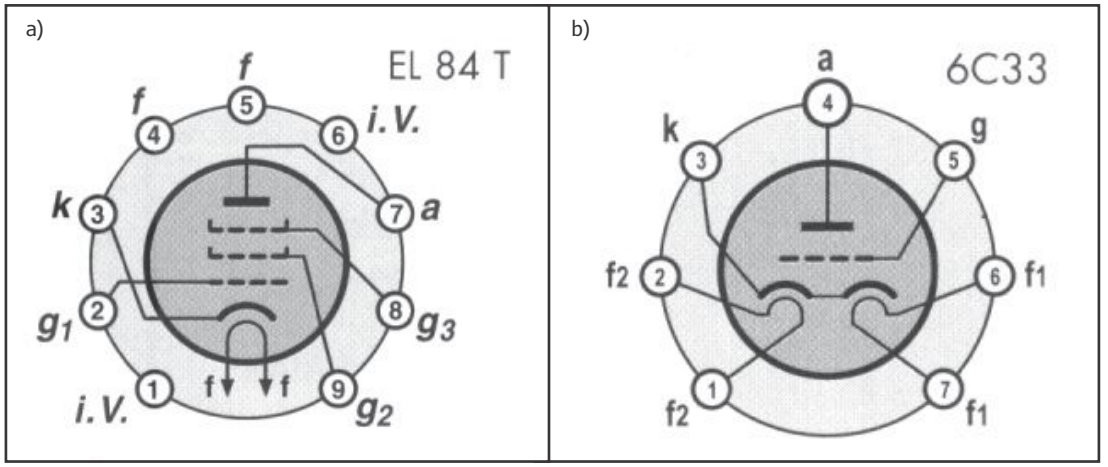


Figure 5: These are the socket pinouts for the EL 84 T (a) and the 6C33 (b) tubes

Parts List for Push-Pull Amp with 6C33 Tubes

Resistors, if not otherwise marked, metal film, 1 % tolerance, 0.6 W, MO = metal oxide 2 W, 2 % tolerance, 4 W metal oxide, 5 % tolerance

DC Heating (Figure 1)

Capacitors

C1, 2	22.000 µF/25 V
C3	220 µF/40 V
C4	220 µF/25 V
C5	1000 µF/40 V
C6	47 µF/40 V
C7	470 pF ceramic
C8	0.1 µF ceramic

Diodes

D1	1 N 4007
D2	BZX 79 C 22

Miscellaneous

Gl	metal bridge 10 A/50 V
1	socket DIL 14 gold-plated contacts

PCB copper laminated, 70 µm, 75 mm × 108 mm, PCB# E-212/833 B*

Op-Amps

IC1	723 DIL

Potentiometer

P1	trimpot 1 kΩ cermet

Resistors

R1	4.7 kΩ
R2	100 Ω
R3	1 kΩ
R4,5,6	47 Ω
R7	2.7 kΩ
R8	470 Ω
R9–14	0.22 Ω, metal foil,5 W
R15	1 kΩ
R16	2.7 kΩ
R17	4.7 kΩ
R18–21	1 kΩ
R22	3.3 kΩ
R23	2.2 kΩ

Transistors

T1, 2	IRFP064NPbF

****

High Voltage Supply for 6C33 (Figure 2)

Capacitors

C1, 2	330 µF/400 V or 470 µF/450 V
C3	not used in this application

Diodes

D1	not used in this application
D2	1 N 4007

Miscellaneous

Gl1	GBU4G
Si	1.6 A slow blow
Dr	filter choke D-1066 1 H/0.6 A (not on board)

PCB copper laminated 70 µm, 80 mm × 110 mm, PCB#EE-309 A*

Resistors

R1,2	150 kΩ MO
R3	not used in this application

****

High Voltage Supply for Driver Stage (Figure 3)

Capacitors

C1	100 µF/450 V
C2,3	47 µF/450 V

Diodes

D1	Zener diode 110 V, 1.3 W
D2,3	Zener diode 120 V, 1.3 W
D4	Zener diode 18 V, 1.3 W

Miscellaneous

Gl1	2KBB80 or B380C1500
1	heatsink SK 68/50 3.8 K/W

PCB copper laminated 70 µm, 80 mm × 100 mm, PCB# EE-252 B*

Resistors

R1	10 kΩ, MO
R2	100 Ω, 4 W
R3, 4	1 kΩ
R5	6.8 Ω, MO
R6	150 kΩ MO

Transistors

T1	BUZ 80 A
T2	BC 546

****

Driver Stage (Figure 4)

Capacitors

C1	100 pF, ceramic
C2	10 µF/450 V
C3	1 nF MKT (see text)
C4, 5, 6	0,22 µF/630 V MKP
C7, 8	47 µF/450 V
C9, 10	100 µF/160 V
C11	10 pF, ceramic (see text)
C12, 13	0.68 µF/630 V MKP

Diodes

D1	1 N 4007
D2	1 N 4148 (see text)

Resistors

R1	33 kΩ
R2, 3	2.2 kΩ
R4, 5	150 kΩ MO
R6	6.8 kΩ
R7	680 kΩ
R8	1 kΩ
R9, 10	15 kΩ, MO
R11	2.2 kΩ, MO
R12, 13	680 kΩ
R14	150 Ω, MO
R15, 16	470 Ω
R17–20	12 kΩ, 4 W
R21	33 kΩ
R22, 23	27 kΩ
R24, 25	100 kΩ
R26, 27	1 kΩ
R28	3.3 kΩ
R29, 30	10 Ω, MO

Miscellaneous

3	noval sockets, ceramic print mounting

PCB copper laminated 70 µm, 111 mm × 141 mm, PCB# EE-336 A*

Potentiometers

P1, 2	25 kΩ trimpot cermet

Relay

Rel1	Relay 12 V 1 x UM (see text)

Tubes

Rö1	ECC 832 vacuum tube
Rö2, 3	EL 84 T or EL 84 (see text)

****

Additional Parts

Transformers

1	Power transformer	NTR-24
1	Output transformer	B-6C33/3
1	Input transformer	E-1220

Tubes

Rö4, 5	6C33 with socket

**Editor's Note: The layouts for PCBs listed in the Parts List can be found in the Supplementary Materials section of the audioXpress website (see Project Files).*

which makes it less sensitive to disturbances. The two EL 84 T connected as triodes in the differential amplifier stage amplify the signal further. These tubes, too, have a very high current output and can be wired at low resistance. If the EL 84 T is used, the two wire jumpers for the third grid need to be mounted. If the EL 84 is used, the wire jumpers are dropped since the third grid is already connected to the cathode inside the tube. Due to the low-resistance circuitry the high-current driver tubes EL 84 (T) have many advantages.

Since the grid bleeder of the 6C33 must not exceed the value of 200 kΩ, it can be dimensioned for a relatively low resistance. Due to the variable manufacturing quality and the persistent tube tolerances, these can be more effectively controlled using a significantly lower impedance grid bleeder. A lower resistance wiring also requires larger coupling capacitances to make sure that the frequency response will not drop too early at low frequencies.

The negative grid bias of the 6C33 is generated in the usual manner. Since it is relatively high, electrolytic capacitors with a dielectric strength of 160 V must be used. The output transformer has taps for 4 Ω, 6 Ω, and 8 Ω.

There are many speakers that have an impedance around 6 Ω, possibly because speakers with 4 Ω or 8 Ω sound better at the 6-Ω tap. In this way, the user has many options. In case of maladjustment, the amplifier will not be damaged, only the power output will slightly fluctuate. The idle current is set using the trimpots P1 and P2. The voltage drop is measured via the cathode resistors. Between the points K1 or K2 and ground, a value of 1 V must be set, which corresponds to an idle current of 100 mA.

Due to the large and massive electrode, the tubes need to warm up for at least 15 minutes until stable conditions are reached. Before that, the idle current is clearly shifting. If the idle current is set to the correct value for the tubes, you will notice a significantly lower idle current than you would get when starting with the cold tubes. However, that is no reason to adjust it since the correct value is achieved only after the warm-up time. If the amplifier is used to listen to music, it will not have optimal sound during the warm-up phase.

Project Files

To download the PCB layouts, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html.

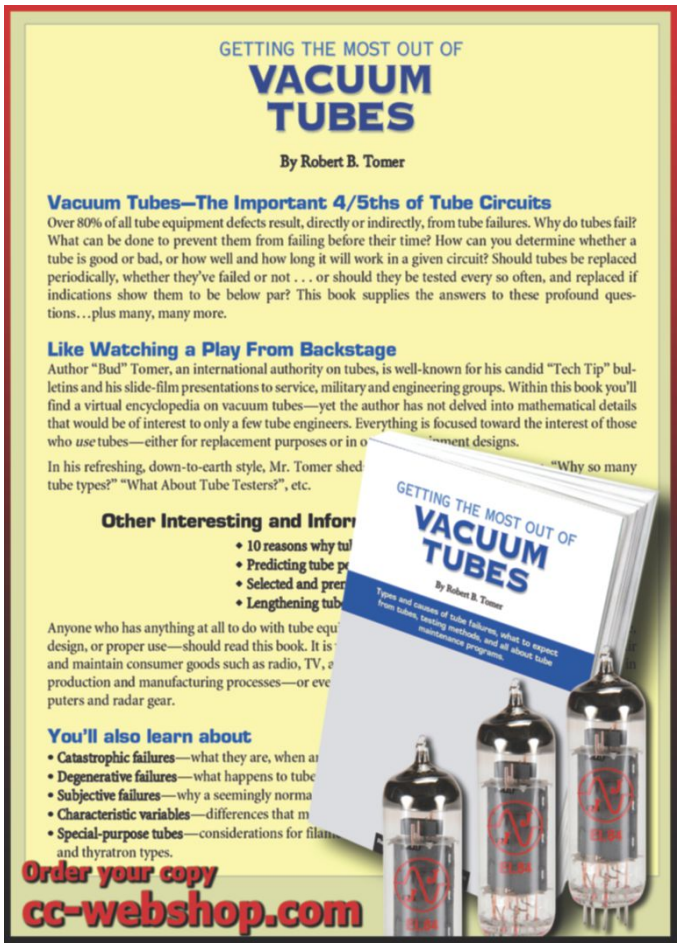
However, I would recommend that after a few minutes you play music at a moderate volume with the speakers connected. In this way, the voice coils of the speakers can also warm up, since about 90% to 95% of the applied amplifier power is converted into heat and the remainder is the desired sound.

The negative feedback line of the transformer output is transferred over a shielded cable (see **Figure 4**). To avoid hum, this cable should not be run in a cable harness with lines featuring mains frequency. C3 and C11 must be dimensioned according to the individual amplifier design. The values specified in **Figure 4** are reference values. C11 suppresses the high-frequency oscillating tendency. If there is acute oscillation, C11 must be enlarged otherwise it should be as small as possible. If during the frequency response verification, it should turn out that the drop toward high frequencies sets in too early, C3 must be used and should have a reference value of about 1 nF. This compensates for circuit and line capacitances. The sample unit has been measured without C3.

Figure 5 shows the socket pinouts of the EL 84 T and the 6C33 tubes. In case of the EL 84 T, note where the suppressor grid is led out. The thicker pin 4 is the anode connector of the 6C33. Both pinouts are shown as they would be seen from below. With the amplifier concept presented here, the user obtains an unconventional hi-fi component. The visual appearance of the 6C33 and the selection of the tube set as well as the pure triode concept have their own special appeal. In this case, triode technology using tubes that will be available in the long term shows its advantages.

About the Author

Gerhard Haas has written books as well as articles for Elektor Germany since 1995. The *Roehren Sonderheft* magazine was his creation, along with some partners. The first issue came out in 2005. Now, there are 10 issues containing a collection of interesting articles. Gerhard has also developed many electronic circuits and equipment. In the last two decades, he primarily worked on tube amplifiers and the circuitry around them, including the audio transformers, filter chokes, power transformers. He sold his company, Experience Electronics, to focus on designing tube amps and audio transformers and write articles for *Sonderheft*. He recently finished an updated version of his book *High-End Mit Roehren* (originally printed in 1995). The new book, also titled *High-End Mit Roehren,* is available in German from Elektor.





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.

May 2017

May 4–6
Consumer Electronics China (CE China) 2017
Shenzhen, China
www.ifa-berlin.com
http://www.cechina-ifa.com/en

May 5–7
Lone Star Audio Fest 2017
The Embassy Suites Dallas Park Central, Dallas, TX
www.lonestaraudiofest.com

May 18–21
HIGH END 2017
MOC Munich-Germany
www.highendsociety.de

May 20–22
142nd AES Convention Berlin 2017
Maritim Hotel Berlin, Berlin, Germany
www.aes.org/events/142

June 2017

June 7–9
CES Asia
Shanghai, China
http://www.cesasia.cn

June 15–17
2017 AES International Conference on Audio Forensics
Arlington VA
www.aes.org/conferences/2017/forensics

June 10–16
InfoComm 2017
WEST building, Orange County Convention Center (OCCC), Halls A-E, Orlando, FL
www.infocommshow.org

June 22–24
2017 AES International Conference on Semantic Audio
Erlangen, Germany
www.aes.org/conferences/2017/semantic/

June 25–29
Acoustics '17 Boston
173rd Meeting of the Acoustical Society of America and
the 8th Forum Acusticum
acousticalsociety.org

July 2017

July 13–15
Summer NAMM
Nashville, Tennessee
www.namm.org/summer/2017

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

August 2017

August 11–13
2017 Hong Kong High End Audio Visual Show
Hong Kong Convention & Exhibition Centre (HKCEC)
www.audiotechnique.com/avshow/2017/

August 21–24
SET EXPO 2017
International Technology Fair on Equipment and Services for Television Engineering, Broadcasting and Telecommunications
Expo Norte, São Paulo, Brazil
http://www.setexpo.com.br

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

September 2017

September 1–6
IFA Berlin (The International Funkausstellung)
www.ifa-berlin.com

September 5–9
CEDIA 2017
San Diego Convention Center, San Diego, CA
http://expo.cedia.net

September 8–10
AES 2017 International Conference on Automotive Audio
Burlingame, San Francisco, CA
http://www.aes.org/conferences/2017/automotive

September 12–14
GSMA Mobile World Congress Americas
Moscone Center, San Francisco, CA
www.mwcamericas.com
www.gsma.com | www.ctia.org

September 14–19
IBC 2017
RAI Amsterdam, The Netherlands
www.ibc.org

September 14–16
Prolight+Sound NAMM Russia
Sokolniki Exhibition and Convention Centre, Moscow, Russia
www.namm-musikmesse.ru | www.prolight-namm.ru

September 17–19
PLASA Show London
London, Olympia, United Kingdom
www.plasashow.com

September 21–24
T.H.E. Show Anaheim
Hilton Anaheim, 777 Convention Way, Anaheim, CA 92802

TBC
CanJam Europe 2017
Congress Center Essen, Germany
http://www.canjam.de

October 2017

October 6–8
Rocky Mountain Audio Fest (RMAF)
Marriott Denver Tech Center, Denver, CO
www.audiofest.net

October 13–16
HKTDC Hong Kong Electronics Fair (Autumn Edition)
Global Sources Electronics Show
electronicAsia |
www.hktdc.com/hkelectronicsfairae

October 16–18
The Loudspeaker Sourcing Show 2017
Nansha Grand Hotel in Guangzhou, China
www.loudspeakersourcingshow.com

October 18–21
AES New York 2017
143rd International Convention
http://www.aes.org/events/143/

October 11–14
Prolight+Sound Shanghai
Shanghai, China
www.prolightsound-shanghai.com

October 17–19
Prolight+Sound Middle East
Dubai International Convention and Exhibition Centre
www.messefrankfurt.com
www.prolightsoundme.com
www.prolight-sound.com/worldwide

November 2017

November 3–5
Capital Audiofest
Hilton Hotel at Twinbrook Metro, 1750 Rockville Pike, Rockville, MD
www.capitalaudiofest.com

November 10–12
New York Audio Show
Park Lane Hotel, Central Park South, New York
www.chestergroup.org/newyorkaudioshow/2017

November 13–19
Live Design International (LDI) Show
Las Vegas Convention Center, Las Vegas, NV
www.ldishow.com

November 15–17
Inter BEE
Inter BEE 2017 | 53rd International Broadcast Equipment Exhibition
Makuhari Messe, 2-1, Nakase, Mihama-ku, Chiba City, Chiba Prefecture 261-0023, Japan
www.inter-bee.com

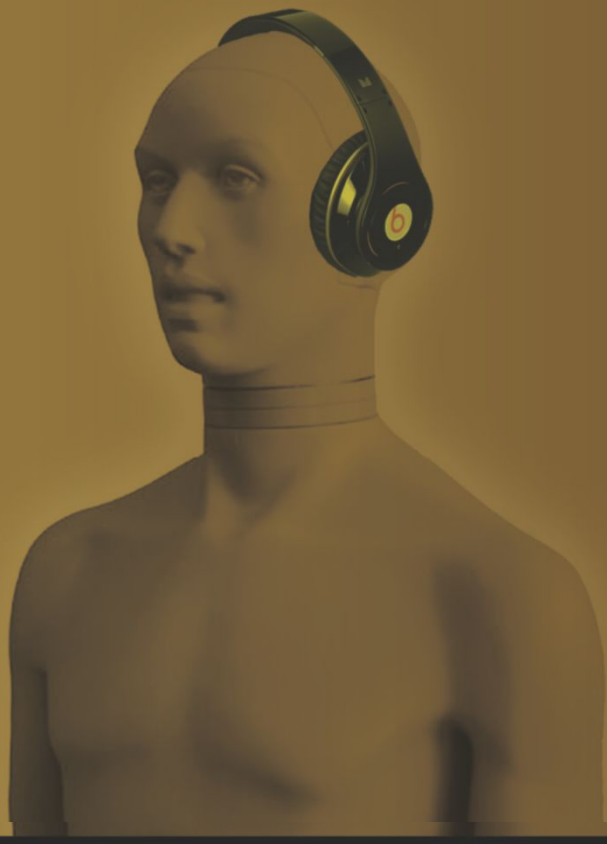
December 2017

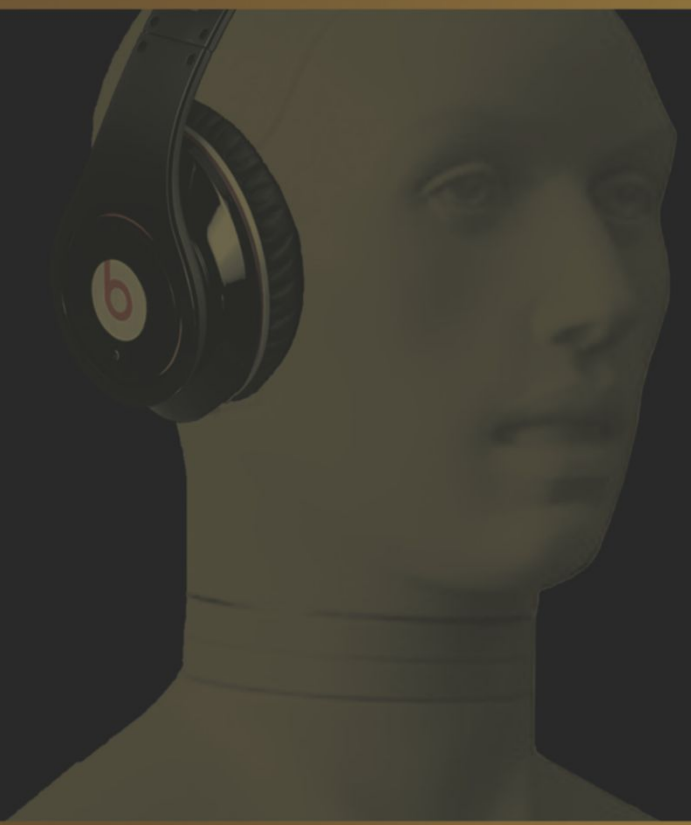
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