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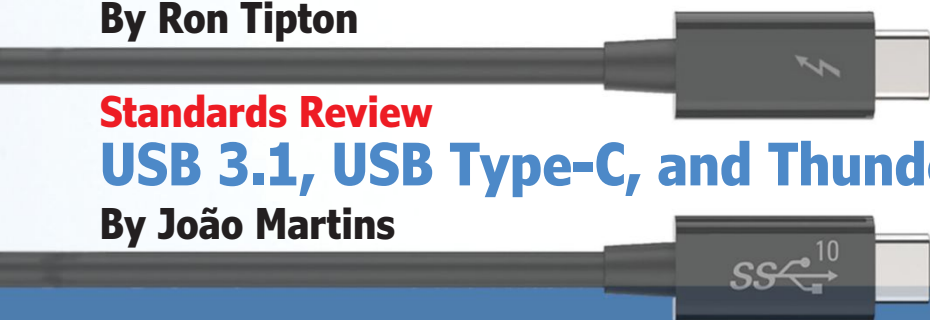
By Ron Tipton



Standards Review

USB 3.1, USB Type-C, and Thunderbolt 3

By João Martins



**You Can DIY!
A Retro-Style
High-Fidelity Preamp**
By Bill Christie

**Audio Electronics
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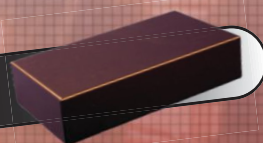


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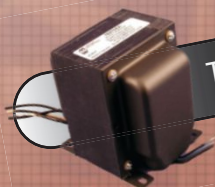
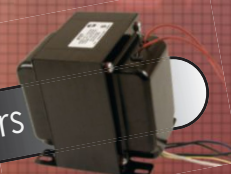


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Yes...It's Complicated.



Audio designers are increasingly confronted with complex technologies that weren't really invented with audio in mind. In this digital age, technologies that were specified for computers and communications, and increasingly molded to the needs of mobile devices, now define "universal" specifications for everything, including audio. So, when one is confronted with the whole debate about the removal of the 3.5 mm analog audio jack from new-generation smartphones, the question is not connected at all with the limited perspective of the audio industry. (If it was, I guess we would be debating the removal of the 1/4" analog jack or an XLR.) In fact, the professional audio community hates the mini-jack and would have killed it long ago if the IT, telecom, and mobile industries hadn't promoted it for the last two decades.

But now, those same industries have moved on and are committed to complete the transition to digital. As we start to look more closely at the recent USB 3.1 and USB Type-C specification updates—an agreement over which not even the promoting companies are still fully aware of the implications—it becomes clear that the times ahead will be challenging for audio developers. In this month's Standards Review article, I share some of the basic facts about those specifications and many of those challenges become more obvious. And we didn't even start to dig deeper into the new USB Audio Device Class 3.0 for Digital Audio Applications over USB Type-C, which was announced by the USB Implementers Forum (USB-IF) just last September.

As the newly published documents state, the new USB ADC 3.0 defines a versatile bus specification with many ways to propagate and/or control digital audio, with "well defined and standardized" audio transport mechanisms (to keep software drivers as generic as possible), with a robust synchronization scheme on isochronous transfers, and class-specific descriptors to handle all possible applications, audio formats, and compression schemes with extensions to support various changes in the core specification. The new USB ADC 3.0 standard is also intended to make connections with digital USB-C headphones easier, featuring special multi-function processing units (MPUs), to handle synchronization, D/A conversion, and all other device specific functions—on the headphones and not located inside source devices.

As the USB-IF also states, the USB Audio Device Class 3.0 specification intends to establish USB Audio over USB Type-C as "the primary solution for all digital audio applications, including headsets, mobile devices, docking stations, gaming setups, and VR solutions." You get a sense of priorities here.

It's now clear that all digital high-quality audio applications (including professional audio) will have to deal with a series of specifications determined by the mobile, telecom, and IT industries, increasingly using the Consumer Electronics channels as a marketing test bed. And that's why the analog jack is being replaced by a completely new connector—USB Type-C—about which the audio industry had no input.

Yes, connectors are important because they are the main interface over which consumers deal with technology—a kind of visual icon on what things are intended. Converting everything to a single USB Type-C connector started as a great idea—a way to simplify things and avoid the current coexistence of multiple incompatible cables that basically serve the same purpose. Nevertheless, in my opinion, the idea of using a single connector for many different applications in multiple types of devices will also generate a lot of unnecessary confusion among consumers. Far more problems than the benefits believed to come out of the "universal" design.

And do you know what the USB-IF calls to the basic (mandated) digital audio profiles on the USB Audio Device Class 3.0 specification, which are supposed to make things simpler? They're called Basic Audio Device Definition (BADD)! Yes...

João Martins
Editor-in-Chief

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Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—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.

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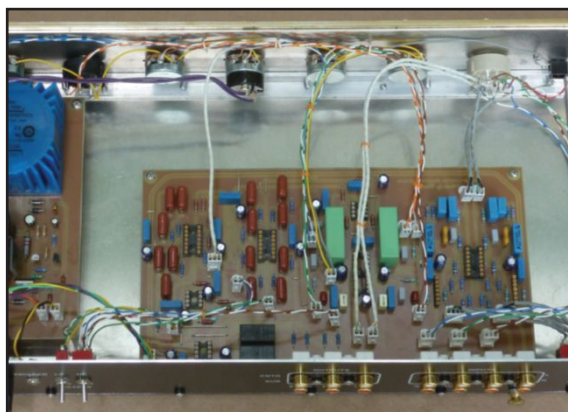


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● Features

8 Experience Disk Recording 1930 to 1960

By James Wood

Unearth this lost archive and explore a bygone era of early recordings from the "golden years of audio" and personal stories told by the late Robert Morrison.

10 Room Correction Revisited (Part 6)

miniDSP DDRC-24

By Ron Tipton

This month we investigate a new compact unit that is now the world's smallest Dirac Live hardware processor.

32 Innovative Measurement Microphones from PCB Piezotronics

By Stuart Yaniger

Learn more about four measurement microphones from PCB Piezotronics' professional line.

40 Mytek Digital Brooklyn DAC Preamplifier and Headphone Amplifier Born in New York

By Oliver A. Masciarotte

Discover Mytek's Brooklyn, an exceptional device capable of decoding all the modern formats as well as one so new that only a handful of DACs provide support.

46 Retro-Style High-Fidelity Preamplifier

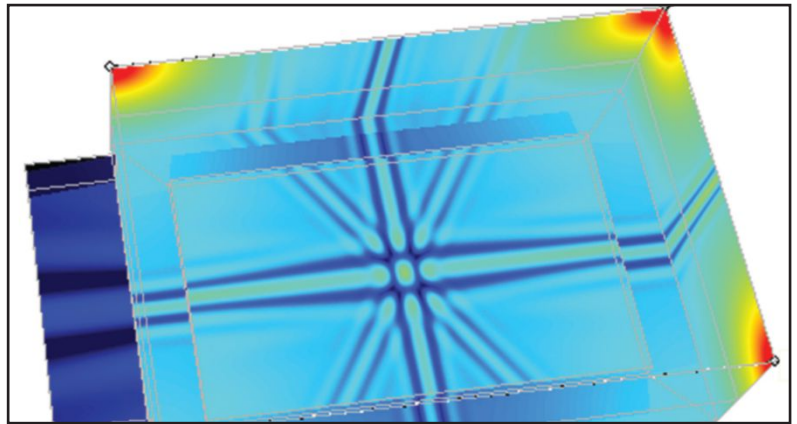
By Bill Christie

Build a full-featured preamplifier with volume, balance, and tone controls.

52 Optocouplers in a MOSFET Power Amplifier Design

By Richard K. Mains

Explore a novel method of temperature compensation in audio power amplifiers that utilize MOSFETs as the output devices.



● Columns

STANDARDS REVIEW

16 USB 3.1, USB Type-C, and Thunderbolt 3 (Part 1)

An Introduction to the Latest Interfaces

By João Martins

SOUND CONTROL

26 A Complementarity Principle in Acoustics

By Richard Honeycutt

HOLLOW-STATE ELECTRONICS

60 Tremolo Circuits in Guitar Amplifiers

By Richard Honeycutt

● Departments

4 From the Editor's Desk

5 Client Index

66 Industry Calendar

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Experience Disk Recording 1930 to 1960

By
James Wood
(United States)



For Robert "Bob" Morrison, the golden years of audio were those dominated by the disc, and his love for disc recording and vintage audio equipment kept him occupied with research and experimentation in this arcane, historical area of the audio industry.

A lost treasure from the past has been resurrected, detailing a bygone era of early recordings, and it is definitely worth a look. The book, *Disk Recording 1930 to 1960: Equipment, Techniques, Recollections* was written by Robert K. Morrison. Morrison said he wrote the book "to salvage for historians and hobbyists some scarce documentation and a view of the period which produced it."

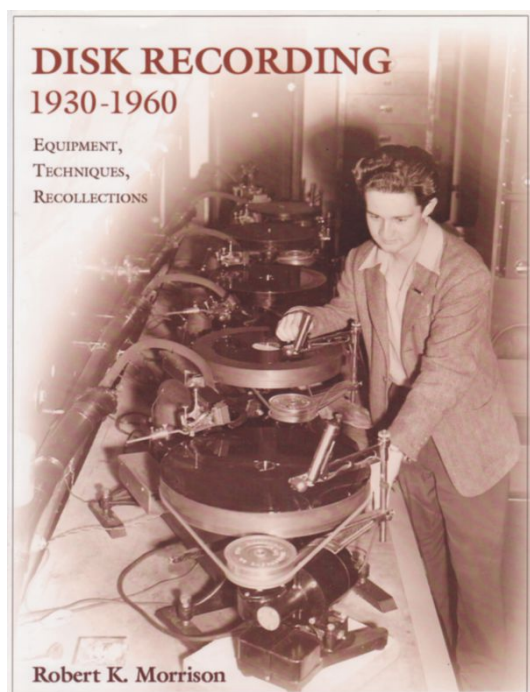
Morrison continued, "Friends with whom I have discussed this project insist that some background introduction before the 'nuts-and-bolts' sections is desirable—both to provide context for the equipment descriptions, and also to convey a sense of the improvisational character of the art in those days." In the telling, Morrison shares a rare look at a three-decade span that is fast fading from memory.

An inscription inside the front cover of this book claims, it is "A review of some equipment and techniques used in making phonograph records and electrical transcriptions." This is a very modest introduction by the author, the greater part of whose life was devoted to sound recording systems and practices. Morrison is probably best known as "The Voice of Ampex," having personally announced test tone frequencies and other information on analog alignment tapes from Ampex, and subsequent to his retirement, tapes from his own Standard Tape Laboratory.

According to Morrison: "My experience with disk recording began at a very early age, and therefore, covered a longer period than that of many of my friends in the field. Over that time, I collected a considerable variety of pictures, diagrams, and catalogs pertaining to recording in that era. A number of friends have encouraged me to make this pictorial material available, along with explanatory text, to help provide a view of mechanical sound recording from the twilight of acoustical recording through the advent of the LP record. Disk recording during those years was pretty much of an art, and there was plenty of room for experimentation and individual initiative. I hope this booklet will help convey some of the flavor of those days."

Morrison was able to provide first-hand accounts of those days because he entered the industry at the start of World War II, when he was only 16. His first job was at Picto Sound Productions, a San Francisco, CA, film and audio studio. Among his many duties, he manned a bank of five Universal disc-recording lathes, delaying network broadcasts, preparing transcriptions for Armed Forces Radio, and recording hours of podium audio from a fledgling United Nations.

Despite his gravitation toward the "hands-on" aspects of sound recording, Morrison earned a BA degree from the University of California at Berkeley. In the early 1950s, he worked as a recording engineer and staff announcer with Voice of



This book was first published in 2003 as a bound version. In the period following Robert Morrison's death later that same year, all unsold stock was lost or destroyed.

America in New York. Morrison then returned to the Bay Area to assume the positions of Chief Engineer and a primary air talent with KRE Radio in Berkeley. During this same period, Morrison worked on contract with Ampex Corp in Redwood City, CA, and ultimately assumed full-time management of its alignment tape facility in 1960. He founded his own firm, Standard Tape Laboratory, in 1969, and continued to manufacture test tapes and magnetic films into the 1990s. However, Morrison's initial experience and his longtime love affair was with disc recording. His book chronicles a period in disc recording history with the benefit of Morrison's hands-on familiarity with that technology through this major period of its innovation and advancement.

The book is packed with solid technical information, sprinkled with his anecdotal recollections, and profusely illustrated with photographs, drawings, and reprints of sales literature from a bygone era. "Seasoned" readers who recall the names: Scully, Audiodisc, Presto, Capps, and Rek-O-Kut will have their memories refreshed, and younger readers may well find themselves in awe of a quality sound recording



Photo 3: Bob Morrison's widow, Patricia, has given permission to reprint the book in digital form. It is being offered as a high-resolution PDF file on a CD-ROM.

process that was basically electro-mechanical.

This book was first published in 2003 as a bound version. Sadly, in the period following Morrison's death, all unsold stock was lost or destroyed. His widow Patricia is the copyright holder. She has given permission to reprint the book in digital form. It is being offered as a high-resolution PDF file on a CD-ROM at a price of \$15, plus \$2 for domestic US postage and handling. All proceeds benefit Patricia Morrison. For more information or to purchase a copy, contact the author James Wood (Inovonics, Inc.) at jim@inovonicsbroadcast.com. 

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Room Correction Revisited (Part 6)

miniDSP DDRC-24



Photo 1: The compact DDRC-24 has its signal inputs and IR window on the front. (Image courtesy of miniDSP)

In July 2016, miniDSP introduced the DDRC-24, a USB DAC+DSP Digital Room Correction processor powered by Dirac Live. This is now the world's smallest Dirac Live hardware processor, making room correction available in a compact hardware unit.

By
Ron Tipton
(United States)

The DDRC-24 is a joint project between miniDSP and Dirac Live (see **Photo 1**). The DDRC-24, available for Windows and Mac, is a versatile stereo processing module. **Figure 1** shows its the block diagram. Among its other uses, which I'll describe later in the article, it becomes a stereo "room equalizer" when programmed with a special version of Dirac Live that makes room measurements. I was fortunate to receive a Windows evaluation unit.

DDRC-24 Setup

Although it is explained in the User Guide, I want to emphasize that the setup steps must be performed in the correct order or it doesn't work properly. The DDRC-24 must be set up for the matrix mixer and the output channel processing before

running a Dirac Live room measurement because the measurements are run through the DDRC-24. That is, the stereo cable to your preamplifier, the power amplifiers, and the speakers is connected to the DDRC-24 output number 1 for the left channel and output number 2 for the right channel (see **Photo 2**).

I'll briefly describe these steps, but I want to emphasize another point or two. First, you measure your room response, set up the target curve, and generate the correction filters. Then you program the DDRC-24 with your stereo filter. While in this screen, be sure to turn the filter On, and set the Output volume level.

At this point, you have a standalone correction module that does not need to be attached to your computer. But wait a moment! The module defaults to analog input (the RCA connectors), the first (and maybe only) filter, and by always having the filtering ON, there is no way to do an On/Bypass comparison. But if this is all you need, then you are done.

However, to be able to select one of the other inputs, another one of the possible four filters, or have a bypass, you need to program a TV IR remote control unit (see **Figure 2**). Units from Apple, NEC, Sony, and Philips RC6 will work. (However, a Sony DVD player remote didn't work. I tried it.) On the DDRC-24 main screen, click IR Remote and then Learn. You will get the screen shown in **Photo 3** which has a list of the programmable items.



Photo 2: The four output RCA connectors and DC power input are on the other end. (Image courtesy of miniDSP)

Click a Learn button and then press the remote control button you want for that function. After a few seconds, you should get a Successful message and a check mark will appear. To change your mind for a reprogram, click the Learn button and let it time out. The check mark will be removed. Try to pick easy-to-remember remote control buttons or maybe make yourself a list until you get familiar with your choices.

Now you are ready to disconnect the computer and put the module in your audio system. However, this presents another problem. The module is lightweight and the IR window is on the audio input side. The weight of the audio cables can make it difficult to keep the IR window pointed toward your listening position.

I solved this with the setup shown in **Photo 4**. An 1/8" thick aluminum bar, a felt strip, and some threaded spacers clamp the module to the 0.25" thick aluminum base plate. Nylon cable placed through holes in the plate manage all the cables toward the rear and four stick-on rubber bumpers complete the assembly. A bit crude, perhaps, but effective.

It may seem that I've written this article backward, but I did so to make a point. If you begin by knowing what needs to be done to properly complete a fully usable module, it saves time and a bit of aggravation.

Initially, I did not understand that I really needed the remote control. I had to remove the module from my audio system to do the programming and then put it back. Not greatly inconvenient, but still... a nuisance. Now it's time to get back to the beginning.

Installing the DDRC-24

The set up and activation process is described in the DDRC-24's User Guide but it's necessary to do each step in the correct order to avoid having to start over. I'll go through the important points, which should be kept in mind when you read the User Guide. I recommend reading the User Guide and I have included it in the Supplementary Material found on the audioXpress website (www.audioxpress.com). Then, visit the miniDSP website (www.minidsp.com) to download the DDRC-24 software. Enter your user name and password to go to Support > User Downloads. Scroll down to DDRC-24 and download the zip file. Connect the DDRC-24 to your computer with the supplied USB cable and wall power supply. Unzip your download into a convenient folder.

For Windows, go to Section 3.1 in the User Guide and follow all the steps in exactly this order. (It's Section 3.2 for Mac.) Then go to Section 3.4 to

activate your Dirac Live license. I have found it necessary to stay connected to the Internet while using this software. It seems to check periodically to make sure you have a valid license.

To configure the DDRC-24, go to Section 4. I used the simple, stereo room correction described in Section 4.3.

Room Measurements

The measurement part of this special version of

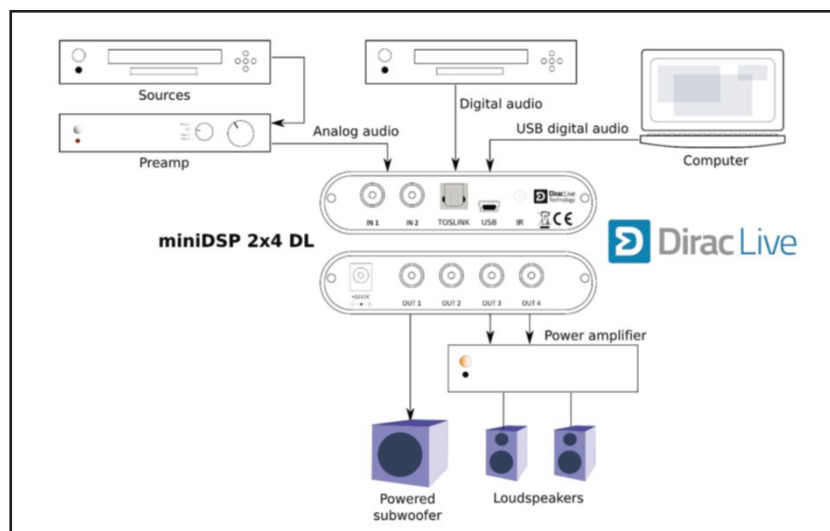


Figure 1: This is the DDRC-24 block diagram shown in the User Guide.

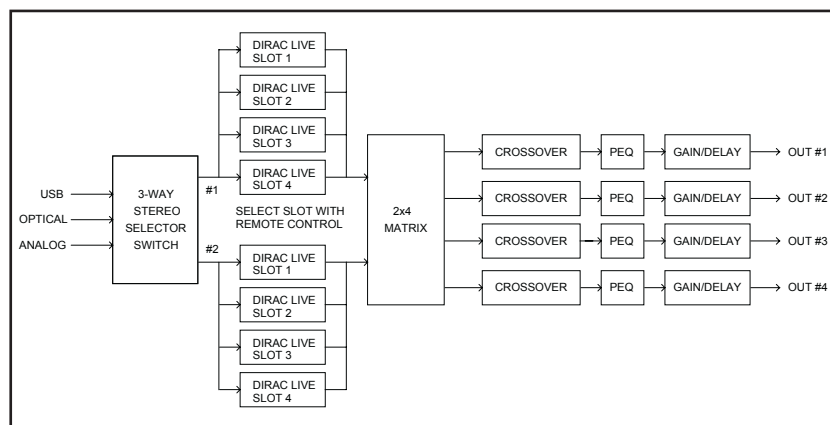


Figure 2: This DDRC-24 block diagram is different from the diagram in the User Guide because I have shown the four Dirac Live slots that can be programmed with stereo filters and then selected with your remote control. The input selection defaults to Analog but can be changed with the remote control.

About the Author

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.

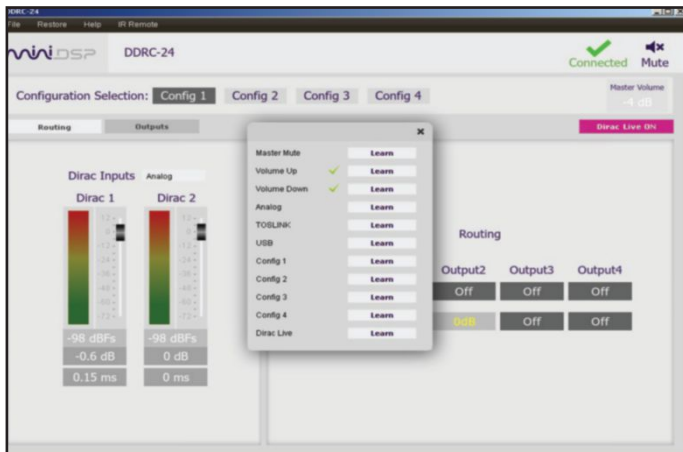


Photo 3: The initial DDRC-24 setup screen lets you configure one or more of the four filter slots and program your remote control. Before leaving this screen, press the Connect button in the upper right corner and make sure you get the Connected message.

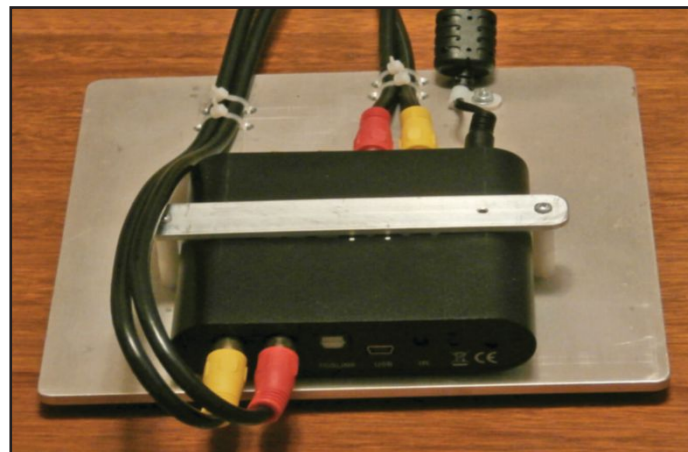


Photo 4: The DDRC-24 module is lightweight so I built this assembly to help keep the IR window pointed toward my chair with the cables routed to the rear. I used a pair of 12" stereo cables with couplers to longer cables so I wouldn't have to cut and redo the nylon ties.

Dirac Live is the same as the standard Dirac Live software I reviewed in "Room Correction Revisited: Part 5" (*audioXpress*, November 2016). I will briefly cover the steps again and there is a help section on the right side of each screen. For more detail, please see Dirac Live's User Guide or my previous article.

Launch Dirac Live. The first screen (see **Photo 5**) should initialize and then automatically display Stereo Sound System and DDRC-24 as the playback device. If it doesn't, quit Dirac Live, run DDRC-24 and make sure Connected is displayed in the upper right corner. Back to Dirac Live and, when all is well, click Proceed to continue.

The next screen does your microphone setup. Apparently, you must use a miniDSP UMIC-1 USB microphone for correct results. Each one is serial numbered and you can download its calibration file from

the miniDSP website. Click Proceed to continue.

Next, set up the test signal levels as shown in User Guide Section 5.3.3. Click the Test button. Adjust the Input gain and Output volume for a signal level in the middle of the "green." Click Proceed.

Nine measurements around your chair or sofa are required. Choose the seating you have, you will get a screen to select either the top, front or oblique view with the nine microphone positions. I found it helpful to use a screen capture program to print a copy of all three views. They are useful to refer to when placing the microphone. Be sure to save your project before proceeding.

Photo 6 shows the averaged left and right

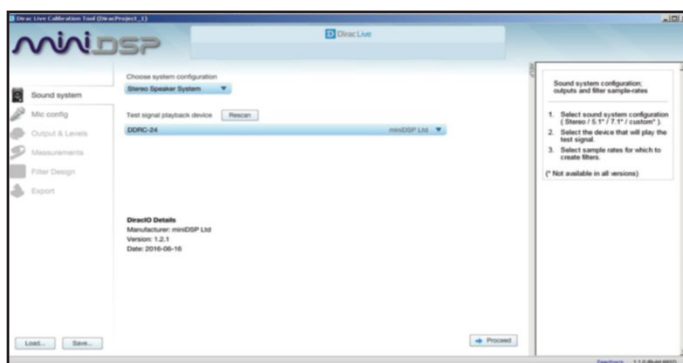


Photo 5: This is the initial Dirac Live screen. It should automatically initialize and display Stereo Speaker System and DDRC-24 as the Test signal playback device. I suggest being connected to the Internet when you launch Dirac Live because it may check your license.



Photo 6: When all nine measurements are completed, this screen displays the averaged left and right responses with a default target curve. Their target has more tilt than I like so I grab the orange dot control points and flatten it some. You can add additional control points by clicking the orange target curve.

Sound thinking in coaxial design

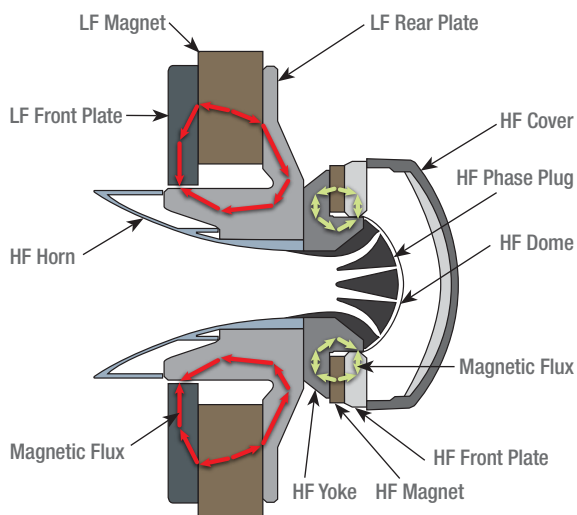
How a common magnet motor delivers a better sounding, lightweight, full-range speaker

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

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

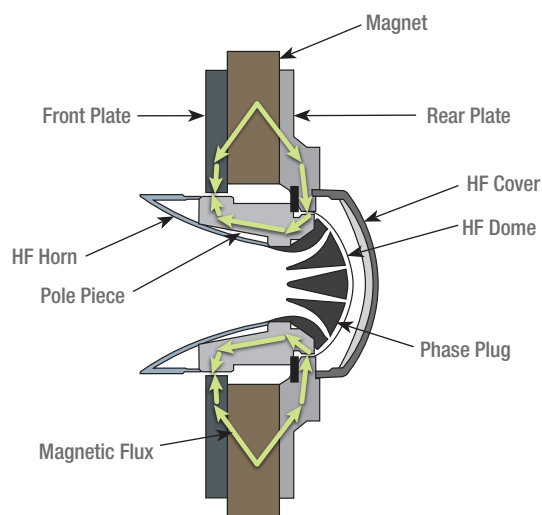
The magnetic flux is optimally distributed between the two elements (calculated using FEA), to achieve the best balance between HF and LF response (see diagrams below). Additionally, the use of a single magnet assembly also means lighter weight and a more compact profile, compared to more conventional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

Specify the innovative choice - Celestion Pro Audio Drivers. See our full range at celestion.com



Traditional Coaxial Driver

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



Celestion Common Magnet Motor Design

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

Find out more



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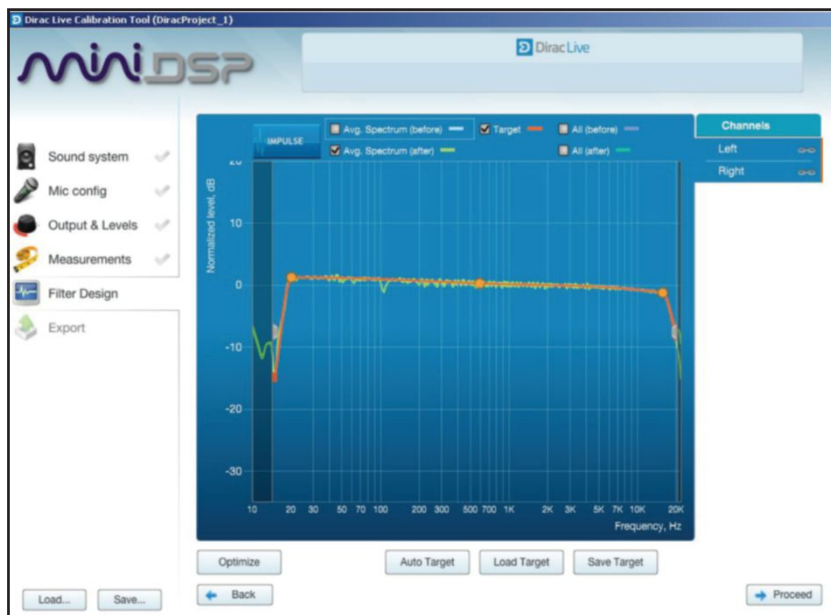


Photo 7: The predicted filter's amplitude response is shown with the target. If you are satisfied with it, press the Optimize bar. This generates the stereo filter. If you are not satisfied, click the Back button and readjust your target.

measurements for my listening room. Also shown is my adjusted target curve. The default curve has several decibels more tilt, but I prefer a flatter response. See User Guide Section 6.2. When you are satisfied with your target, click Optimize. This generates the stereo filter and displays the target with the predicted response (see **Photo 7**). Click the Export tab.

If this is your first filter, the next screen shows a blue square labeled Dirac Project 1, unless you renamed it, and four empty slots. Grab the blue square and drag it into Slot 1. This programs the DDRC-24 and displays the image shown in **Photo 8**. Before exiting, be sure to turn the filter On, and adjust the Output volume slider, if you don't want to accept the default. Now, you are ready to do some stereo listening.



Photo 8: Drag the blue square with the project name into Slot 1 to program the DDRC-24. I captured this screen after the programming so Slot 1 displays Left and Right as the channel names. Remember to turn the filter On, and to set the Output volume before exiting.

Testing the Results

I listened to several CDs processed through the DDRC-24 and was impressed by the improved stereo imaging and clarity. But, I still wanted quantitative results so I used Room EQ Wizard (REW) to measure the room's acoustics. **Figure 3** shows the averaged left and right frequency responses with the target. The slight downward tilt from low to high frequency is still apparent. Although not as smooth as predicted, they are greatly improved over the measured curves shown in **Photo 6**.

In **Figure 4**, we see the left RT60 reverberation time with the tolerance band relative to the arithmetic average value for a small room (refer to the AESTD1001.pdf included in the Supplementary Material). The right channel RT60 is similar. The Dirac Live filters use mixed-phase correction for an optimally predicted time-domain correction. I can't explain the peak at around 400 Hz, but I ignored it because all my listening tests were pleasant.

The Versatile DDRC-24

Connect your analog or digital output music sources to the module and switch inputs with your remote control. They all play through the stereo filter to improve your listening experience. You can have four different filters, to correct different music moods to fit your preference. If you refer to **Figure 2**, you will notice the sets of three unused

Project Files

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

Resources

"Multichannel Surround Sound Systems and Operations," subcommittee of the Audio Engineering Society (AES) Technical Committee on Multichannel and Binaural Audio Technology, www.aes.org/technical/documents/AESTD1001.pdf.

R. Tipton, "Room Correction Revisited (Part 5): Dirac Live," *audioXpress*, November 2016.

Sources

Dirac Live

Dirac Research AB | www.dirac.com

miniDSP

miniDSP, Ltd. | www.minidsp.com
(Get the special version of the Dirac Live software from the miniDSP website and not from Dirac Live)

Room EQ Wizard (REW)

Room EQ Wizard | www.rew.com

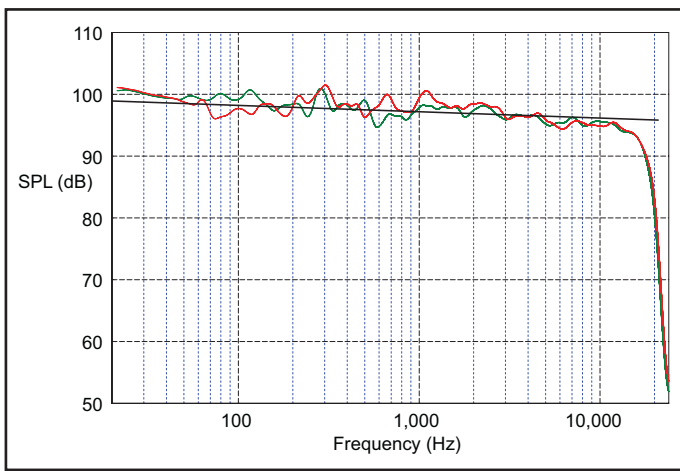


Figure 3: The averaged left and right DDRC-24 frequency responses as measured with Room EQ Wizard are shown with the target curve. Four measurements were made for each channel, 1/6 octave smoothing was applied to each measurement, and then they were averaged. The green line is the left channel and red is the right channel.

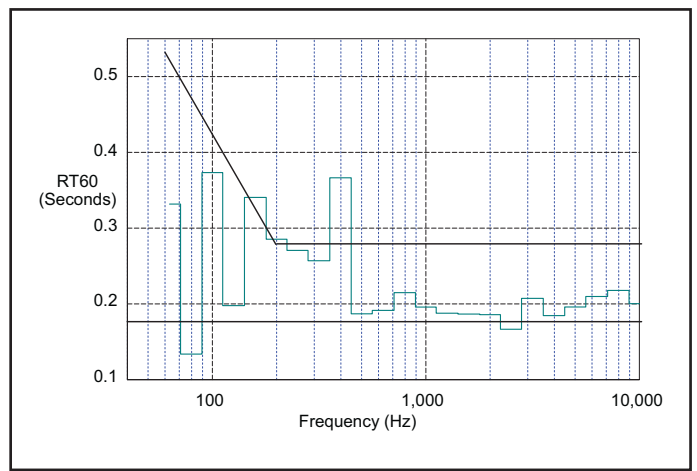


Figure 4: Shown here is the left channel reverberation time, RT60, along with the tolerance band for a small room. The peak at about 400 Hz does not seem to adversely affect the music playback.

blocks for each output. You can design and add crossover networks for two-way stereo speakers or to add a subwoofer. Gain and time delay are programmable to equalize the stereo speakers' sound if they are of different designs or not at the same distance from your listening position. You can also skip using the Dirac Live filters. Instead, use REW or another measurement program that can generate bi-quad filters to program the parametric equation (PEQ)

blocks. All the features are described in the DDRC-24 User Guide.

Final Words

The DDRC-24 and the UMIC-1 package is currently priced at \$525 plus shipping from Hong Kong. Although not inexpensive, it is one of the best ways I've found to listen to excellently processed stereo music from all my sources. 

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USB 3.1, USB Type-C, and Thunderbolt 3 (Part 1)

An Introduction to the Latest Interfaces



This article is intended as an introduction to the new SuperSpeed USB 10 Gbps (USB 3.1 Gen 2) interface, and the new USB Type-C connector specification, including the recently published USB Audio Device Class 3.0 specification, which standardizes audio over USB Type-C. And because, one cannot be discussed without the other, we also cover the related Thunderbolt 3 technology. First, we will explain the updated specifications fundamentals. Then, we will explore how this will affect the audio industry and important considerations for audio product developers.

By
João Martins
(Editor-in-Chief)

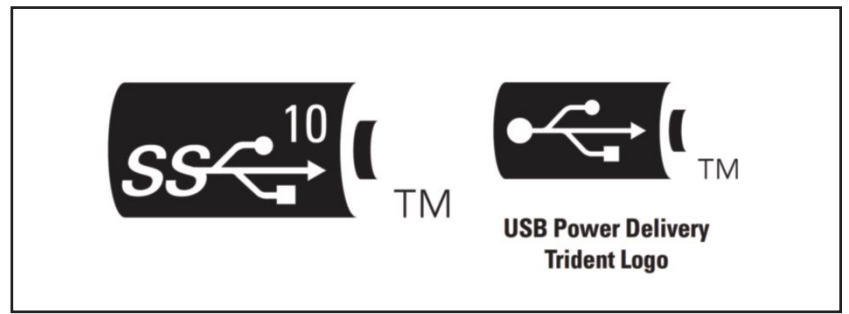
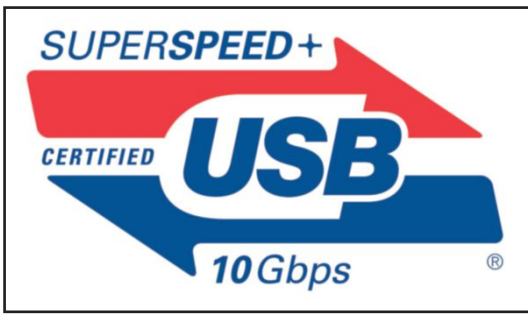
Why is the recently updated USB 3.1 specification and the new USB Type-C connector, or USB-C, associated with Thunderbolt 3? What's new with the USB Audio Device Class 3.0 specification? How can manufacturers and developers plan ahead for new designs based on those new technologies and specifications? We find the answers by questioning some of the key executives involved in the process—the USB Implementers Forum (USB-IF), the group of companies that developed the Universal Serial Bus specification, and the Thunderbolt consortium of companies, both led mainly by executives from Intel Corp.

Given the announced release schedules and the pace of new product introductions, we started the research process at the CES 2016 show in Las Vegas, NV, where we assumed it would be a key topic to address in this magazine just a

few months later. However, that did not come to fruition. Several factors might explain the reason why those “hot” technologies at CES 2016 are only now barely reaching the stores at the end of 2016. Still, before we provide more announcements on these topics, which are certain before and after CES 2017, we decided it was time to publish the available information and start with some fundamentals.

What is USB 3.1 and USB Type-C?

The new SuperSpeed USB 10 Gbps (USB 3.1 Gen 2) specification adds a 10 Gbps speed mode that uses a more efficient data encoding, over enhanced, fully backward-compatible USB connectors and cables. The specification describes the mechanical, electrical, protocol, and hub definition while maintaining compatibility with existing USB 3.0 software stacks and device class



USB 3.1 enables speeds up to 10 Gbps, supporting audio/video for USB hosts, hubs, and devices. Combined with USB Type-C, USB 3.1 and USB Power Delivery define a truly single-cable solution for audio/video, data, and power delivery, building on the existing global ecosystem of USB/IEC 62680 series of International Standards compliant devices.

The USB Trident Power Delivery logo was created to help identify the latest generation USB 3.1 connectors and might include the "10" (Gbps) indicator.

protocols as well as with existing 5 Gbps hubs and devices and USB 2.0 products.

USB Type-C, or USB-C, is a separate specification for connectors and cables. It describes a new generation of symmetrical and reversible connectors—replacing all existing previous USB connectors, including mini or micro connectors. Both sides (top and bottom) can be inserted in the port in either direction, meaning that the user no longer has to make sure a cable is inserted "right-side" up. It also can deliver up to 100 W of power and supports alternate modes, such as DisplayPort, Thunderbolt, and so forth.

USB 3.1 cables with the Type-C connector enable users to charge and power any USB-C enabled device, sync and transfer data at up to 10 Gbps transfer speeds. This is up to 20 times faster than USB 2.0 (480 Mbps), 12 times faster than FireWire 800, and twice as fast as USB 3.0 (5 Gbps). This is fast enough to carry full HD uncompressed video and multichannel audio.

The new symmetrical/reversible Type-C to Type-C cable assembly also provides a user-

friendly reversible design and enables the cable to be inserted into the devices in either orientation or direction. A connection between two Type-C devices still requires that one device is configured as a host and the other remains a device. This is determined on connection. But since USB Type-C cables are fully symmetrical, there is no host- or device-specific end of the cable.

USB Type-C connectors also define a number of new capabilities starting with USB Power Delivery, developed to provide flexible, bi-directional power capabilities by enabling faster charging and increased power levels up to 100 W. The USB Power Delivery specification defines standardized features that support the global adoption of interoperable power supplies, helping to reduce electronic waste and increase re-usability of adapters and chargers for consumer electronics.

USB Type-C connectors also support analog audio and can replace a standard headphone connector. New USB Type-C alternate modes enable support for other technologies and interfaces, because the specification defines a framework for new



USB-C is the generic name being used to identify what is in fact a combination of two separate specifications: the SuperSpeed USB 10 Gbps (USB 3.1 Gen 2) interface and the new USB Type-C specification.

The mobile industry was the first one to adopt USB Type-C connectors for new generation smartphones, allowing for thinner devices.



applications with dedicated USB Type-C physical layers and controllers.

The USB-IF is also working in conjunction with other industry consortia, such as DisplayPort, Mobile High-Definition Link (MHL), and High-Definition Multimedia Interface (HDMI) to establish support in all USB Type-C hosts and devices.

MHL technology—and its latest superMHL specification—enables consumers to connect a variety of mobile and CE products to TVs and monitors and provides up to 40 W of power. With an ecosystem of more than 750 million products, MHL delivers up to 8K 120 frames per second (fps) for home-theater products and 4K 60 fps and beyond for mobile devices. It also supports advanced audio formats such as Dolby Atmos, Dolby TrueHD, DTS:X, and DTS-HD.

The MHL Consortium has already developed an MHL Alternate Mode for the USB Type-C specification. Using the MHL Alt Mode, a USB Type-C connector and cable can support the MHL 3 specification, which includes 4K video, multichannel surround-sound audio, HDCP 2.2, and is backward compatible with existing versions of the MHL specification.

Mobile devices designed with a USB Type-C

receptacle and built to support the MHL Alt Mode will be able to connect to a large ecosystem of MHL TVs, monitors, AVRs, Blu-ray disc players, projectors, set-top boxes, and automotive products using USB Type-C to HDMI Type A MHL cables. Mobile devices with USB Type-C MHL Alt Mode can also connect to HDMI legacy displays, using adapters that support MHL-to-HDMI protocol conversion.

During CES 2016, the HDMI consortium was already demonstrating the adoption of USB Type-C connectors. In September 2016, HDMI Licensing, announced the release of an HDMI Alternate Mode specification, developed for the USB Type-C specification, allowing for HDMI-enabled source devices to utilize a USB Type-C connector to directly connect to HDMI-enabled displays, and deliver native HDMI signals over a simple cable without the need for cumbersome protocol and connector adapters or dongles.

As the announcement stated, “This enables two of the most popular solutions for connectivity to come together—the small form factor, reversible, and multi-purpose USB Type-C connector being adopted by smartphones, tablets, and PC products, and HDMI, which is the leading display interface with an installed base of billions of displays.”

HDMI Alt Mode supports the full range of HDMI 1.4b features such as: resolutions up to 4K, Audio Return Channel (ARC), 3D, HDMI Ethernet Channel, and Consumer Electronic Control (CEC). The HDMI cable will utilize the USB Type-C connector on the source side and any HDMI connector on the display side.

In July 2016, the International Electrotechnical Commission (IEC) formally adopted the latest USB-IF specifications for USB Type-C Cable and Connector, USB Power Delivery, and USB 3.1 (SuperSpeed USB 10 Gbps) specifications.



Thunderbolt 3 defines a superset of capabilities over USB-C connectors and cables. At 40 Gbps, Thunderbolt 3 is the fastest connection available and is bi-directional with four lanes of PCI Express Gen 3 and eight lanes of DisplayPort 1.2.

oppo

Sonica DAC

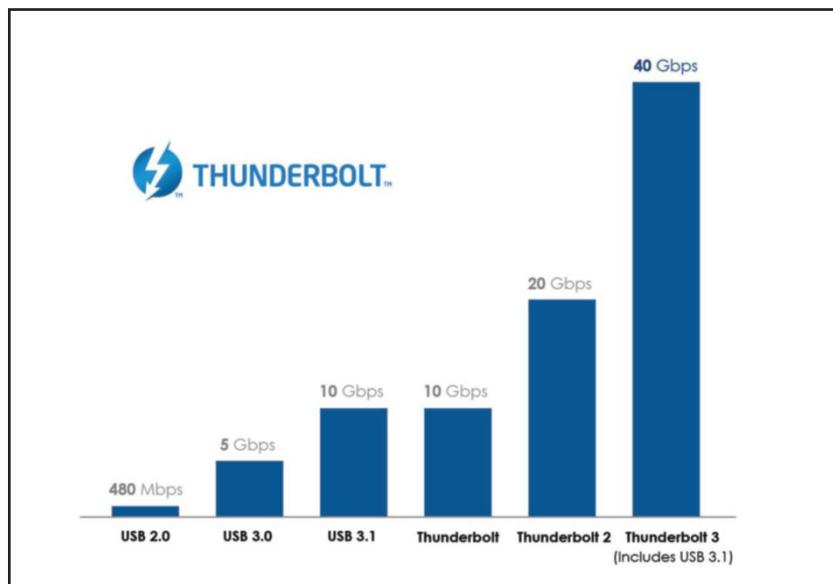
Audiophile DAC &
Network Streamer



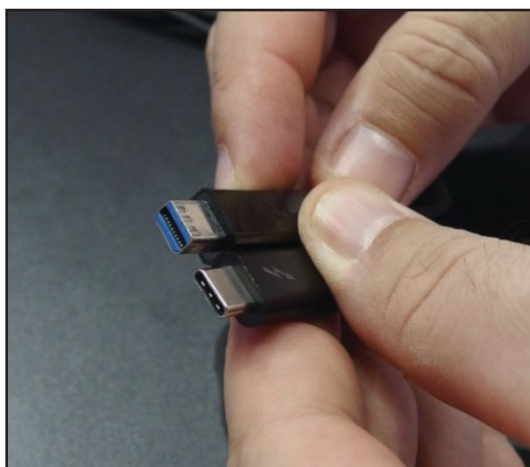
The OPPO Sonica DAC enhances your digital audio experience with an ES9038PRO DAC chip, fully balanced design, a wide variety of inputs, and a high-resolution audio player that plays well with your network shares, connected USB drives, and Sonica Wi-Fi Speakers.

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The evolution of new USB and Thunderbolt interfaces improves data transmission from 480 Mbps in the old USB 2.0 standard to 40 Gbps in the new Thunderbolt 3 technology.



The Thunderbolt original mini-DisplayPort connector is compared with the new Type-C connector with a Thunderbolt symbol.

What Is Thunderbolt 3?

In 2015, Intel unveiled Thunderbolt 3, updating the previous Thunderbolt specifications, developed in collaboration with Apple. Thunderbolt 3 brought Thunderbolt to USB-C at 40 Gbps, creating one compact port that connects to Thunderbolt devices, every display, and also billions of USB devices. Thunderbolt 3 introduces support for USB devices to its original specification and replaced the original mini-DisplayPort connector being used for Thunderbolt for the USB Type-C “universal” connector.

A single Thunderbolt 3 cable provides four times the data and twice the video bandwidth of SuperSpeed USB 10 Gbps (USB 3.1 Gen 2), while also supplying up to 100 W of power for system charging and up to 15 W to bus-powered devices.

The original Thunderbolt technology was a transformational high-speed, dual-protocol input/output (I/O) that provided unmatched performance with 10 Gbps bi-directional transfer speeds—updated to 20 Gbps bi-directional in 2013 with Thunderbolt 2. It provides flexibility and simplicity by supporting both data (PCIe—from which Gigabit Ethernet, FireWire, or eSATA devices are easy to interface) and video (DisplayPort—compatible with DVI, HDMI, and VGA displays via adapters) on a single cable connection that can daisy-chain up to six devices.

The original Thunderbolt specification enabled new and flexible designs and thinner form factors by providing the ability to expand the system capabilities on a single cable. The Thunderbolt 2 update in 2013 saw the technology adoption grow from a standard feature of Apple’s Mac computers, to a few PCs and motherboards, and several peripheral devices.

Still, for almost a year, nothing happened on the Thunderbolt front. PC OEMs never started shipping many of the prototypes shown, Microsoft never introduced support for Windows, and the industry started exhibiting signs that something big was in the works.

The announcement that the USB Type-C specification, approved in September 2014, would adopt a reversible connector similar to Apple’s Lightning connector, was also a clear sign that Cupertino was indeed working with Intel and the USB consortium behind the scenes. Then, Intel made the Thunderbolt 3 announcement and suddenly everything seemed to make sense.

The USB Type-C connector is small, reversible, fast, supplies power, and enables other interfaces, in addition to USB, to run on it, maximizing its potential.

In “Thunderbolt mode,” a single cable supports

A Sampling of the Many Silicon Manufacturers and Development Platforms Consulted for This Article

Cadence | cadence.com
 Cypress Semiconductor | www.cypress.com
 Lattice Semiconductor | www.latticesemi.com
 Microchip | www.microchip.com
 NXP | www.nxp.com
 Parade Technologies | www.paradetech.com
 Qualcomm | www.qualcomm.com
 Qualtek Electronics | www.qualtekusa.com
 Silicon Labs | www.silabs.com
 STMicroelectronics | www.st.com
 StreamUnlimited | streamunlimited.com
 Synopsys | www.synopsys.com/home.aspx
 Texas Instruments | www.ti.com
 XMOX | www.xmos.com

two 4K displays or displays with 5K and more, single-cable docks with charging, external graphics with four lanes of PCI Express Gen 3, and built-in 10 Gigabit Ethernet (GbE) networking. Thunderbolt 3 supports up to eight lanes of DisplayPort 1.2 (HBR2 and MST) and also provides a peer-to-peer connection at 10 GbE speeds to quickly transfer files between computers, perform PC migrations, or set up small work groups with shared storage. Solutions and products previously built to Thunderbolt and Thunderbolt 2 specifications will work with Thunderbolt 3 via adapters.

According to the official Intel/Thunderbolt website, more than 60 PC designs in different form factors were introduced from leading vendors in 2016 or are scheduled to be in the market in early 2017. Products scheduled to be released in 2017 also include PC docks, high-speed storage solutions, external graphics devices, and more.

The Universal Connector: Dream or Nightmare?

Considering that USB-C connectors are being quickly adopted on new-generation smartphones (not necessarily with updated USB 3.1 specs), Microsoft supports USB 3.1/USB-C on Windows 10 and the fact that new Windows 10 machines are also supporting Thunderbolt even if in proprietary alternate implementations (not supported directly by Microsoft), the implications for all industries are huge, with video projectors to cameras to audio interfaces standardizing to a single connector—with support for multiple interfaces.

Initial products with Thunderbolt 3, mainly storage devices, started to ship in 2016. Unfortunately, Apple delayed the introduction of its Mac computers with the new interface while waiting for the availability of a new generation of Intel processors with significant improvements. The result—as it turns out to be the norm in past years—was that the entire PC industry also delayed the introduction of new machines with Thunderbolt 3 during most of 2016. And even the few models that were introduced have not implemented it to the full Thunderbolt 3 specification.

As Greg LaPorte, vice president of sales and marketing for Sonnet Technologies—a provider of Thunderbolt-to-PCIe card expansion systems and storage systems—explained, new computers already introduced in 2016 did not fully support all the features of Thunderbolt 3.

According to LaPorte, some computer models have not shipped with support for four times lanes of PCIe Gen 3, but only provide two times lanes to Thunderbolt, have only support for one



This is a Windows Dell laptop with support for Thunderbolt 3 as demonstrated at CES 2016.

stream of DisplayPort 1.2 instead of two (which means supporting only one 4K display and not 5K). Even worse, instead of providing 15 W of power to Thunderbolt 3, they only provided 7.5 W, which means that not all Thunderbolt 3 bus-powered devices will be supported by that computer. “I am hopeful that the upcoming crop of computers will have fewer models lacking these features,” LaPorte told us.

The reason why PC makers are waiting has to do with the conviction by many that even Apple will implement only USB 3.1 in some machines, such as its MacBook series, making Thunderbolt 3 only available for the Mac Pro, iMac, and the MacBook Pro range.

From the Windows camp, apart from workstation-class applications, there is a conviction by many vendors that 10 Gbps and USB 3.1 would be sufficient for generic PCs. Maybe the need to



HDMI 2.0a is now available over USB-C. Parade Technologies also implemented DisplayPort Alt Mode over USB Type-C to HDMI protocol converters.



Video Electronics Standards Association's (VESA) DisplayPort digital display interface now also runs over USB-C.

support 4K displays and gaming applications could be the incentive needed for those companies to fully embrace Thunderbolt 3, but only if Microsoft decides to introduce native support.

This uncertainty is likely to remain within the computer industry, but the mobile industry is not waiting. New smartphones will ship with USB Type-C connectors and compatible new-generation chargers. From Apple, as we've seen with the iPhone 7, there is a Lightning connector for everything and the new flagship phone now ships only with Lightning-

equipped earphones. There is no Lightning connector even on Mac computers, so this means that either users go back to their original headphones with the old analog jack when they use a computer, or they need a Lightning to USB Type-C converter to listen to music on their MacBooks. This means that the market for adapters will be booming, but more worrisome will be the fact that not all cables will be the same, even if they seem to have the right connectors.

For consumers, this will mean that they will have to learn to actually recognize the tiny symbols on ports and cable connectors, to distinguish if it is a Thunderbolt connection, a USB-C, or an USB Type-C port used for some of the Alternate Modes. And this could become even more confusing with devices that supply power and charge other devices, not to mention that there are great limitations on the length of USB-C cables (1 m or 3.3' on USB 3.1 Type-C Gen 2) if you expect to supply bus-power on a cable.

Limitations in length are less with Thunderbolt cables, but they also exist—up to 2 m if providing power. A 2 m Thunderbolt 3 cable can also be used with USB 3.0 and 3.1 devices, with data transfer speeds of up to 5 Gbps. Great lengths, from 5 m up to 60 m are possible with Corning's optical Thunderbolt cables but obviously those cables don't supply power.

The USB-IF is already working on certification programs for cables and chargers, and it recently published a Certified USB Charger Compliance and Logo Program, based upon the USB Type-C and USB Power Delivery specifications. Certified USB Chargers will now display a new logo identifying the power capability in watts. That program was announced in August 2016.

The more we learn about it, the more complicated the idea of having a single "universal" connector type seems to be than initially envisaged—since the same connector can be used for many different purposes and is able to support Alternate modes, something which is not immediately obvious to the user. Also, because people tend to store all the cables and power adapters/chargers that came with their smartphones, computers, and other devices in the same drawer—and when all those cables use the same connector—users will pick the first one they find and the one with the desired length. Those users might be disappointed when they plug it in and things don't work... or worse.

As recently published stories have shown already, the market is already being flooded with USB Type-A to USB Type-C cables that are not implemented according to the specifications and have the wrong resistors, causing damage on laptops. And, we are just getting started. 

Resources

M. Christiansen, "Implementing USB Type-C in High-Speed USB Products" Application Note: USB IP, Synopsys, www.synopsys.com/Company/Publications/DWTB/Pages/dwtb-usb-type-c-2015q3.aspx.

—, "Converting Existing USB Designs to Support USB Type-C Connections" Application Note: USB 2.0 PHY and Type-C connectors (available through Solvnet), www.synopsys.com/Company/Publications/DWTB/Pages/dwtb-convert-usb-design-type-c-2015q1.aspx.

S. Haskell, T. Nguyen, F. Yan, and A. Salazar, "Challenges of USB 3.1 IP Certification," www.synopsys.com/COMPANY/PUBLICATIONS/DWTB/Pages/dwtb-challenges-usb-certification-2016q2.aspx.

High-Definition Multimedia Interface (HDMI) | www.hdmi.org

Mobile High-Definition Link (MHL) | www.mhltech.org

Sonnet Technologies | www.sonnettech.com

Thunderbolt Technology Community | thunderbolttechnology.net

USB Implementers Forum (USB-IF) | www.usb.org.

Sorting Out USB Type-C, Thunderbolt 3, and USB 3.1

Greg LaPorte is vice president of sales and marketing for Sonnet Technologies, a provider of Thunderbolt-to-PCIe card expansion systems, pro media readers, RAID storage systems and storage, network, and other interface cards for professional users in the audio, video, and broadcast industries. He kindly answered our questions on these technologies.

audioXpress: What are the most common misconceptions about USB Type-C?

Greg LaPorte: There are many—including the assertion that USB Type-C and USB 3.1 are the same thing. In reality, a Type-C connector supports many different protocols in addition to USB 3.1, including USB 3.0, Thunderbolt 3, and many types of video protocols such as HDMI.

I have also heard that Type-C is replacing Thunderbolt, which is also not true. The next-generation Thunderbolt 3 interface has adopted the Type-C connector and incorporated the ability to accept USB 3.1 and USB 3.0 devices, as well as video protocols.

audioXpress: What are the benefits of USB Type-C/USB 3.1, and how is it being received in the market? Will it eventually replace all other USB formats?

Greg: At a purely physical level, USB Type-C is a more convenient connector because it has no up or down orientation and can be plugged in either way without flipping. It's also very thin, allowing notebook computers, tablets, and even cell phones to trim down even more. The high-speed USB 3.1 Gen 2 protocol can transfer data at 10 Gb/sec, as fast as the original Thunderbolt protocol and double the 5Gb/sec of USB 3.1 Gen 1 and USB 3.0. However, Thunderbolt 3 is now even faster, clocking in at an amazing 40 Gb/sec.

USB 3.1 has a more robust power delivery profile, allowing up to 100 W of upstream power for charging host computers from capable devices. You can push power up from a device to the host through the same port that the device is using to communicate with the computer. There are complex power delivery specifications for managing that, but the upshot is a lot more flexibility and convenience. A docking station is a good example. With one cable to connect multiple peripherals (attached to the dock) and provide power to the computer, you can keep your laptop's power adapter in your travel bag.

audioXpress: What's driving the convergence of USB-C and Thunderbolt 3? How are they complementary?

Greg: With Thunderbolt 3 adopting the Type-C connector, it appears the handwriting is on the wall—eventually USB Type-C will become the de-facto connector. What is really exciting is Intel's strategy

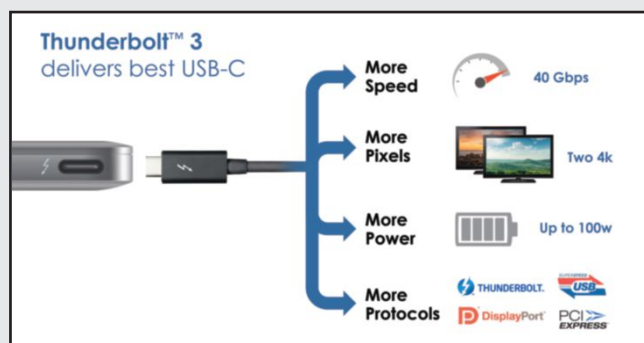


Greg LaPorte is vice president of sales and marketing for Sonnet Technologies.

to create what they're calling "the USB-C that does it all."

In addition to adopting the Type-C connector, Intel has embedded USB 3.1 Gen 2 support in its Thunderbolt 3 chips. This has paved the way for a superset universal port that can support just about everything, including all the popular video protocols (video and PCIe data over a single cable has always been one of the key features of Thunderbolt technology). In other words, with the right Type-C adapter, devices supporting any Thunderbolt version, any USB version, and (DisplayPort, DVI, HDMI and VGA) video protocol can connect to a Thunderbolt 3 port and oh, by the way, if the device supports the advanced power delivery features that I previously mentioned, Thunderbolt 3 handles that as well.

While earlier Thunderbolt versions were primarily adopted by Apple, with Thunderbolt 3 we are seeing broad adoption by Windows PC manufacturers. The first wave of these computers



The next-generation Thunderbolt 3 interface has adopted the Type-C connector and also incorporated the ability to accept USB 3.1 and USB 3.0 devices, and video protocols.



Sonnet Fusion PCIe Flash Drive SSD storage device with Thunderbolt 3 interface for Mac and Windows.

is sporting one or two Thunderbolt 3 ports plus a few legacy USB 3.0 Type-A ports. I think in time we will eventually see only Type-C USB connectors on notebooks and, if I had to bet on it, I'd put my money on Thunderbolt 3 (and its successors) becoming the de facto computer interface.

audioXpress: What advice would you give consumers to sort out the confusion?

Greg: If I were choosing a new computer today, I would definitely select one with Thunderbolt 3 over one equipped solely with USB 3.1 because it offers so much flexibility and versatility. You have access to a much greater bandwidth, and you can connect to anything and daisy chain many devices from one connection. With Thunderbolt 3 you don't have to worry about whether you can connect the device you want to use. If it has a Type-C connector, or if it will connect to an adapter with a Type-C connector, it will work.

From a device perspective, it's a different question. If the computer you're trying to connect to only has USB 3.1, you have no choice but to select a USB device. If the host has Thunderbolt 3, your options are wide open, depending on what you are expecting from the device. If the ability to support daisy chaining of multiple devices is important, select a Thunderbolt device with two Thunderbolt ports. If performance above 800 MB/sec is important, then choose one that's equipped with Thunderbolt 2—or Thunderbolt 3 if you need greater than 1,375 MB/sec. If you need to support one or two 4K displays, Thunderbolt 3 is the correct choice. If you are selecting an external storage device with one or two spinning drives or one SATA SSD, then a USB- Type C 3.1 interface will provide plenty of bandwidth and may be a good choice as it can get all the bandwidth it needs from either a USB 3.1 or Thunderbolt 3 interface.

audioXpress: What do you see as the biggest applications/greatest uses of USB-C or Thunderbolt 3 for media production?

Greg: The increased bandwidth in Thunderbolt 3 (2,750 MB/sec compared with 1,375 MB/sec for Thunderbolt 2) will be welcome in video postproduction, DIT, and broadcast markets. For example,

with our Thunderbolt 2 PCIe card expansion chassis, the Red Rocket-X, and some video I/O cards require more bandwidth than Thunderbolt 2 can deliver—but with Thunderbolt 3 there will be plenty of bandwidth to spare.

Video ingest from Thunderbolt 3 equipped media readers, such as our CFast 2.0 Pro Card Reader, will be much faster too, leading to big time savings. Another example is our currently shipping Fusion PCIe Flash Drives, the world's fastest bus-powered portable drive. The Thunderbolt 2 version transfers data at up to 1,350 MB/sec, and the Thunderbolt 3 version up to 2,100 MB/sec. We have many customers using these as scratch drives for video editing on location. An increasing number of these customers need the bandwidth of our Thunderbolt 3 edition to handle raw 4K video at higher frame rates. Then, there is the ability to support two 4K displays over a single Thunderbolt 3 chain of devices, also a big leap forward.

In the pro audio market, there are very few use cases that even come close to maxing out the bandwidth of Thunderbolt 2, so the increased speed won't be as much of a factor. However, the convenience of a single Thunderbolt 3 port that can support devices with all types of interfaces will be a real plus. Finally, Thunderbolt 3 provides the lowest latency for PCI Express audio.

audioXpress: What should customers look for when evaluating PCIe card expansion chassis and other products that rely on Thunderbolt 2 now, and Thunderbolt 3 in the future?

Greg: Core features should include reliability, ruggedness, variable-speed cooling fans, ultra-quiet operation, and easy upgradability to the next generation of Thunderbolt. Your investment in a PCIe card enclosure goes a lot further when you can migrate to Thunderbolt 3 simply by replacing a Thunderbolt board. Beyond that, it is important to select a product with a sufficient number of PCIe slots to accommodate the cards you need today as well as those you may need in the future. For example, many of our customers find it highly desirable to expand their attached storage with high-performance SSDs mounted on one of our Tempo SSD drive cards;



An example of Sonnet's expansion chassis—the Thunderbolt Echo Express III-D for up to three PCIe cards.

in other words, storage expansion that can be mounted within the same chassis as the other cards. Without the extra card slot, they would miss out on being able to add this capability.

In some cases, users need an expansion chassis that can be rack-mounted, such as our Echo Express III-R, or even a rack-mount chassis that can also securely hold the computer, such as our xMac Mini Server or xMac Pro Server. Because of the bandwidth limitations of Thunderbolt 2, some users have hesitated to put more than one high-performance card in the same Thunderbolt 2 chassis, such as a dual-port 10 Gb Ethernet card combined with a RAID storage controller card. With Thunderbolt 3, most of these concerns will disappear.

audioXpress: What is Sonnet's product strategy for USB-C? Is it in line with Apple's strategy for the Mac Pro?

Greg: While we don't have any inside knowledge of Apple's product plans, we assume they will migrate at some point to Thunderbolt 3 for next-generation Macs. For us, the bigger question is when we will see new product from Apple with Thunderbolt 3, and which Mac models will be first. But it's not just about Apple. At this point, in 2016, there are more than 60 shipping Windows computers with Thunderbolt 3, and as we roll into the 7th Generation Intel

Core processor equipped PCs (some of which have already started shipping) it is expected that more than 120 models will be equipped with Thunderbolt 3.

Uncertainty about Apple's timing aside, we have created Thunderbolt 3 versions of many of our current products and are innovating with some new product designs that will help users benefit from the advantages of Thunderbolt 3. All of our current PCIe chassis are upgradable to Thunderbolt 3, and we will soon be offering an upgrade card for that purpose, but it will be a while before Thunderbolt 2 devices are no longer needed. Intel has indicated that there were 100 million computers equipped with Thunderbolt or Thunderbolt 2 in existence at the end of 2015, so that market isn't going away anytime soon.

There are Thunderbolt adapters on the market that enable users to connect Thunderbolt and Thunderbolt 2 devices to the new Thunderbolt 3-equipped computers. The adapters will benefit those who already have Thunderbolt devices as they migrate to newer Thunderbolt 3 computers. And users purchasing their first Thunderbolt computer will have access to a large variety of ready to purchase devices. We are hopeful that in the not too distant future, there will also be bi-directional adapters that will allow new devices with Thunderbolt 3 to be used with Thunderbolt and Thunderbolt 2 equipped computers.

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A Complementarity Principle in Acoustics

Quantum mechanics complementarity: The complementarity principle in physics is a tenet that a complete knowledge of phenomena on atomic dimensions requires a description of both wave and particle properties. The principle was announced in 1928 by the Danish physicist Niels Bohr.

By
Richard Honeycutt
(United States)

One of the most convenient properties of physics is also sometimes one of the most confusing. This property is often called "complementarity," and it is the property according to which a complete understanding of certain phenomena can only be

achieved by using two very different models, with two very different mathematical formalisms. One example is the long-running debate over whether light was a wave or a particle phenomenon.

In 1690, Christiaan Huygens proposed that light is a wave phenomenon. However, in 1704, Sir Isaac Newton announced that light consisted of tiny particles, or "corpuscles." Thomas Young and A. J. Fresnel demonstrated diffraction and the interference of light in 1801 and 1814-1815, respectively; and since diffraction and interference are wave phenomena, light was thereafter considered to be a wave phenomenon.

Defining the Complementarity Principle

In 1905, Albert Einstein explained the photoelectric effect by considering light to be made up of packets of energy called quanta or photons. When a photon strikes certain materials, an electron is released from an atom on the surface. This is how a photomultiplier tube works. We now understand that light has a dual nature: both wave and particle.

Electrons were originally considered to be tiny

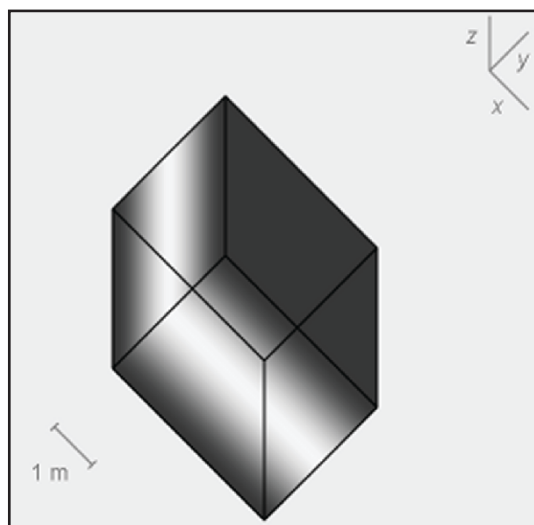


Figure 1: Wave analysis can give results that can be plotted to show strong and weak areas in a sound field at a modal frequency. The dark regions correspond to the high-loudness areas.

particles of matter, but in 1924, Louis de Broglie proposed that particles of matter should also behave as waves. Three years later, George Paget Thomson of the University of Aberdeen, and Clinton Joseph Davisson and Lester Halbert Germer of Bell Labs demonstrated the wave nature of electrons. In fact, if we create an electron beam and place a particle detector in its path, we will find electron particles. But if we split the beam and make it pass through two slits, interference patterns will be created, showing a wave nature.

In 1928, Neils Bohr announced the Complementarity Principle, which states that for complete knowledge of phenomena of atomic dimensions, both wave and particle properties must be taken into account. This is only one of the counter-intuitive facts of quantum mechanics!

Acoustical phenomena also exhibit a dual nature, but the two aspects are described as waves and rays rather than waves and particles. This is most commonly seen in contexts of our conceptualization of sound, and in computer modeling. (Actually, EASE electro-acoustical software refers to the elements used for "ray-tracing" as "particles.")

When we attend an outdoor concert, we try to stand or sit where we have a clear line of sight to the speakers. In this case, we are thinking of sound as rays that travel only in straight lines. In computer modeling, the most computationally efficient method of predicting audience coverage by a sound source, echoes, and several other acoustical phenomena is to use ray tracing. In many cases, ray tracing gives useful predictions. But, there are several conditions under which ray tracing will not reveal important acoustical effects.

Ray Tracing Limitations

One such case is the development of standing waves (modes) in a listening room. **Figure 1** shows a map of relative sound pressure level (SPL) at the 16th mode in a 4 m × 3 m × 3 m room. The sound sources are not shown. This plot was made by the software at huenecke.de (www.huenecke.de/en/calculators/room-eigenmodes.html).

Figure 2 shows the same room as modeled in Computer Aided Room Acoustics (CARA). **Figure 3** shows the SPL map for the same mode in the same room, as plotted by CARA software. Note that the sources are included in this mapping calculation, resulting in higher SPL (orange) at the side of the room where the speakers are located. This SPL map is based upon the SPL after four reflections.

Figure 4 shows the SPL map produced by an architectural acoustics modeling software program.

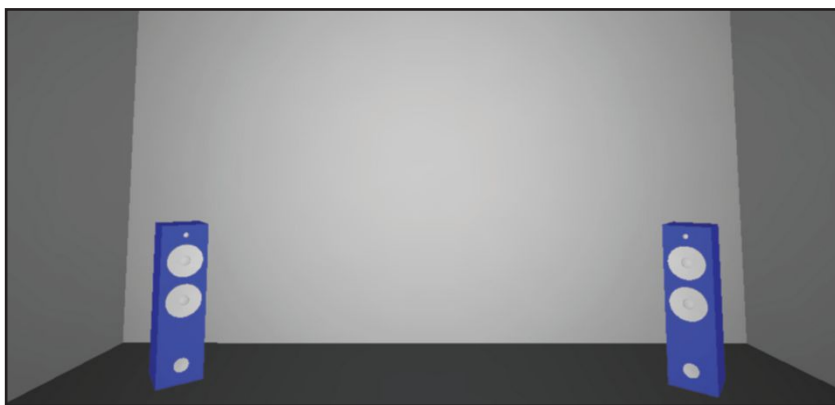


Figure 2: The 4 m × 3 m × 3 m room is shown with the speakers in place.

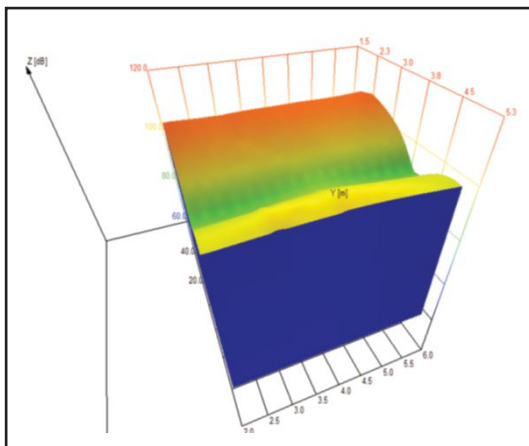


Figure 3: Specially designed ray-tracing engines such as CARA can present plots of the SPL at modal frequencies.

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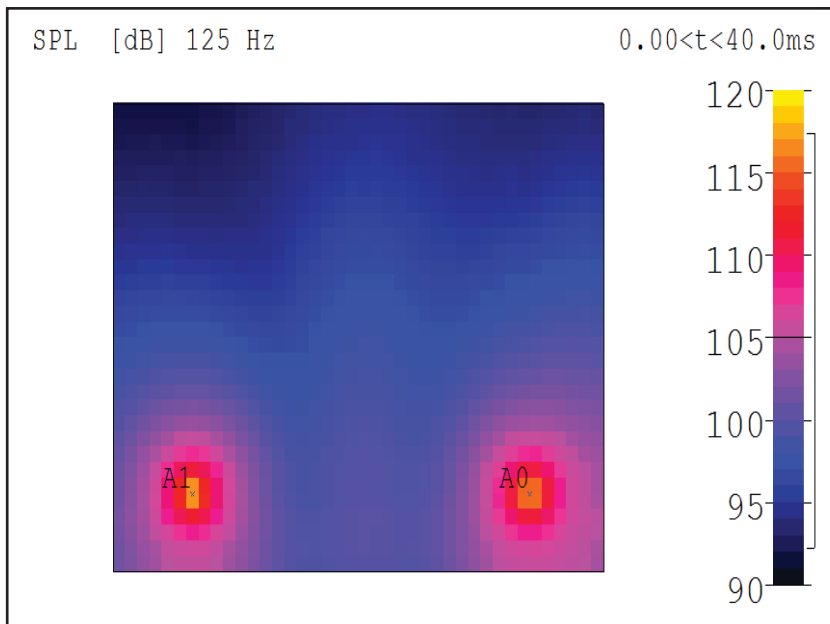


Figure 4: General-purpose architectural acoustics software is not designed to show modal behavior.

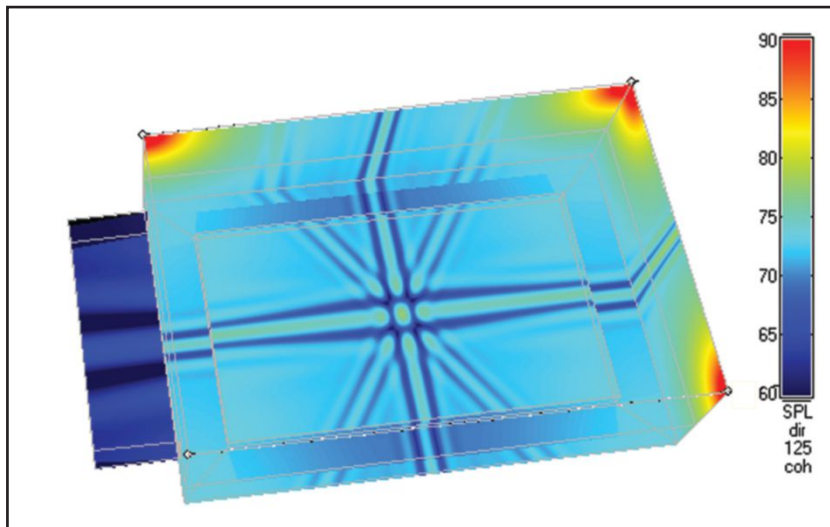


Figure 5: The dark lines show interference bands (which would vary with frequency) caused by sound from multiple speakers arriving at the same place in different phase relations. (Because of the perspective in which the drawing is shown, the lower left speaker does not show, although its interference effect is shown.)

As with most such programs, the resolution is limited at best to 1/3 octave. (In this figure, one-octave resolution was used.) Notice that the modal behavior does not show at all. This is not a weakness in the software, since the rooms for which it was designed are generally far too large for modes to be a concern. But it is important for the user to remember to use the appropriate software for the space (s)he is modeling.

Figure 5 shows the interference caused by a common mistake committed by amateurs who install sound systems in medium-sized rooms: what I call “the four-corner disaster.” The dark lines represent places that are covered by sound from more than one speaker. Since the distance from each speaker to a given listener is different, the received sound will be out of phase, and will reinforce, causing peaks in the frequency response, or cancel, causing dips. The resulting frequency response looks like the profile of a comb, hence the name. Comb filtering creates a very unnatural sound and interferes with speech intelligibility.

Refraction

Refraction in acoustics is primarily considered in cases where the temperature (and hence, the

Waves and Particles

The distinction between a wave and a particle hinges upon behavior. A particle obeys Newton’s laws of motion, moving in a straight line at constant velocity unless acted upon by an external force.

A wave can be composed of particles, as in water waves and sound waves, but is a system, exhibiting “emergent” properties (the whole is greater than the sum of its parts). These include:

- Interference, by which two waves interact, creating points of greater or lesser intensity. A standing wave is a wave that travels repetitively in a certain constrained space, creating stationary points of greater or lesser intensity.
- Refraction, which is the bending of a traveling wave’s trajectory due to the wavefront crossing from one medium to another non-perpendicularly
- Diffraction, which is the bending of a traveling wave’s trajectory due to the wavefront upon passing across the edge of a boundary or an obstacle.

References

- [1] W. J. Davies, “The Effect of Seating on the Acoustics of Auditorium,” PhD Thesis, Salford University, 1992.
- [2] K. Ishida, “Investigation of the Fundamental Mechanism of the Seat-Dip Effect Using Measurements on a Parallel Barrier Scale Model,” *Journal of the Acoustical Society Japan*, 1995.

Source

Computer Aided Room Acoustics (CARA) software
ETS Services GmbH & Co. | www.cara.de

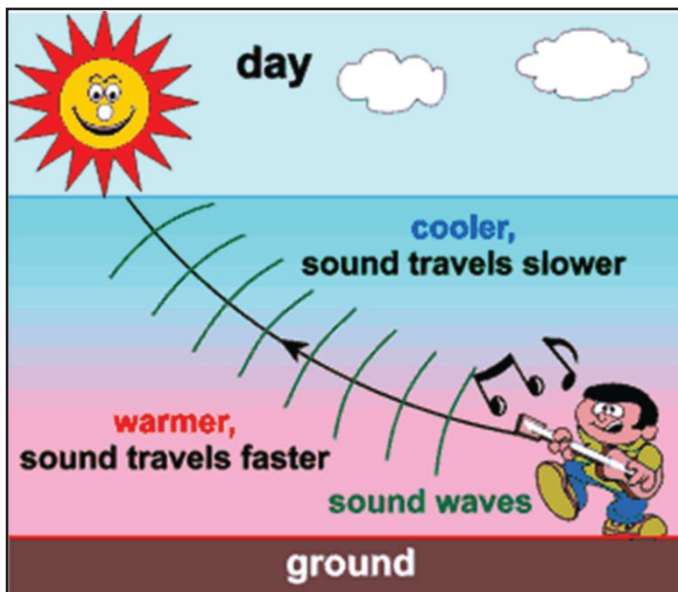


Figure 6: When the upper air is cooler, sound is refracted upward.
(Image courtesy of Physics World)

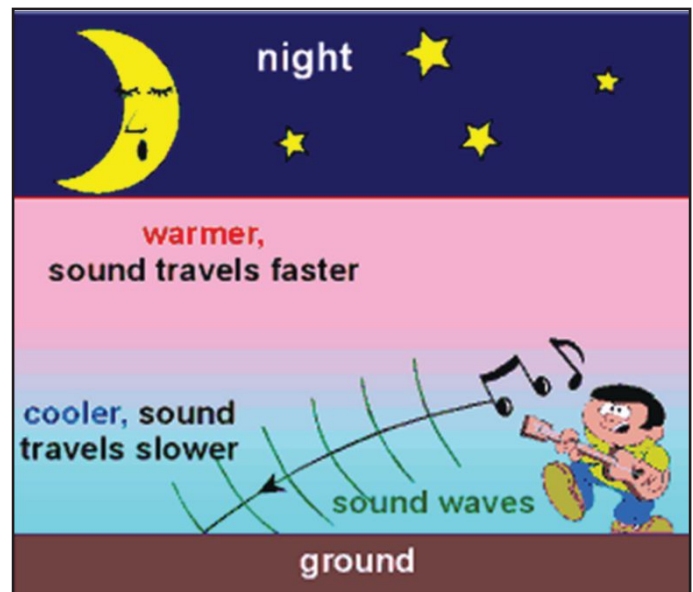


Figure 7: When the upper air is warmer, sound is refracted downward.
(Image courtesy of Physics World)

speed of sound) varies with height. In an outdoor concert, the temperature gradient affects how loud the sound is for listeners at varying distances from the speakers (see **Figure 6** and **Figure 7**). This can cause annoyance for people in their homes far away from the concert. The same issue can affect the sound coverage of an audience in a coliseum.

Diffraction

Diffraction in acoustics is the reason that we can hear sound that passes around corners, especially low-frequency sound. **Figure 8** illustrates that sound having a small wavelength is diffracted more by a small gap than by a large one, and that sound with longer wavelengths is diffracted more than sound with small wavelengths. In other words, low-frequency sound is diffracted more than high-frequency sound.

Just as sound passing through a slot is diffracted into what would be the shadow region if analyzed using the ray approach, so sound passing around an obstacle will be diffracted into the shadow region. As with diffraction of sound passing through a slot, low-frequency sound passing around an obstacle is diffracted more than high-frequency sound.

Figure 9 shows the shadow of a rectangular obstacle, predicted by a computerized acoustical modeling program that does not include diffraction effects. (The square with an X inside represents the speaker; the white rectangle is the obstacle; the black area is the acoustical shadow on the

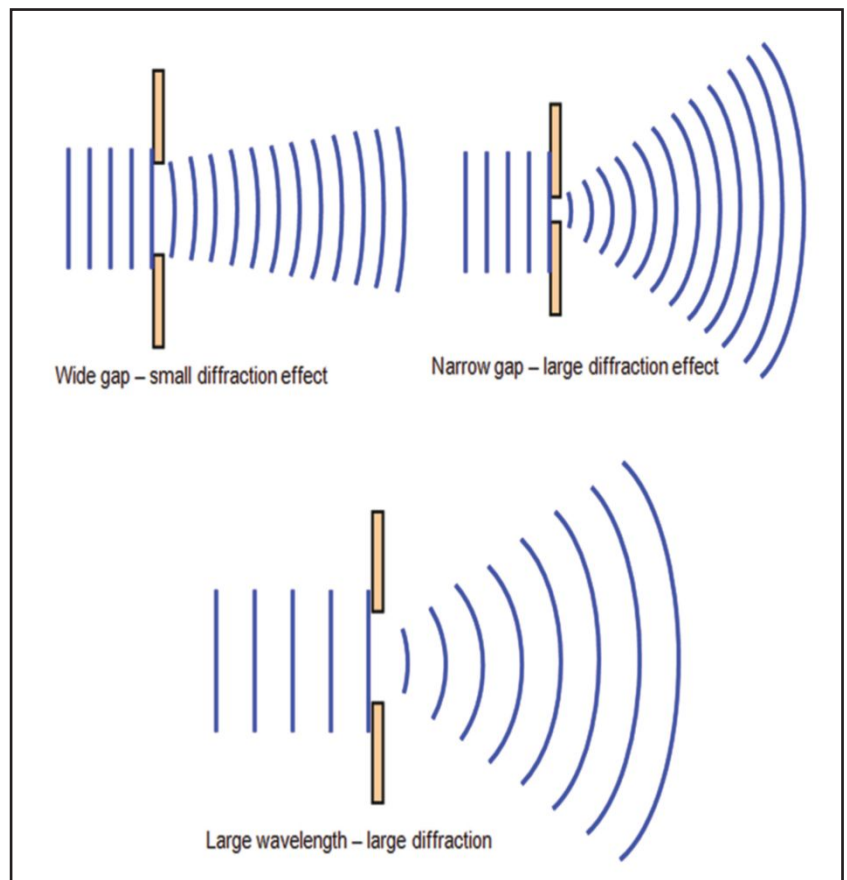


Figure 8: Diffraction depends upon the relation of the wavelength to the width of the gap or slot. (Image courtesy of schoolphysics)

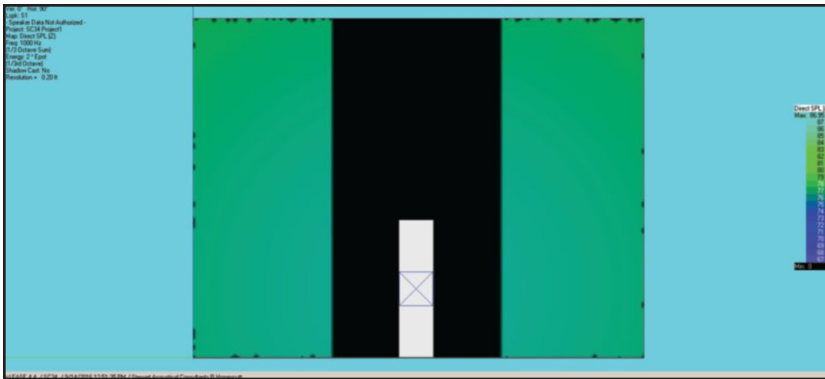


Figure 9: Modeling programs that do not include diffraction will falsely predict sharp acoustic shadows.

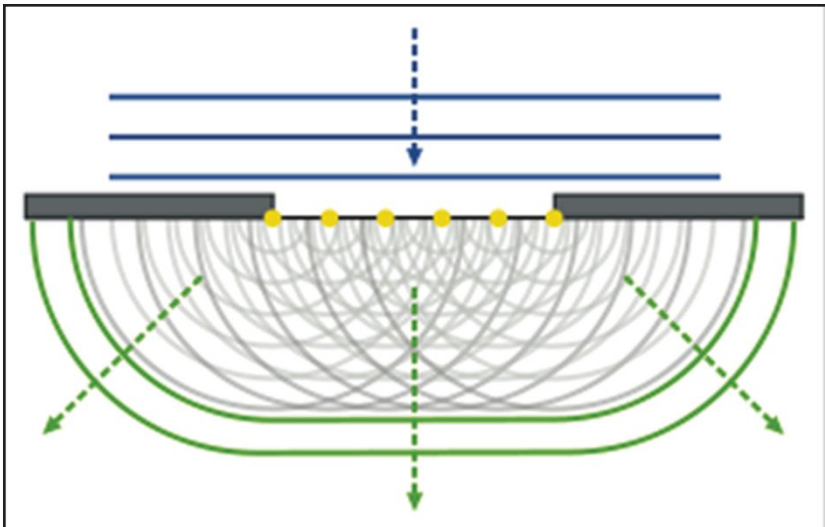


Figure 10: The Huygens-Fresnel Principle can be used to predict diffraction. (Image courtesy of Wikipedia)

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opposite wall; and the green area is the portion of the wall that is exposed to sound.) The predicted shadow does not vary when the frequency of the source is changed from 100 Hz to 10 kHz.

In a real situation, the shadow would be similar to the shadow shown in the figure for a frequency of 10 kHz, but would be much smaller for a frequency of 100 Hz. Also, if diffraction were considered, the SPL map would shade gradually from green to black, rather than transitioning sharply.

Diffraction can be modeled using Huygens-Fresnel Principle: "Every point which a wave reaches becomes a source of a spherical wave; the sum of these secondary waves determines the form of the wave at any subsequent time." This principle is illustrated in **Figure 10**.

In the figure, the blue lines at the top represent a plane wave impinging upon a slot in a wall. The yellow dots are representative "sources of spherical waves." The green lines represent the composite wave front of the sound emerging from the slot. This same principle can be used to predict the spreading of sound from a source such as a speaker or horn. It also explains why small horns only control directionality at high frequencies.

Seat Dip Effect

Another acoustical phenomenon resulting from diffraction is the "seat dip effect," which was discovered by chance during measurements in the New York Philharmonic Hall. The resulting dip in frequency response could be as severe as 10 to 20 dB in the frequency range between 100 and 300 Hz.

In 1992, W. J. Davies found that this dip is perceptible to an audience if it is over 7.1 dB.^[1] Subsequent investigation by Koji Ishida revealed that the cause of the seat dip is diffraction produced as sound crosses the seat back.^[2] Seat dip effect will not be predicted by modeling programs using only ray-tracing algorithms.

The point of complementarity principles is not that there are different ways of looking at physical phenomena; it is that for a complete understanding of the phenomena, both the wave and the particle behaviors must be considered.

When a playing card is suspended in a propagating sound wave, the card will become oriented parallel to the direction of propagation of the wave. This behavior is most easily predicted by considering the particles of air that make up the wave. But as we have seen, interference, refraction, and diffraction can only be predicted using the wave properties.

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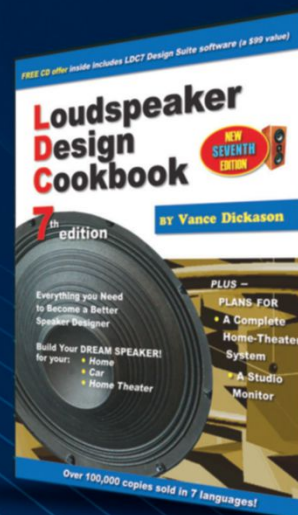
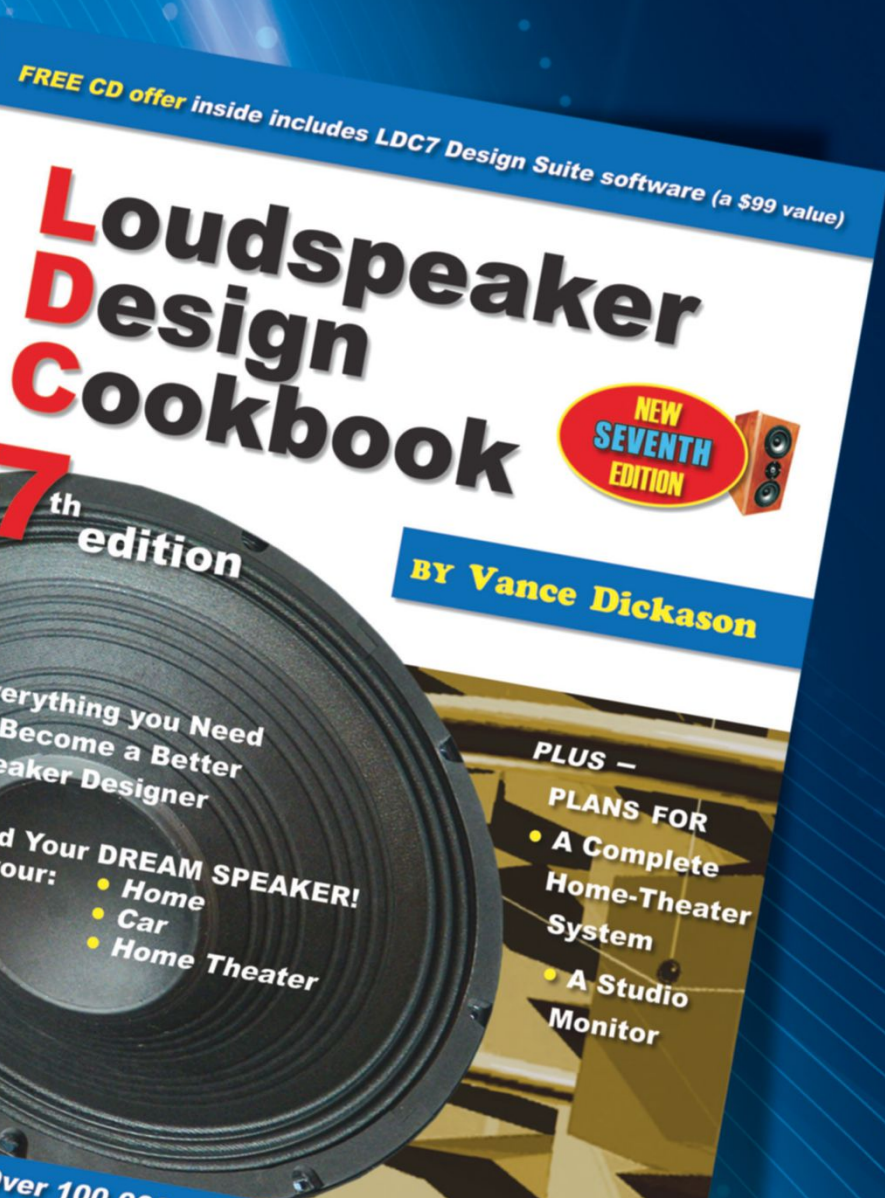
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Innovative Measurement Microphones from PCB Piezotronics



Photo 1: The 480M122 provides a 4 mA constant current for operation and amplifies the microphone preamp's signal. (Image courtesy of PCB Piezotronics)

Recently, I had the privilege to test a few of PCB Piezotronics professional-grade measurement microphones: the 378A04 free-field microphone, the 378B02 free-field microphone, the 378A21 diffuse-field microphone, and the 130E20 array microphone. Here are my findings.

By
Stuart Yaniger

(United States)

A recurring theme in my test and measurement articles has been the 80/20 rule: 80% of the results come with 20% of the expenditure (or, equivalently, 80% of the cost is needed to get the last 20% of performance). And there's a lot of good low-cost options out there for hobbyist and casual designer use. But sometimes, you need a professional-grade instrument, especially for any task where there could be regulatory or compliance issues.

For professional acoustic measurement, there's a whole slew of choices. I have tended to stick with the tried and true, and in my day-job research, Brüel & Kjær has always been a reliable go-to, albeit an expensive one. But as a confirmed miser, I'm always on the lookout for alternatives that can deliver the same performance in a less costly package. So when I was contacted by PCB Piezotronics (Depew, NY) to check out four microphones in its professional line. I hesitated for about 3 nanoseconds before agreeing. The company sent me its ICP signal conditioner (see **Photo 1**) and four microphones—the 378A04 free-field microphone, the 378B02 free-field microphone, the 378A21 diffuse-field microphone, and the 130E20 array microphone (see **Photo 2**).

It's well known that technologies designed for top-of-the-line equipment often migrate down and improve the performance of lower-cost products.

What's interesting is when the opposite occurs—a less complex and expensive technology is optimized to punch out of its weight class. In this case, it's the upward migration of electret microphone technology to the hallowed halls where polarized condensers once ruled.

There are two basic technologies used for measurement (as opposed to recording) microphones: externally polarized condenser and self-polarized electret. Electret microphones dominate the low-end market, and many of them have been surprisingly good. An example popular with speaker hobbyists, dealers, and small shops is the Panasonic WM capsule series, most notably the WM-61A used in Joseph D'Appolito's Mighty Mike, as well as dozens of other similar projects.

The issues with inexpensive electret capsules have traditionally been noise, dynamic range, stability, and repeatability. However, their frequency responses can be reasonably consistent, and with regular calibration, they can cover a lot of measurement territory. Because of the favorable cost/performance tradeoffs, inexpensive electret capsules have become the basis of the latest generation of inexpensive and remarkably good-performing loudspeaker measurement gear.

Recognizing the limitations of this technology, a few people have tried improving the interface portion

of the inexpensive electret capsules—most notably Siegfried Linkwitz (a very simple modification that increases the dynamic range of inexpensive Panasonic mic capsules) and Scott Wurcer (a complete treatise on electret mic interfacing)—with great success. But to get the highest possible performance and measurement-grade stability, we need to use something better than low-cost capsules.

Silk Purses From Coppa Piacentina

One way a serious measurement microphone can be distinguished from the available good-performing hobbyist microphones (which work great for basic speaker and room measurement) or most recording microphones is the use of higher quality, more stable materials. For example, most inexpensive microphones will use a metallized polyester film as a diaphragm.

The use of plastic for this component compromises temperature and humidity stability and measurement-to-measurement repeatability. This is less important for recording and basic speaker measurement, but for repeatable and traceable measurement, a more stable material (e.g., stainless steel) needs to be used. The tradeoff, besides cost, is mass, so a very thin (and expensive) foil needs to be used, which was PCB Piezotronics' choice.

There's a range of electret materials. The best and most expensive microphones show high stability with varying temperature and humidity and are also highly stable over time, which is a significant factor at the price levels commanded by measurement microphones.

PCB Piezotronics was reticent to reveal which electret material it uses, and since these microphones were on loan and not inexpensive, I demurred on disassembly and analysis! If I were to guess, it would be an ion-implanted fluoropolymer, due to the excellent thermal stability and high resistivity, but this is sheer speculation.

Figure 1 shows a diagram of the PCB Piezotronics capsule construction (self-polarized) contrasted with a condenser capsule (externally polarized). Another way in which lab-grade measurement microphones differ from conventional recording or inexpensive measurement microphones is the method of getting the signal from the diaphragm to a voltage and current level compatible with measurement inputs (e.g., analog-to-digital-converter cards).

Conventional electret capsules generally require a DC voltage of typically 5 to 15 V to power their on-board circuit (usually a source follower). Condenser microphones are generally balanced output and use an external polarization voltage from 48 to 200 V.

PCB Piezotronics developed an interface method using a current source rather than a voltage source, called ICP. The advantages of ICP include compatibility



Photo 2: The four PCB Piezotronics microphones that I tested vary in size. (Photo courtesy of Cynthia Wenslow)

with all measurement products in the PCB Piezotronics line, simple and inexpensive interface circuitry, and coaxial-based input and output connections. Many laboratory and industrial data acquisition cards have constant current input capability, so the ICP microphones can be used directly with their inputs. The PCB Piezotronics microphones all attach to a mount or stand via a 0.5" diameter clip, available as an accessory.

Interfacing: 480M122 ICP Signal Conditioner

The 480M122 signal conditioner comes in solid but fairly plain packaging and it takes three standard 9 V batteries for power. Note that although the faceplate for the unit in the photo shows a different model number (480E09), the bar code serial number indicates the '122, which is identical to the '09 except for a higher guaranteed current (4 mA vs. 2mA).

The meter indicates battery condition or transducer status (charge and connection) and the unit's input provides the 4 mA constant current required by all the non-condenser microphones and other transducers in the PCB Piezotronics line. Gain is adjustable to be $\times 1$ (0 dB), $\times 10$ (20 dB), or $\times 100$ (40 dB), and input and output connections are standard BNC, which is convenient for most auxiliary test equipment.

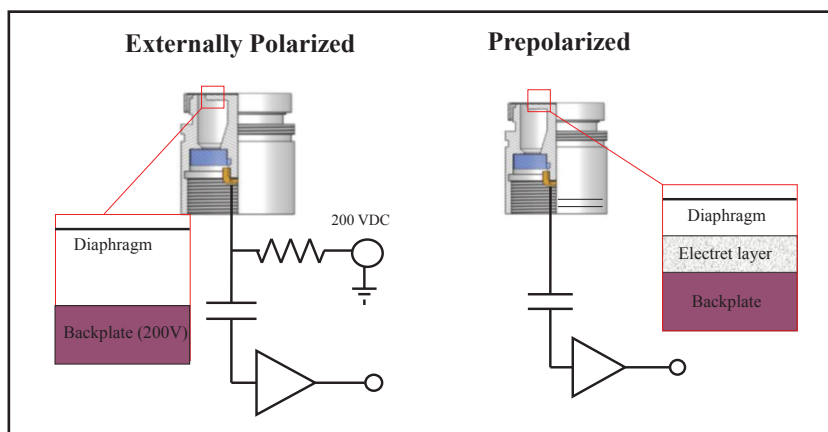


Figure 1: The difference between an externally polarized vs. self-polarized microphone is the source of the DC when the diaphragm is at rest.



Photo 3: The 378A04, a 0.5" self-polarized microphone, features exceptionally low noise and high output. (Image courtesy of PCB Piezotronics)

Figure 2 shows the conceptual schematic. An inspection of the innards shows it to be not much simplified from the real circuit. It comprises a constant current diode, a battery pack, and a switchable gain voltage amplifier. The constant current supply (CCS) diode acts as a remote drain load for the field-effect transistor (FET).

This signal conditioner was used for all microphone testing described here, and the output was connected to the input of my M-Audio Delta 192 sound card. For software, I used a Virtins Multi-Instrument and ARTA, with calibrations of the inputs performed at each startup. The 480M122 performed silently and flawlessly throughout all of the testing. It might not look as cool as a fancier microphone signal conditioner, but it costs a small fraction of the lab preamp I used for comparison and was never the limiting factor in any measurements. Retail price for the 480M122 is \$375.

378A04 Free-Field Microphone

I will be a bit lengthier in my exposition here than for the other PCB Piezotronics products. I am doing so because some methodology common to all my measurements needs to be explained, because there are technological aspects here common to the other PCB Piezotronics microphones, and because this is perhaps the most interesting product of the four microphones that I have on hand.



Photo 4: The BLANK method of noise floor measurement involves wrapping the microphone in plastic and duct tape to protect it, then burying in a mixture of sand and cat litter.

The 378A04 is a 0.5" self-polarized microphone (see **Photo 3**). It features exceptionally low noise (A-weighted noise rated at a remarkably low 5.5 dB SPL typical, referenced to a standard 20 μ Pa level) and high output (typically 450 mV/Pa). Typically, noise this low is only found in microphones with larger diaphragm sizes (self-noise scales roughly as the inverse square of diameter), with the tradeoff being a reduced frequency at which lobing and increased directionality start becoming significant. This microphone also features Transducer Electronic Data Sheet (TEDS) IEEE P1451.4—an on-board chip that identifies the particular microphone being used by serial number. When read by a TEDS-compliant preamplifier or data acquisition system, it can also detect the calibration information.

The pattern is free-field, so this microphone is most suited for measurements of single sources in a largely non-reverberant field. The individual and serialized two-page calibration sheet included with the microphone indicates a microphone sensitivity of 473.29 mV/Pa with an inherent noise of 5.07 dB. The calibration sheet also includes free-field frequency response measured anechoically, shown graphically and as a table displaying frequency (1/12 octave spacing) and magnitude (to 0.01 dB precision).

As part of the traceability standards, reference equipment, calibration data, measurement procedures, and all document control indications are also included. The price of the capsule plus the preamp body is \$2,200.

Noise measurements at that low of an equivalent SPL are not trivial. Even with a high-quality anechoic chamber, which I do not have, getting a background noise level significantly below the 5.5 dB rating of this microphone is a difficult task. In my basement lab, during the stillest of nights, and surrounded by every blanket and towel I could muster, I couldn't get below 20 dB. Besides the ambient noise, it is easy to ruin a measurement just from the vibration of the microphone-to-preamp cable, as well as electrical noise pickup. So, as usual, we have to substitute resourcefulness for resources.

Following a hint from Scott Wurcer, and furiously improvising, I set up my Basement Lab Ambient Noise Killer (BLANK) chamber, using the finest of duct tape, inexpensive storage bins, plastic bags, kitty litter, and sand. **Photo 4** shows the idea: I wrapped the microphone in a plastic food storage bag (Glad), then wrapped the bag and microphone in duct tape in the manner of an impoverished Pharaoh's mummified corpse. Then, I buried the microphone mummy in about 150 lbs of a mixture of veterinary vermiculite (Fresh Step) and sand (Sakrete) in a large plastic storage bin. I placed half the sand/vermiculite in

THE EXPLORER

The Explorer represents the evolution of portable audio, and our refusal to accept the limitations of digital audio. Smartphones, with their mediocre DACs, integrated, noise-prone circuitry and lossy files, just don't do justice to the emotion behind our favorite music — that's why we've carefully crafted the Explorer using nothing but the the highest quality components to recreate a portable listening experience that's as close to real vinyl as it gets on-the-go.

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Authentic Sound — TI Burr Brown PCM1792 DAC and TPA6120A2 amp section recreate a rich, smooth, unusually spacious listening experience.

Connected — WiFi and Bluetooth connectivity allow use of apps, streaming services, and cutting-edge peripherals.

❖❖ **TIDAL** — Every Explorer comes with a free three month trial of TIDAL HiFi, giving you unlimited access to 40 million CD-quality tracks via streaming and locally stored audio.



GET READY FOR SOMETHING DIFFERENT

Echobox is a company founded and run by music lovers and audio geeks who believe that most portable audio gear is either low quality or overpriced, and we want to help audiophiles step into the digital age. We know that people who truly love their music are never satisfied with "good enough," and listening on-the-go should be no exception.

As a company, we believe in the power of music to change the world for the better. That's why we donate 1% of all company profits to charities that use music to make the world a better place.

Find us on www.echoboxaudio.com.

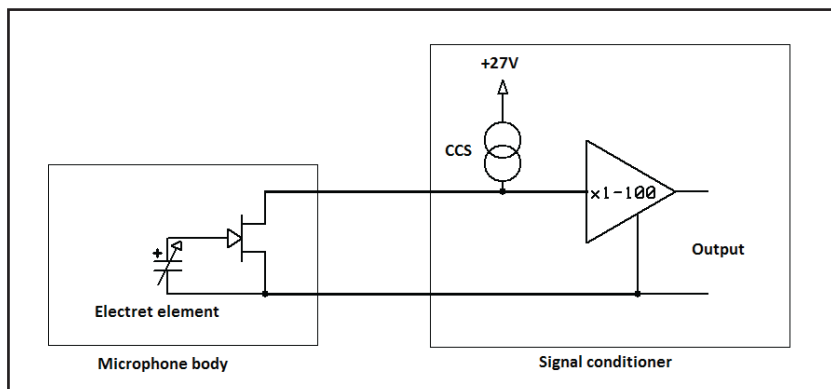


Figure 2: A functional schematic of the microphone and the interface for PCB Piezotronics' self-polarized microphones and constant current supply shows the simplicity of this system.

the bin, placed the mummy on top, and poured the other half of the sand/vermiculite in over the mummy effecting its burial. I filled the inch or so gap between the lid and the top of the sand mix with rags, and ran the cable out through a hole in the top, gasketed by an old sock. I kept the cable between the microphone and preamp as short as possible (under 2') to minimize vibration, and suspended it off the basement floor using more wadded-up rags. I then covered the entire chamber and preamp with a pile of acoustic blankets. In the interest of full disclosure, although I would love to say that I thought of the entire thing at once, I must admit that each of these steps was

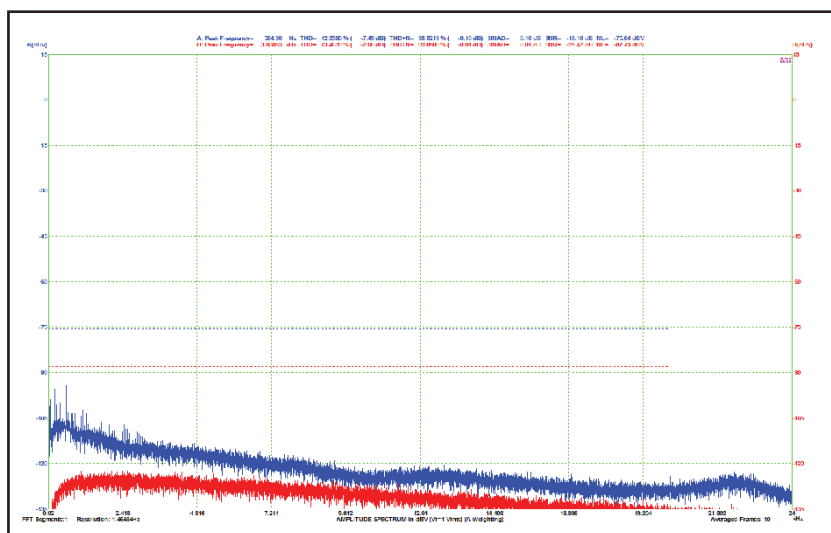


Figure 3: The noise spectrum of the 378A04 microphone amplified by 20 dB confirms the remarkably low noise specification. The top (blue) trace is the microphone. The bottom (red) trace is the measurement system noise floor.

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.

taken sequentially, with measurements confirming that there was a reduction in the microphone output with each "improvement."

As an aside, I should mention that I tried a very thin cable using PTFE dielectric with the idea that increased flexibility should result in lower vibration pickup. Unfortunately, this turned out poorly, but I now know a novel way to make a vibration sensor. I don't know what was to blame, the increased triboelectric potential or the lowered wire-to-shield rigidity of PTFE, but there was no question that the conventional coaxial cable performed better.

I set the preamp for a gain of $\times 10$ (20 dB) to make sure that the microphone, rather than the preamp, set the noise floor. I ran it into the line-level input of my sound card (M-Audio Delta 192), and measured using the A-weighted noise function of the Virtins Multi Instrument software set for a 20 Hz-to-20-kHz bandwidth.

Figure 3 shows the noise spectrum, with the mic channel in blue and the (open circuit input) unused channel shown in red as a reference. A-weighted noise was calculated at -75.64 dBV. The noise floor calculation is then simple: the microphone's output at 1 Pa (94 dB) is -6.5 dBV. There's 20 dB gain in the signal conditioner, so the noise referenced to 0 dB (20 μ Pa) is 4.86 dB. This is a bit lower than the rated noise, but the noise rating is determined at 74°F (23°C), whereas my basement lab is about 58°F (14.4°C). Lower temperatures generally mean lower noise, so this is possibly one source of this minor (0.7 dB) discrepancy. Likewise (and perhaps more importantly), for the equipment used in this measurement (the sound card and the Multi Instrument software), my calibrations are good, but perhaps not to perfectly NIST-traceable standards. Either way, for a 0.5" microphone (the largest diameter suitable for normal audio range measurement), this is a very quiet microphone!

This noise exposition was a bit long, but I think that this is a useful and low-cost method that can be used by hobbyists or small businesses to measure microphone noise without having a specialized chamber. Likewise, each of the isolation steps I took was validated by the microphone, so if I were really trying to hush a room or silence a piece of machinery, I would absolutely choose this microphone.

The tradeoff for the low noise is limited high SPL capability. For this microphone, the distortion specification of 3% total harmonic distortion (THD) is met at 80 dB (20 μ Pa), broadband, and 100 dB when bandwidth limited to under 5 kHz. Even if this specification is conservative, this is a disqualifier for typical loudspeaker or industrial machinery measurement. Fortunately, the SPL rating before damage is greater than 130 dB, so the accidental loud

noise is unlikely to shatter anything. Purely speculation on my part, but this is another factor that suggests that one of the tricks to achieve this impressively low noise and high sensitivity is lower tension on the diaphragm, and thus, greater travel for a given impinging pressure.

I used a Brüel & Kjær 4231 calibrator with a 0.5" adapter, set to at 94 dB (20 µPa), and a 1 kHz excitation frequency to check the sensitivity. I measured 492 mV with the interface box set to x1 gain. This varies from the calibration sheet by 0.3 dB, well within the error limits of the calibrator and PCB Piezotronics' measurement. The distortion might also lead to a slightly higher than rated number.

I measured the frequency response by comparing it to a reference microphone. Since I don't have an anechoic chamber at my disposal, I used a gated response before the first reflections (quasi-anechoic). This limits low-frequency data, so I show response only above 1 kHz. The frequency response shown in **Figure 4** is within 0.5 dB of the calibration data supplied by PCB Piezotronics.

Figure 5 shows the polar pattern data from PCB Piezotronics' anechoic chamber. The microphone is nearly omnidirectional at low frequencies, with increasing directionality at higher frequencies because of the microphone geometry. The usual setup suggestions for non-reverberant measurement apply here—the microphone should be pointed directly at the source so that the high frequencies can be added with correct amplitude. Note the absence of lobing and the smooth transition in directionality with increasing frequency, both of which are necessary for accurate free field measurement.

PCB Piezotronics will be glad to hear that I didn't perform the IEC 60068-2-31 test, which calls for dropping the microphone onto a hard surface from a 1 m height.

378B02 Free-Field Microphone

A more general-purpose 0.5" test microphone will have a higher noise floor than the 378A04, but will also be able to measure at higher SPLs. In the case of the 378B02 (see **Photo 5**), the maximum sound pressure is 138 dB, which will allow its use in near-field measurement. Noise floor is specified as less than 18.5 dB, which is pretty much average for the genre. The 378B02 comprises a 377B02 capsule and a 426E01 preamplifier/mic body. Retail price for this combination is \$1,040.

On test, the noise floor—determined by the BLANK method I described earlier—easily met specification, clocking in at a moderately quiet 15.1 dB, A-weighted, well under the specified 16.5 dB(A) maximum. **Figure 6** shows the measured frequency response above 1 kHz,

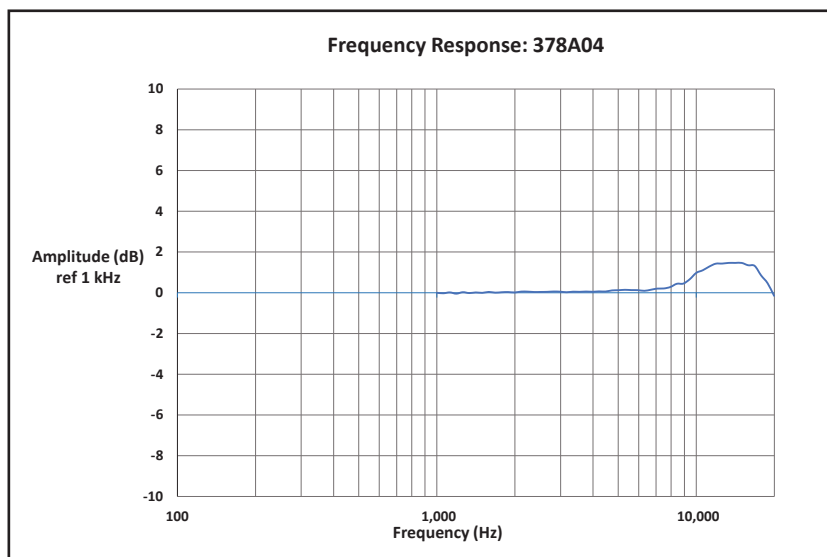


Figure 4: The frequency response of the 378A04 was determined by comparison to a reference microphone. Frequencies below 1 kHz have been cut to account for the removal of reflections.

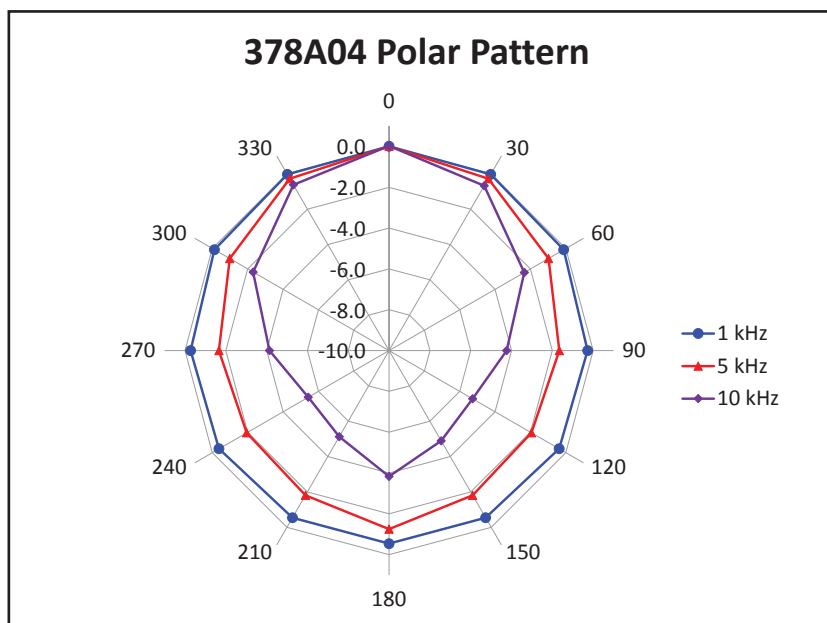


Figure 5: The polar pattern of the 378A04 measured in PCB Piezotronics' anechoic chamber shows a smooth narrowing of the directivity with increasing frequency.



Photo 5: The 378B02 0.5" free-field microphone looks well-suited to precision loudspeaker or other acoustic transducer measurement. (Image courtesy of PCB Piezotronics)

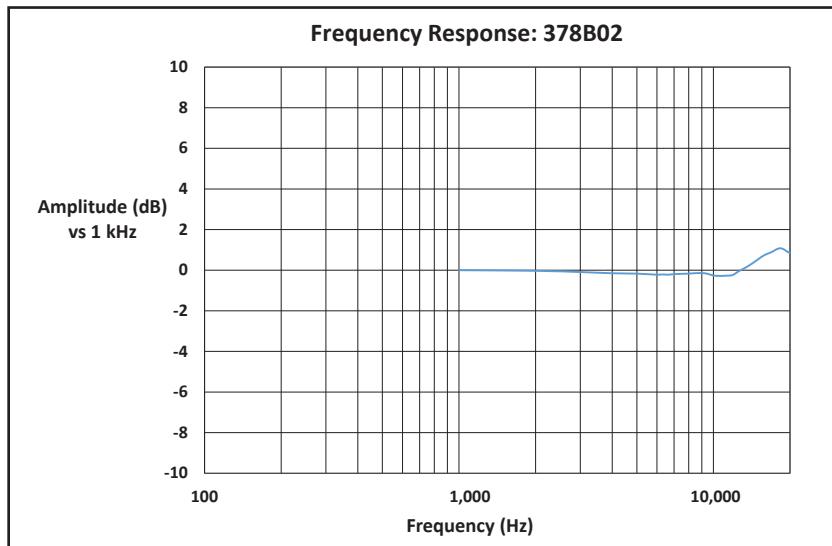


Figure 6: The frequency response of the 378B02 was measured by comparison to a reference microphone and conforms closely to the supplied calibration curves.



Photo 6: The 378A21 0.5" diffuse-field microphone looks like a great choice for measuring explosions, gunshots, or space shuttles, thanks to the microphone's remarkable 160 dB, 3% distortion limit. (Image courtesy of PCB Piezotronics)

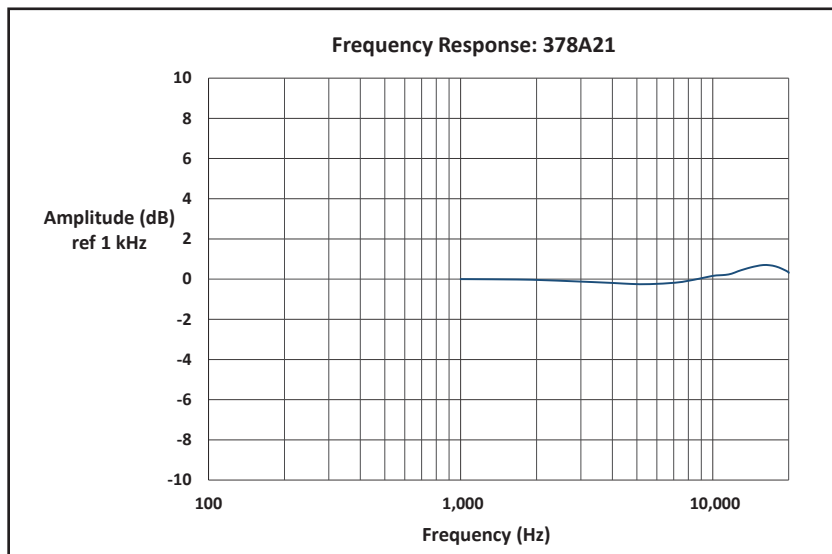


Figure 7: The frequency response of the 378A21, obtained by comparison to a reference microphone, shows excellent conformance to calibration values.

and is within 0.4 dB of the calibration sheet data. Sensitivity at 94 dB SPL (1 Pa) measured 48.5 mV, within measurement error of the supplied calibration. This microphone is suited to precision loudspeaker or other acoustic transducer measurement.

378A21 Diffuse-Field Microphone

The 378A21 0.5" diffuse-field microphone is set up for diffuse-field measurement (i.e., in a reverberant field with sound impinging from multiple sources). It comprises a 377A21 capsule and a 426E01 preamplifier/mic body (see **Photo 6**). Typical applications are sound reinforcement measurement and measurement of noise levels in a factory or other work environment. The specialty of the 378A21 is very high sound levels—its rated 3% distortion point is 147 dB SPL minimum and that's limited by the preamp. The microphone itself is rated to 160 dB SPL. I am simultaneously abashed and relieved that I was unable to test that limit.

What I did instead was check it in a reverberant room with two speakers playing uncorrelated white noise at about 90 dB SPL. **Figure 7** shows the frequency response taken with the microphones at a 70° measurement angle.

Sensitivity measured 11.8 mV/Pa, a bit below the rated 12.08 mV/Pa, but within measurement tolerance. The noise floor measured with the BLANK method was 21 dB, A-weighted, so this would not be the first choice for low noise applications.

However, this looks like a great choice for measuring explosions, gunshots, or space shuttles, thanks to the remarkable 140 dB between the noise floor and the 3% distortion limit. Retail price for the microphone (capsule plus preamp/body) is \$1,100.

130E20 Array Microphone

Microphones yield data at a single point in space, so multiple measurements are needed if you want to map out an acoustic source. Generally, there are two ways to do this—move the microphone to get data at multiple positions (which only works for steady-state signals) or construct an array of microphones to simultaneously measure at multiple points. The latter method is a powerful means of characterizing both spatial and frequency distribution of a sound source and has found use in sonic holography and sound intensity mapping.

The obvious disadvantage of this method is cost. For a 4x4 array of microphones plus the necessary 16 channels of signal conditioning, a test lab would have to lay out something on the order of \$50,000. For this specific application niche, PCB Piezotronics makes a smaller less expensive model, the 130E20 0.25" electret array microphone (see **Photo 7**).

The microphone is only 2.6" long, which helps reduce mass and size in an array configuration. Retail price for the 130E20 (which includes a capsule and a preamp) is \$295, significantly lower than the lab microphones, while still offering individual calibration, high stability, and traceability. Like its bigger and more expensive brethren, the 130E20 is TEDS-compliant. Given the likely applications (acoustic mapping), the pattern for these microphones is free-field. The specified noise floor is 30 dB, A-weighted, about the same noise as a very quiet residential room. Maximum SPL is rated at 122 dB at 3% distortion. Sensitivity is rated at 45 mV/Pa (-29.6 dB).

According to the calibration sheet, the sensitivity of my particular unit was -42.7 mV/Pa (-27.4 dB). Unfortunately, I could not verify the sensitivity since my calibrator does not have a 0.25" adapter. But based on an in-room comparison with the 378B02, that value seems about right. **Figure 8** shows the measured frequency response, which closely compares to the supplied response on the calibration sheet. Noise level using the BLANK method was 28.5 dB, A-weighted, compared to the specified 30 dB maximum.

Wrap-Up

I've been fortunate to have used some fine test microphones over the years, and even more fortunate to have a few of them at my disposal for day-to-day test and measurement use. These are the first professional-grade measurement microphones using current source interfacing that I've used in my home lab. Despite the simplicity and reduced cost compared to conventional condenser microphones, they performed flawlessly, effortlessly, and reliably. Interfacing and setup were simple, painless, and reliable.

The provided documentation was Spartan in appearance, but complete and informative—a model of what test and measurement documentation should be. Methods, calibration, and traceability are complete and traceable to NIST standards, which is one of the things you're paying for in a lab-grade microphone. Likewise, the packaging was plastic boxes and foam, rather than something that looks like a Zero Halliburton case, but what's inside is first-rate. PCB Piezotronics has concentrated on the function, not the facade.

Where I might suggest improvement is in the format for the documentation: It would be nice if all calibration information was provided either as a downloadable digital file or on a thumb drive so it can be directly imported into programs (e.g., Excel). With TEDS-compliant interfaces, this is less of an issue, and when I requested the data in digital format, PCB Piezotronics quickly provided it for me. Bottom line: Packing these microphones up for their return was a traumatic experience for me. 📦



Photo 7: The 130E20 0.25" array microphone offers individual calibration, high stability, and traceability. (Image courtesy of PCB Piezotronics)

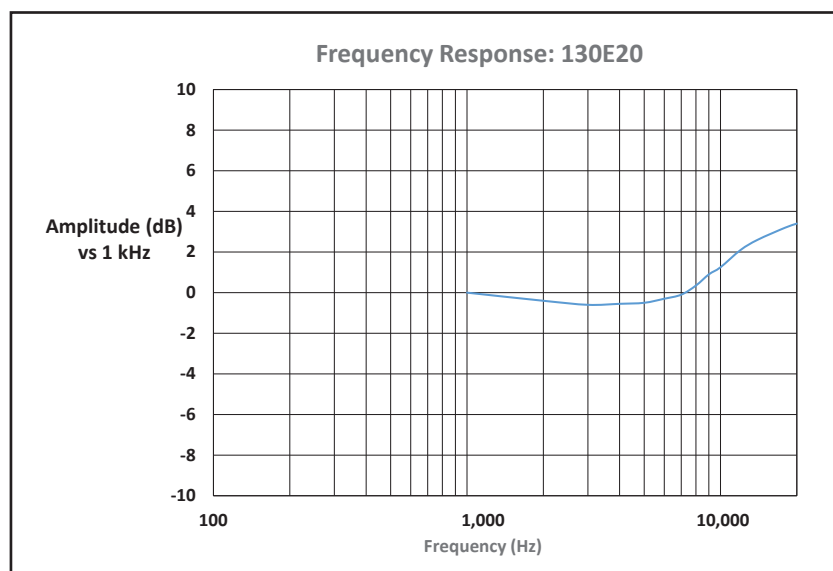


Figure 8: The frequency response of the 130E20 array microphone was measured by comparison to a reference microphone and conforms closely to the supplied calibration curves.

Resources

Microphone Handbook, PCB Piezotronics, www.pcb.com/microphoneHandbookFiles/microphone_handbook_lowres.pdf.

S. Wurcer, "DIY Low Noise Microphone Preamplifiers: Part 1" *Linear Audio*, Volume 1 (available at www.linearaudio.net).

——, "DIY Low Noise Microphone Preamplifiers: Part 2" *Linear Audio*, Volume 3 (available at www.linearaudio.net).

Sources

4231 calibrator with a 0.5" adapter
Brüel & Kjær | www.bksv.com

M-Audio Delta 192
M-Audio | www.m-audio.com

Virtins Multi Instrument software set
Virtins Technology | www.virtins.com

Mytek Digital Brooklyn DAC Preamplifier and Headphone Amplifier

Born in New York



These days DACs are a dime a dozen... or so it seems. There are the inexpensive models for less than \$200, and on the opposite spectrum, the \$7,000 or more examples. But somewhere in between, especially in the \$900 to \$3,500 range, is where I think the real value lies. A recent addition to that middle ground is an exceptional device capable of decoding all the modern formats as well as one so new, only a handful of DACs provide support.

By
Oliver A. Masciarotte
(United States)

In the late 1990s, I was working at Sonic Solutions on the then-new DVD-Audio format. In those days, double and quad speed AES/EBU, at 88.2/96 kHz and 176.4/192 kHz respectively, was exotic and the physical transport layers (PHY) were not yet fully standardized. The only vendors crazy enough to wade into that mess were the real high-end companies—dCS, Pacific Microsonics, and their ilk.

Into that ultra-premium pool swam a newcomer, a little company in Brooklyn, NY, run by a cowboy from Poland. Michal Jurewicz's converter sets sounded great for the price but had maintenance problems. Fast forward to today... The reliability issues are a thing of the past, and Mytek Digital's current stable of products have garnered praise from professionals and consumers alike.

About Mytek

Jurewicz, then a member of Skyline's engineering staff, established Mytek at Skyline Studios in New

York City in 1992. His first product, released that same year, was the Private Q headphone monitoring system for studio musicians. His first converter, in 1993, was an 18 bit mastering ADC, the AD2018. In 1995, Mytek became his full-time gig. Currently, its corporate offices, R&D, and studio are in Greenpoint, Brooklyn, NY. Manufacturing and some R&D is in Poland. That small factory employs 20 people in Warsaw, Poland, with many subcontractors. The PCBs are produced in Belgium, while the fancy casework is fabricated in California, Massachusetts, and Germany.

With Jurewicz's penchant for innovation manifested in his adoption of IEEE1394 (a.k.a. FireWire) and early native support of DSD and Avid's Pro Tools hardware standard, it was no wonder that Jurewicz would jump ahead of the pack and support the emerging Master Quality Authenticated (MQA) format in a reasonably priced USB DAC. Actually, to call the Brooklyn a mere DAC would do it a disservice.

Photo 1: A silver-faced model nicely highlights the user-adjustable logo color.



The Brooklyn, named after the location of the company's offices, is a 1U, half-rack-sized brick of metal, 8.5" deep, with Mytek's signature perforated logo steel top and bottom plates, and a CNC-textured aluminum fascia (see **Photo 1**). The tony case covers are not only pleasing, but provide ventilation as well. Unlike many DACs in it's \$1,995 price range, the product has an integral power supply unit (PSU), which accepts a standard 3 pin IEC 13 plugmold. Mac OS and Linux as well as embedded Linux audio streamers are plug and play. ASIO and WASAPI drivers are provided.

MQA Rendering

When asked how he managed to be the first vendor after Meridian to offer a DAC with MQA rendering, Jurewicz replied that he's had a long-standing awareness of Bob Stuart since the early 1990s. Stuart's numerous Audio Engineering Society (AES) papers with co-authors Peter Craven and the late Michael Gerzon on various aspects of digital audio caught Jurewicz's attention, starting with the evaluation of 20 bit quality, then the Meridian Lossless Packing (MLP) compression scheme developed for DVD-Audio, and later, Craven and Stuart's work on apodizing filters. Jurewicz said that Stuart, "...has been one of the key scientists in the field of digital audio. I have always had a great respect for his work, which has impacted our quest for ever better analog-to-digital conversion."

In early 2015, Jurewicz heard about MQA after its launch at that year's CES. When he found it was from Stuart, he knew they "...might be onto something interesting and original. Subsequently, some time in the Spring of 2015, Alan Silverman (of Arf! Mastering in New York City), a mastering engineer who we often collaborate with when developing new converters, called me saying 'MQA sounds better!' At first I did not get that, I thought MQA was just a compression scheme and found it strange that it would 'sound better,' so I eventually started reading about it," Jurewicz explained.

After Jurewicz studied the patents and read some public information in the popular press, Meridian/MQA had a large display at Munich 2015. Jurewicz attended Stuart's press conference, which included a good amount of scientific theory and discourse. "I felt like I was the only person in the room with sufficient background to grasp all the technical details of the process," Jurewicz explained.

"I actually appreciated these details as it all seemed to fall into place. Later, Bob did a private demo playing Keith Jarrett's 'Koln Concert' before and after. On this closely mic'd recording, I immediately noticed the piano hammers sounding much more realistic and the 'digital hardness' was gone. It seemed like an audible improvement."

About a month later, Jurewicz, Stuart, Spencer Chrislu (MQA's Director of Content Services), and Jeff Dean (Business Development for MQA) met at Silverman's studio in Manhattan for a MQA listen and discussion. "Then Bob handed me a small MQA decoder circuit board with a decoder chip on it," Jurewicz said. "When I saw this, I thought we could just lift that chip and stick it in the Brooklyn. We worked all summer and presented the first third-party MQA DAC at Rocky Mountain Audio Fest in October 2015." By December of that year, Mytek had starting shipping the product. "It was a matter of quick decisions and focus," Jurewicz said, "but I also listened to MQA enough to be convinced about its sound quality first. The business side of it was still a bet, but the risk was worth the effort."

The Brooklyn

Brooklyn's design goals are consistent with Mytek's philosophy of offering the best bang for the buck. Given a budget of \$2,000, there's a lot to



Photo 2: Here is a view of the tidy interior of the Brooklyn.

Resources

"A Comprehensive Q&A With MQA's Bob Stuart," *Computer Audiophile*, April 2016, www.computeraudiophile.com/content/694-comprehensive-q-mqa-s-bob-stuart.

J. Martins, "Master Quality Authenticated (MQA): Redefining the Source for Music," *audioXpress*, June 2016, www.audioxpress.com/article/master-quality-authenticated-mqa-redefining-the-source-for-music.

O. Masciarotte, "Cloak, No Dagger," The Bitstream Column, *Mix Magazine*, August 2002, www.seneschal.net/annex/the_bitstream/bitstream29.html.

Source

Brooklyn DAC

Mytek Digital | mytekdigital.com/hifi/products/brooklyn

offer (see **Photo 2**). Mytek attempted to piggyback on the success of its predecessor, the Stereo192-DSD DAC, which, in 2011, was the first affordable DSD USB DAC. "That feature and price made it extremely popular," Jurewicz enthused, "we sold thousands. I figured we'd try to repeat it with the first MQA DAC with a phono stage. Audiophiles like new things, especially for high-resolution digital."

Software is often the largest stumbling block in a modern audio hardware product, and Brooklyn was no exception. Its digital engine was new, with a 32 bit integer USB Audio Class 2-compliant interface. Most of the heavy software lifting was done in Warsaw, with some help from Germany. Since integrating MQA was also new, Mytek's engineers worked in daily collaboration with their counterparts at MQA in the UK.

Master Quality Authenticated (MQA)

It isn't often that an entirely new, major encoding method is offered for storing digital audio data. It seems that the last widespread attempt to change the file format landscape was the introduction of direct-stream digital (DSD) via the 1999 roll out of the Super Audio CD (SACD) format, and the 2001 debut of Free Lossless Audio Codec (FLAC). Last year, it happened again. Led by Bob Stuart, the same team that brought us Meridian Lossless Packing (MLP) has come up with some new, novel, and clever data manipulation schemes, and this time it's all about steganography. That obscure term refers to hiding information, in plain sight, within other information. In the case of audio, the data is there to hear, as long as you know where to look and listen. A now rare example of audio steganography is watermarking for asset tracking.

In the case of MQA, high-resolution audio (HRA) data is hidden as pseudo-random noise in the lowest order bits of linear pulse-code modulation (PCM) data words. To a non-MQA player, the file "looks" like Red Book standard audio. If a decoder is present, then the hidden data is decoded or "unfolded," resulting in a true HRA virtual file. This means that MQA files, and their resulting bit stream when played back from storage, are equivalent to files ripped from a CD. The decoder renders the HRA output in real time from the MQA data packed into the file.

The trouble with hiding data within the audio payload is its fragility. Any alteration to the MQA bitstream will prevent the decoder from functioning. That means no EQ, gain changes, redithering or any DSP. Upsampling, thank goodness, is rendered obsolete by MQA as a high sample rate master file, once encoded, will be transformed into an easily streamed or stored 44.1 or 48k file. Decoded, it blossoms into the high sample rate data it once was.

The other advantage I like is the MQA encoder's ability to compensate or pre-equalize for group delay occurring in the



Bob Stuart's press conference explaining MQA at the High End Munich show in 2015.

production chain. The encoder has several preset values that an engineer can choose from, based on the ADC hardware and file provenance which cancel, whole or in part, the temporal blur inherent in the digitizing process. This feature benefits those with or without an MQA decoder, as it's "baked into" the file during encode. MQA decoding will also correct for a DAC's additional temporal blur at render time.

As this is new technology, there are obstacles to be cleared. One is selection of less than optimal de-blurring settings for the encoder, which was probably responsible for one of my album's razor sharp sibilance and high end upon decoding. Often times, encoding is not ideal, as the temporal correction aspect of the process can only be estimated and/or applied after the recording has been made. Experience with the final release toolset should cure that. Another subtle, more chronic issue is isolated industry reluctance to jump on any bandwagon driven by Meridian, et al. I can only speculate what the cure for that might be.

As with any modern DAC, the Brooklyn can handle linear pulse-code modulation (LPCM) up to 384 kHz, along with native direct-stream digital (DSD) up to four times DSD at 12.288 MHz on the aforementioned USB interface. Pros are not left out of the mix, with a critical Word Clock (WC) output, and input, on BNC. WC Out enables Brooklyn to act as the clock master for an entire digital audio suite, and is a requirement when using an analog signal chain during mastering.

For DSD playback, three anti-image filters are available. Each is tailored to the data rate: 1, 2, or 4 times. Though they are user selectable, Mytek recommends you leave that setting in auto mode and let the DAC choose for you, based on the stream rate.

As I hinted earlier, the Brooklyn does quite a bit more (see **Photo 3**). It acts as a preamplifier, using an analog gain section to cleanly scale inputs. It also has several AES3 digital inputs, including balanced on an XLR3F and two user-selectable serial coax connections on RCA. All of the AES3 inputs accept LPCM up to 192 kHz while the RCA can be switched between AES and baseband DSD DOP. It even has a "TOSLINK" F05 plastic optical fiber input



for legacy components, like that DAT machine you keep meaning to relegate to e-waste just as soon as you transfer your collection of tapes. In addition to the WC input, any of the digital audio inputs can be selected as clock reference and the DAC will slave to that signal.

The DAC also does vinyl playback, utilizing its "precision" phono preamplifier. When it comes to vinyl playback, my setup is currently in total disrepair. Because of that sad state, I was unable to realistically evaluate the built-in phono preamplifier. The switchable, moving coil/moving magnet, low noise phono preamplifier design is based on Mytek's respected microphone preamps. In moving magnet mode, the input impedance is 47 k Ω , and the gain is 45 dB relative to -20 on the fader. With moving coil,

Photo 3: This is a black-faced version of the Brooklyn, with its business end below.

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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.

you get 65 dB of gain, and an input impedance of 100 Ω . The de-RIAA filter is analog. Due to it's wide band behavior, Jurewicz mentioned that, "Particularly with lightweight M/C cartridges, tracking transients is better than typical (preamps)."

Brooklyn also includes two independent

unbalanced headphone outputs with balanced internals. The device can be configured to feed signal only to the main outputs, only to the headphone outputs, to both at the same time, and to your cans while automatically muting the main outs. Headphone gain is separate from main output gain, which enables you to quickly switch between loudspeaker and headphone playback once your levels are set.

An Apple remote is included in the package, as is a quotidian USB cable. Your favorite RC5 infrared remote can also be used, although I didn't test that feature either.

Putting Brooklyn to the Test

At the AXPONA show in April of this year, Michal Jurewicz sent me home with a shiny new, silver Brooklyn. The next day, I updated the unit to the 2.05 firmware that woke up the MQA functionality. During this review, I continuously updated the unit to 2.20, eight revisions beyond what the unit was initially running. The update was accomplished through the free Mytek Control Panel utility, and took all of six seconds. The Control Panel replicates the front panel controls but, due to a lack of a horizontal scroll bar, required more screen real estate than my 1920 x 1080 pixel display could muster. Fortunately, the firmware update function packs everything into the upper left corner of the utility.

As promised, with my player apps set to bit-perfect mode (no gain change, EQ, or redithering), the MQA indicator illuminated when I played MQA-encapsulated content. A very helpful feature for evaluating MQA mastering is Brooklyn's ability to disable rendering, defaulting to what MQA calls "legacy" playback mode, where an MQA file is treated as a Red Book version without rendering any additional high frequency and/or low amplitude content that may be hidden in the embedded MQA datastream.

Since I've had a Stereo192-DSD mastering version for quite some time, I was able to compare that model to the Brooklyn. The Stereo192-DSD tended toward a lean, factual

delivery, while the Brooklyn simultaneously provides a more realistic presentation without euphonious color. My guess is that the Brooklyn's jitter is lower.

While the Stereo192-DSD did not possess a reference-grade headphone amplifier, the Brooklyn's headphone amp clearly bested the previous model's version. I also compared apples to oranges by level matching and listening to the headphone amps of both my exaSound e22 mkII and Woo Audio WA7 Fireflies 2nd Generation. Through my Audeze LCD-3 cans, the Brooklyn had a more open and linear frequency response than the hollow-state competition, though the big bottom and euphonious bloom of the WA7 Fireflies is always fun. When comparing the Brooklyn to my reference e22, the exaSound's slightly more open sound and better low amplitude detail bested the Brooklyn, but the latter was certainly no slouch. From memory, I'd have to say that the Brooklyn's headphone output was easily on par with an SPL Phonitor I recently sold. The latter is a physically large, high voltage, pro headphone (only) amp, so having that performance built into the Brooklyn is yet another value add.

Line Level Comparisons

Moving on to line level comparisons was equally interesting. Most DACs have a signature sound, subtle though it may be until you make a direct comparison. The Brooklyn offers a far more linear and open response than the majority of DACs in its price range, even if you discount the formidable array of extras it offers. Coming from a pro audio background, Mytek tends toward unhyped delivery of the bits, without color or attitude. In this regard, the Brooklyn excelled where far more costly pieces, especially those voiced by enthusiasts rather than engineers, fail to reveal the flaws of a recording as well as the flattering features. Brooklyn's bottom octave, an issue for me with the Stereo192-DSD, was extended, solid, and fast. Its ability to paint a broad, deep, and detailed soundstage exceeds what I've heard



The large knob at the right side of the Brooklyn's front panel is both the menu selector and the controller.



The default display mode shows the MQA decoding is active.

from the vast majority of stand-alone “renderers” out there.

Fidelity

You might be wondering what an extra \$1,200 buys you in terms of fidelity... The ability to tease apart microdynamics and complex harmonic structures are slightly less refined with the Brooklyn when compared to the e22. There was a thickness to massed choral voices that the more expensive piece did not exhibit. Also, separation of instruments with the Brooklyn was not quite as refined.

Usability

Something I should mention about the Brooklyn—its excellent usability. The large knob at the right side of the front panel is both the menu selector and the controller. There are four small buttons arrayed along the front panel's lower edge, each correspond to a section of the color display when it's in what I'll refer to as “tweak” mode. By pushing one of the buttons, you're selecting the parameter in the corresponding section of the display. The big knob lets you alter the value and, with a push of the small button, enter that new value into the device. It's quick and easy to peruse, select, change, and accept new settings without excessive scrolling. When you're done tweaking, a push on the big knob reverts the display to showing just sample rate, gain, and MQA decoding status. A very nice aspect of that big knob... in default mode, it's a continuous, remote-able analog gain control, a feature many DACs lack, including my e22.

Additional Features

About those extras I mentioned earlier, the principal one is MQA decoding... An awful lot has been written about MQA, its advantages, disadvantages, and sound quality. My personal take listening to a variety of encoded files from over a half dozen labels, using the Brooklyn in decode mode with all player DSP disabled, was one of appreciation in every sense of the word.

I enjoyed the good qualities it brought to my listening, and was pleased by the aesthetic value it offers. In decode or render mode, a thickness to envelopes is removed, while the soundstage and phantom imaging becomes respectively deeper and far more precise. Transients became finely wrought while low amplitude details (e.g., reverb tails) become more defined and expansive. Note decays are simply audible longer. Certain engineering techniques, such as close mic'd and panned elements or artificial reverb in a mix, stand out as

anomalous and disjointed more when the decoder is enabled. And, though I've heard a report from an engineering colleague that un-decoded legacy playback of encoded files resulted in nastiness, I have heard no such thing while investigating that data point.

Test Equipment

The source for all my tests was an El Capitan Mac mini running my two reference software players, Amarra Symphony 3, and Audirvana Plus 2.5. The balanced out from both DACs was connected directly to my Lindell AMPX Class A solid state amp via Soundstring GEN II Beta 2-22S, while the DACs were connected to the host with my reference VUE Digital VU-3 USB cable. AntiCable's signature Level 3 red wires fed, from the AMPX, a pair of Spatial M3 Hologram Turbo open-baffle loudspeakers. Power was supplied through Soundstring GEN II Digimax-18 cables from a custom, high current mains conditioner. For all these comparisons, I used my software player's gain controls to level match the units so as to avoid changing the hardware's volume control.

Final Opinion

Now that the product is shipping, the Mytek team is working on two new projects. Manhattan II, Mytek's flagship DAC will soon acquire MQA decoding, and is scheduled for release before the end of 2016. Jurewicz and his teams are also working on developing an end-to-end mastering chain for MQA workflows. A Brooklyn ADC, complementing the DAC, will also incorporate an “MQA Kernel.”

Brooklyn represents an affordable leap forward in digital audio decoding. With its linear, neutral sound, plethora of additional features, and ease of use it should be on your short list when auditioning converters. Jurewicz's assured me that “...we always try to make the next product sound better than previous generations. We are working (on all) this now as we speak,” and I can only say that I'm enthusiastically looking forward to it!



My test rig is displayed, with the host computer peeking out just above the keyboard.

Retro-Style High-Fidelity Preamplifier

Having completed a phono preamplifier project that I was pleased with, I decided to build a full-featured preamplifier with volume, balance, and tone controls.

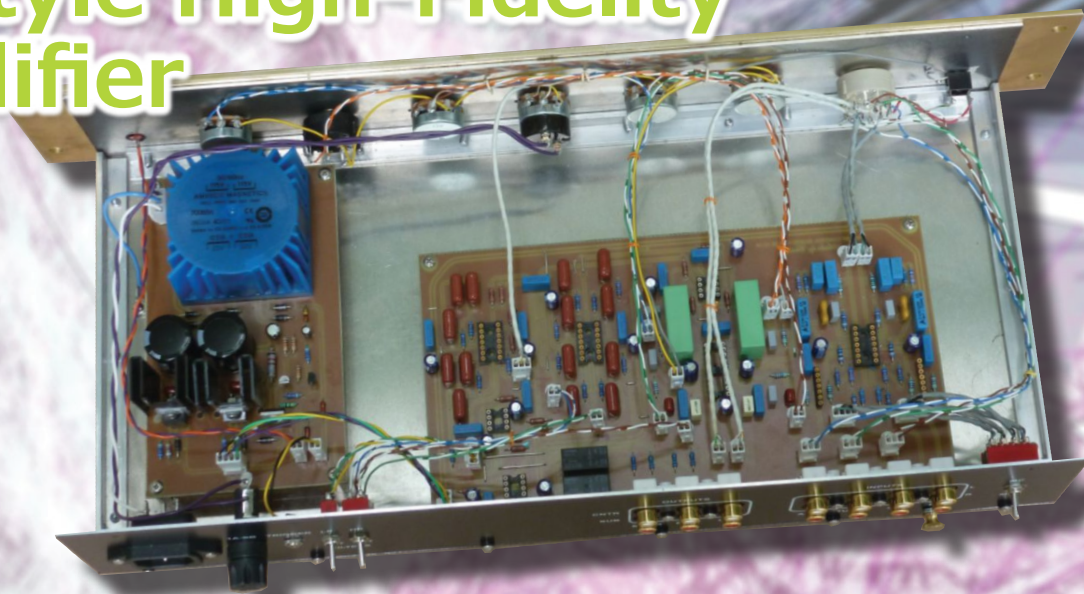


Photo 1: This is the inside of my retro-style full-featured preamplifier with volume, balance, and tone controls.

By
Bill Christie
(United States)

To begin my preamplifier project, I had to choose a chip. During my search, I found that Texas Instruments (TI) provides a chip, the LM1036, which greatly simplifies the design. However, I decided to add several features to the basic design published in its datasheet. First, I incorporated the phono preamp. Next, I added a derived center channel output by summing the right and left channels. This was a feature found on a Fisher preamplifier that I owned and found useful. And finally, I included an active crossover to provide a subwoofer output. I also provided switches to defeat the crossover (see **Photo 1**).

The Design

The basic components of the design are shown in **Figure 1**. The output of the phono preamp feeds the input selector along with the line level inputs. The selector switch in turn feeds the inputs of the LM1036. The LM1036 then branches to feed a stereo high-pass

filter and a summing amplifier tied to a low-pass filter for the subwoofer output. Summing the right and left channel output provides the center channel output. If a subwoofer is not used, the high-pass filter should not be used. So, I included switches to bypass the filters.

Preamplifier Circuit

Figure 2 shows the schematic of the phono preamplifier, which is essentially the same design as the one I used for my preamplifier design, which I shared in "Build an External Phonograph Preamplifier," (*audioXpress*, July 2013). A complete description of this circuit can be found there. The only change I made was to increase the gain of the second stage to make the output level more compatible with the output of my DVD player. This was accomplished by changing the feedback resistors, R21 and R22, from 3.32 k Ω to 4.7 k Ω .

The outputs of the phono preamp feed a front panel input selector switch (S2) that switches between phono and four line level inputs. The selector switch then feeds the inputs of TI's LM1036 control IC.

Figure 3 shows the heart of the preamp using the TI chip. This chip has an internal reference voltage that feeds four linear potentiometers for controlling volume, bass, treble, and balance. The output of the pots feed back to the chip and provide the necessary control voltages. I used the component values specified in the datasheet. However because this chip does not provide gain, I added a stage with a gain of 2 (approximately 3 dB).

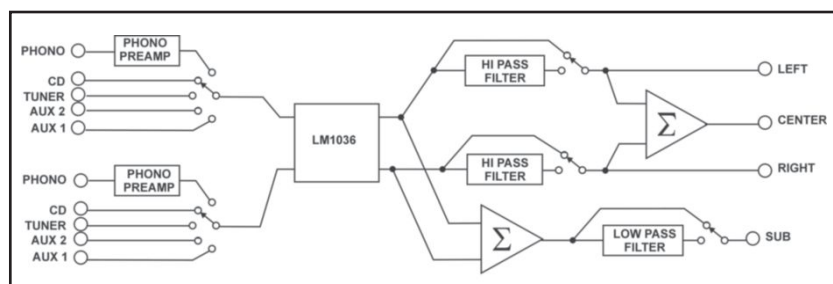


Figure 1: The design's basic components are shown in this block diagram.

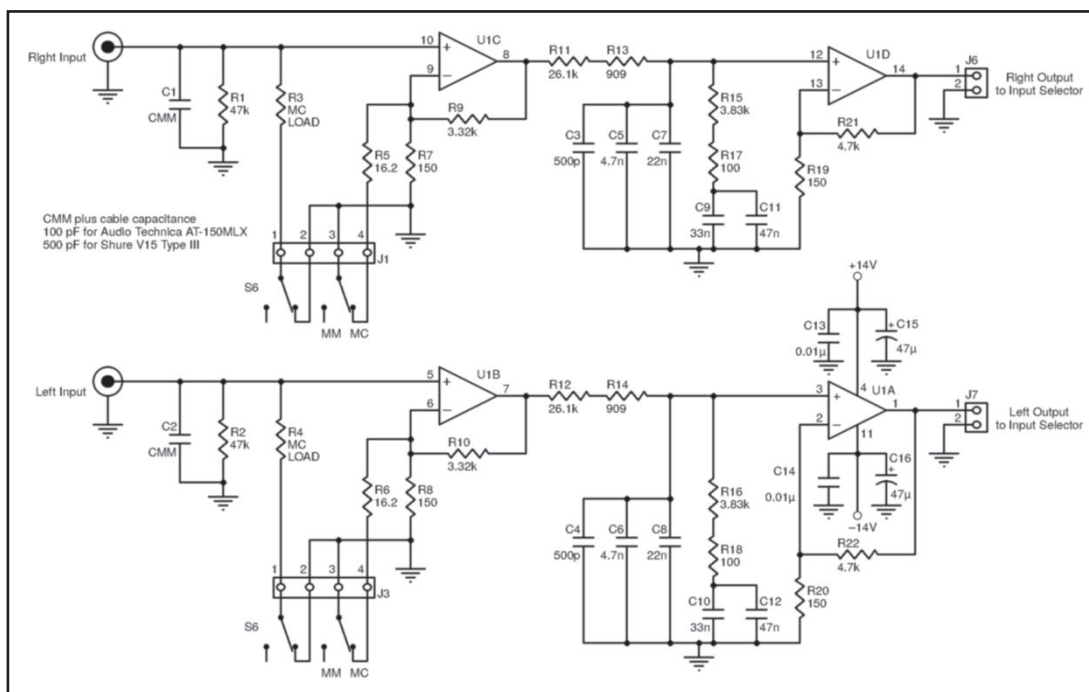


Figure 2: The schematic for the phono preamplifier is essentially the same one I used for the preamplifier I described in my article, "Build an External Phonograph Preamplifier," (*audioXpress*, July 2013).

Figure 4 shows a schematic of the circuitry used to derive the center channel. The summing amplifier is inverting so I added an op-amp to re-invert the signal and provide some gain to allow the proper range of adjustment for balancing the output with the left and right channels. The center channel level control pot is linear with a resistor, R14, added to approximate an audio taper. I also added another op-amp to buffer the output, which isolates the circuit from any load from the external power amplifier. Note that the master

volume control raises and lowers the center channel. The potentiometer (P5) allows the desired balance between right left and center speakers.

The Filters

The filter shown in **Figure 5** is an 80 Hz stereo high-pass filter. Each channel is a Linkwitz-Riley fourth-order high-pass filter that provides a 24 dB/octave rolloff. The switch (S5) provides a selection of filtered or unfiltered signal to the output jacks and the center

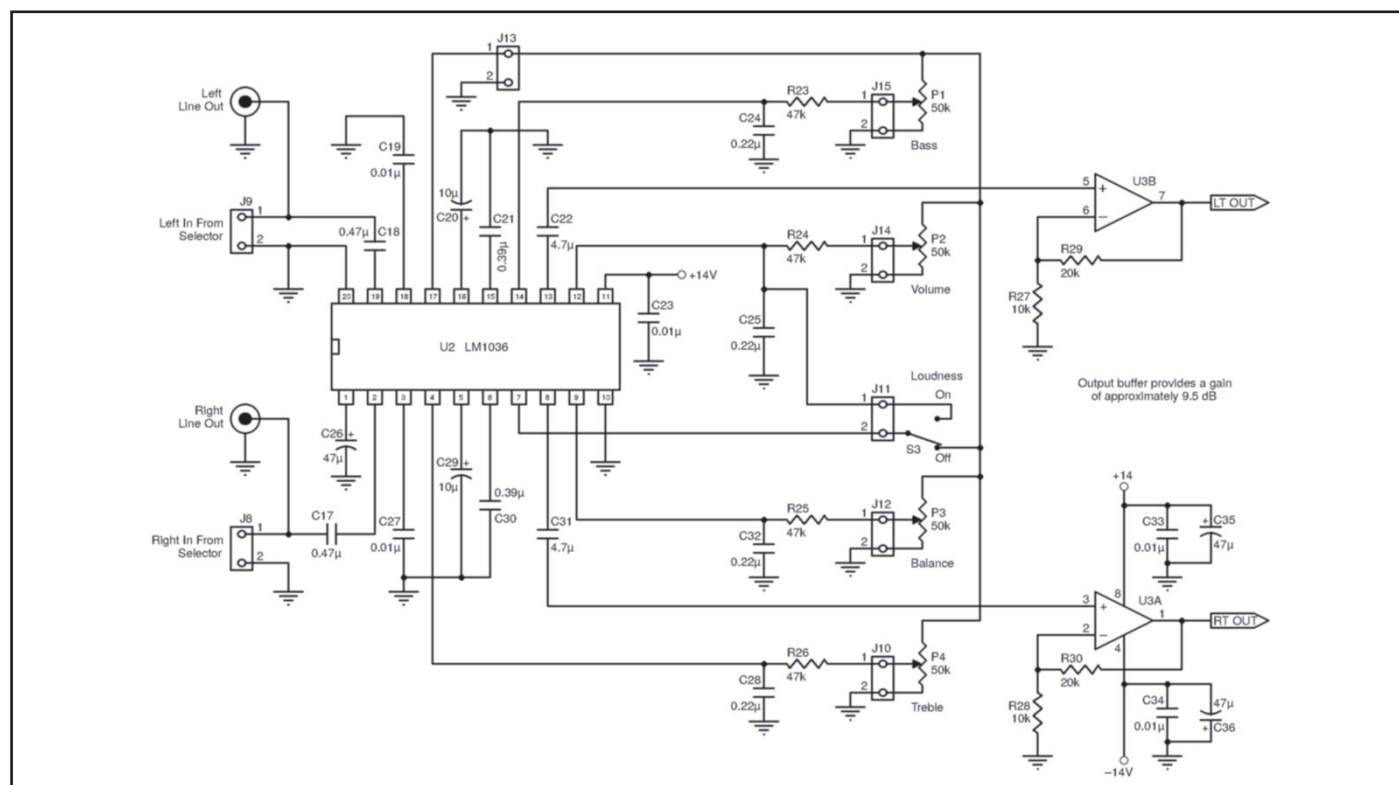


Figure 3: Here is the heart of the preamp, using the Texas Instruments chip.

Figure 4: This is the circuitry used to derive the center channel.

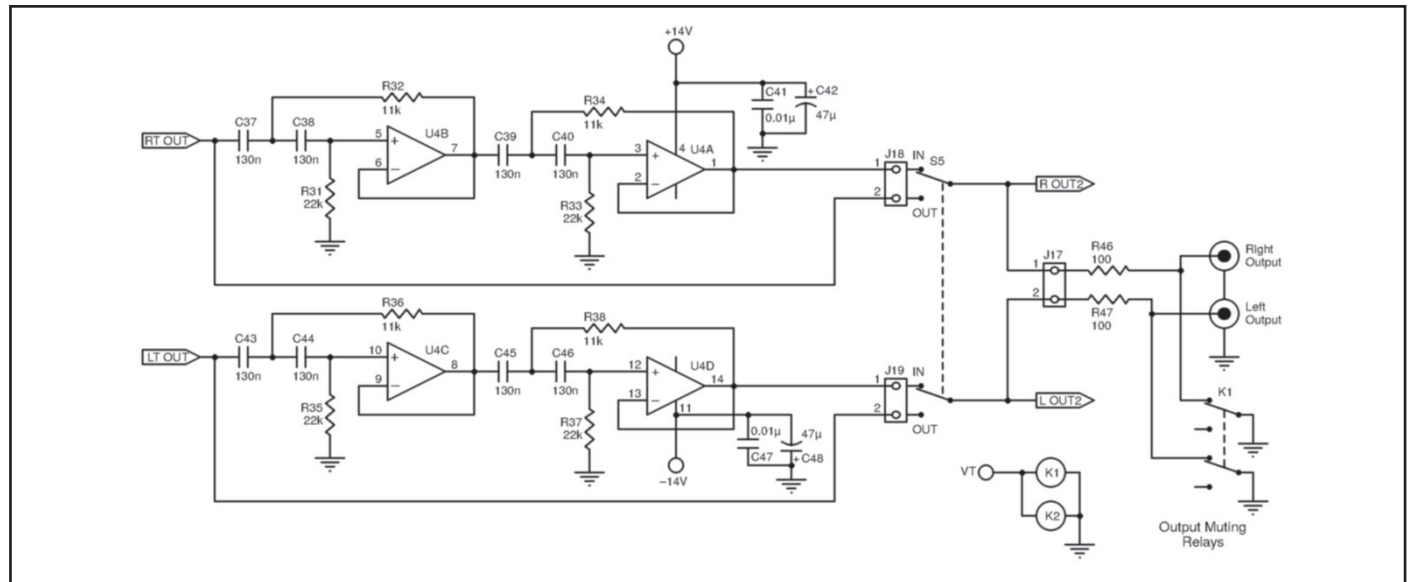
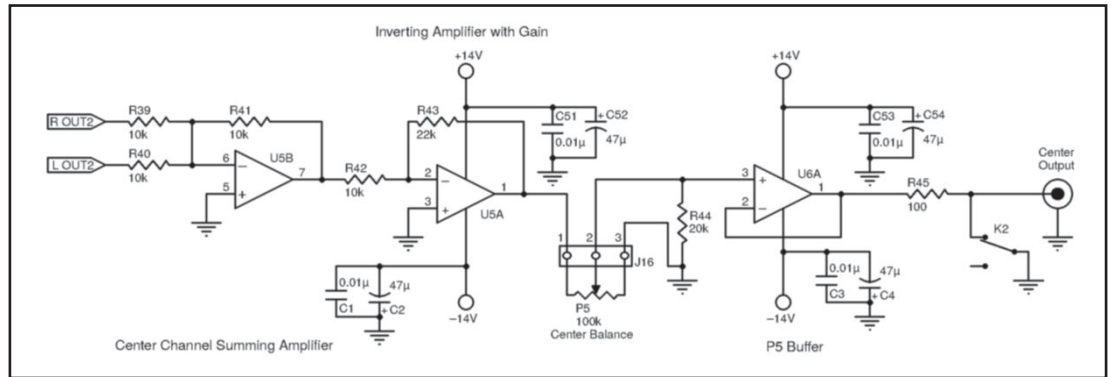


Figure 5: The high-pass filter is an 80 Hz stereo high-pass filter.

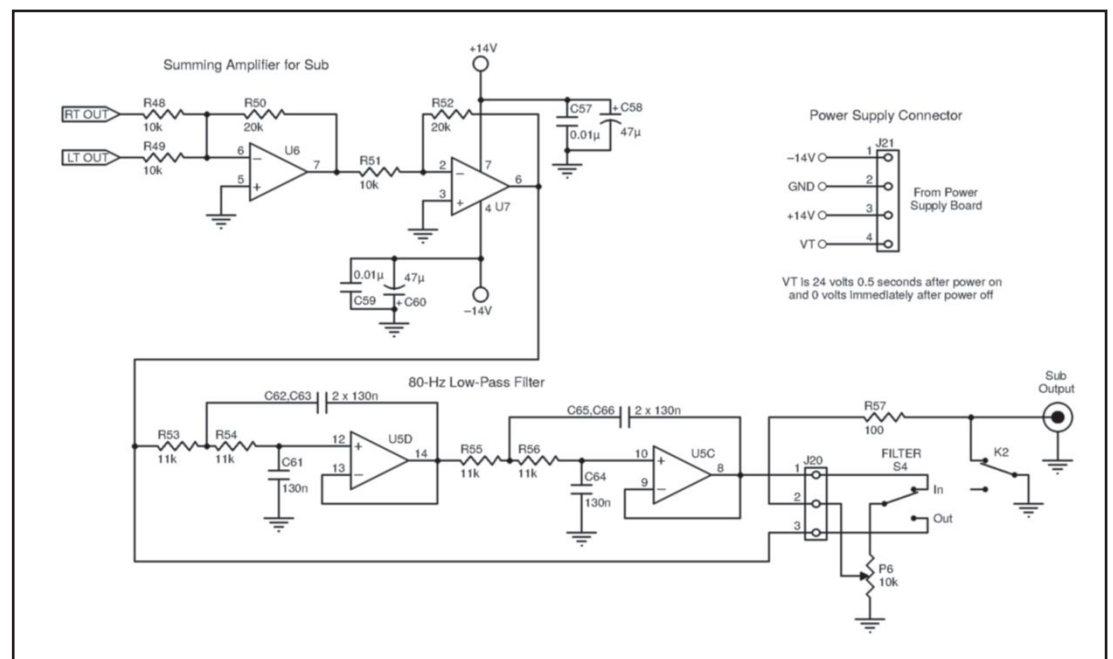


Figure 6: This is the low-pass filter.

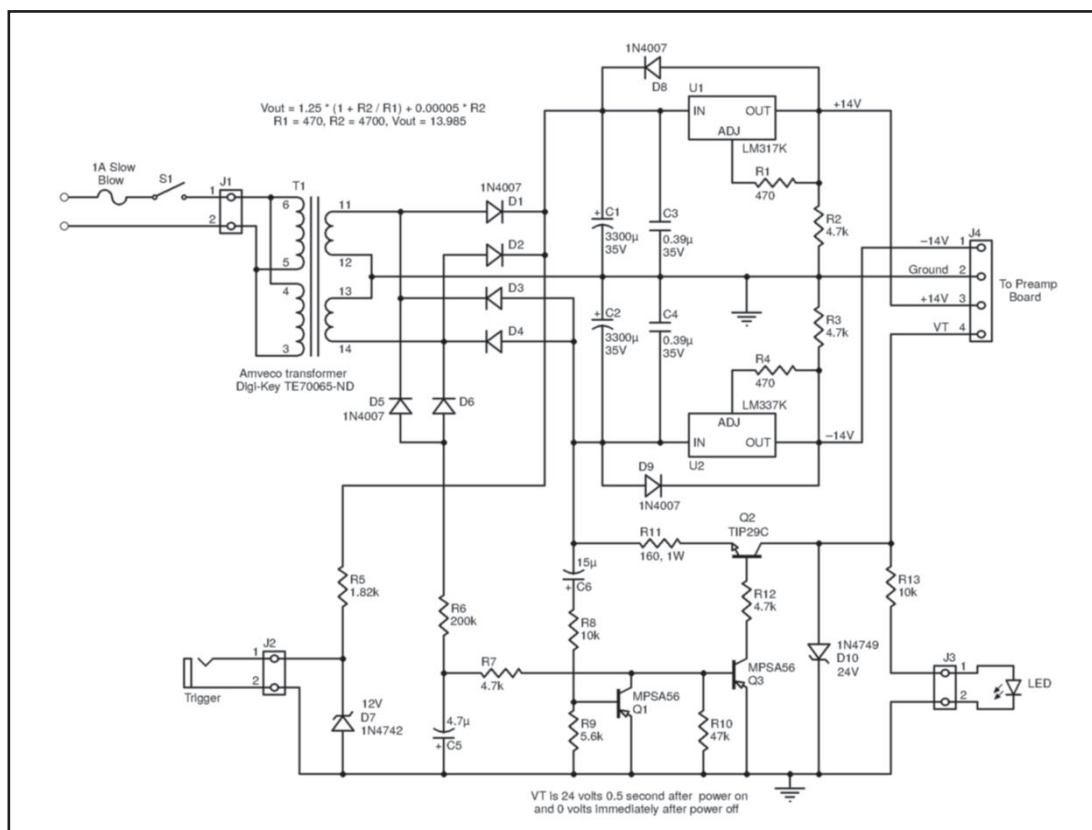


Figure 7: The power supply is on a separate PCB, which could be used in other applications requiring the same voltages.

channel summing amplifier. The relays, K1 and K2 mute the output during power up and power down. The power supply provides an on-delay voltage to power the relays. There are some very objectionable sounds that occur without the muting function.

Figure 6 shows the schematic for the low-pass filter. The unfiltered left and right signals are summed with an inverting amplifier with gain of 2 and fed to an inverting amplifier also with gain of 2 that in turn feeds the 80 Hz low-pass filter.

A single output is provided for a subwoofer. This is acceptable, as direction of the sound below 80 Hz is not perceptible. The filter is a fourth-order Linkwitz-Riley low-pass filter. The nice thing about the Linkwitz-Riley design is that the signals are phase coherent at the crossover frequency. S4 selects between a filtered or unfiltered output.

As with the center channel output, the subwoofer output is controlled by the master volume control. The overall gain of the low-pass filter is 4. This is to provide enough output to match the subwoofer level with main speakers. A basic power amplifier with level control can then be used to power the subwoofer.

Power Supply

The power supply is essentially the same as the one I used in the phono preamplifier (see **Figure 7**). The power supply is on a separate PCB that could be used in other applications requiring the same voltages (i.e., electronic crossovers, headphone amplifiers, etc.). I changed the delay circuit to operate from the negative supply to better balance the load. (The LM1036 operates from the positive supply.) The timed output delays a

24 V supply by approximately 0.5 seconds after power up and turns off the supply almost instantly after shut down. The resistor R11 and Zener diode D10 are used to provide a relay voltage to 24 V. These may have to be adjusted if used in a different application with different relays.

To keep high currents out of the preamplifier, the switch (S1) only directly powers on the preamp. I added a 12 V trigger output that can be used to active a solid-state relay to switch on power amplifiers. Many power amplifiers include a trigger input for that purpose.

Fabrication

The design is implemented on two PCBs. The main board is 9" x 5" single side 1/16" material (see **Figure 8** and **Figure 9**).

About the Author

Bill Christie developed an interest in electronics at the age of 6 after his father helped him build his first crystal set. Since then he has built numerous kits. Bill started building amplifiers from scratch while he was still in high school, starting with a mono tube amp designed by Mullard. He graduated from the University of Akron in Ohio with a BS in Physics. Bill is now a retired electrical engineer from Goodyear Tire & Rubber Co., who spent 35 years designing controls and instrumentation for manufacturing.

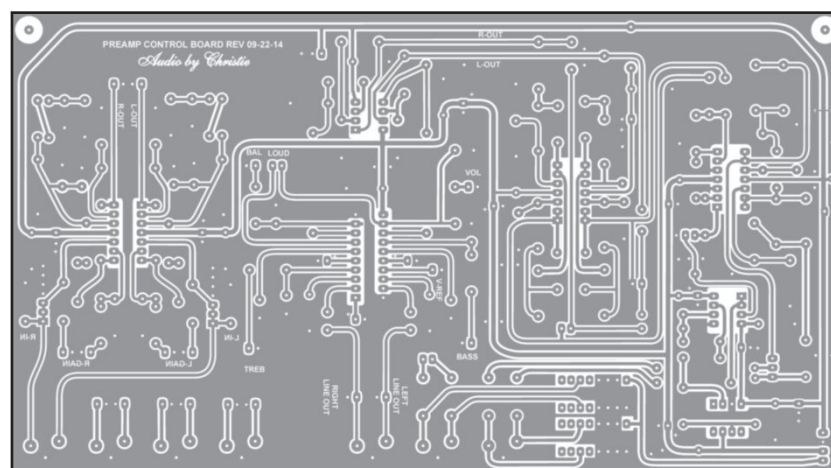


Figure 8: This is the main PCB.

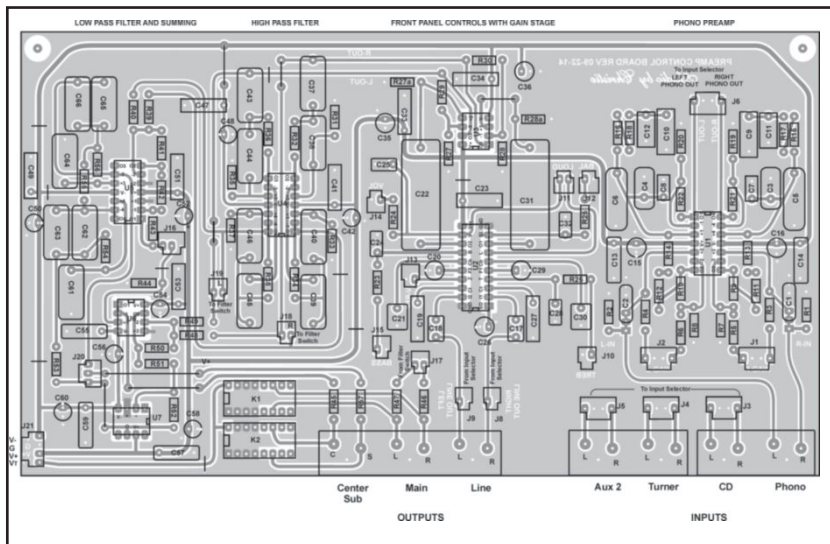


Figure 9: Here is the main board component layout.

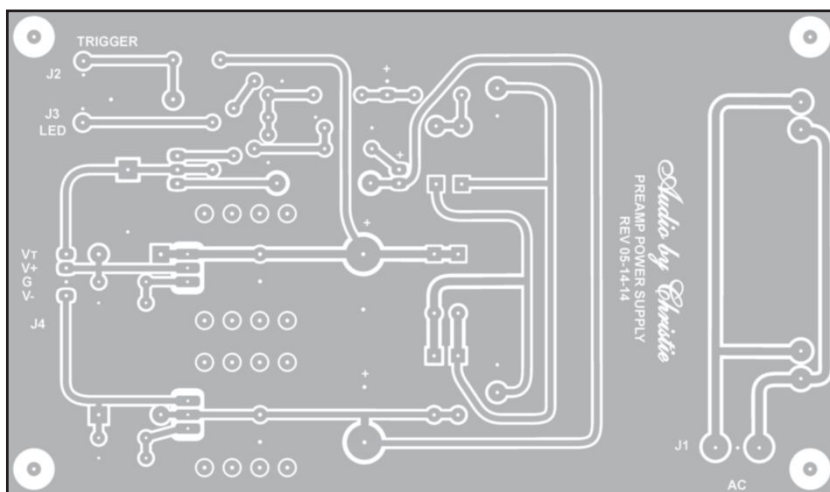


Figure 10: The complete power supply is implemented on a second board.

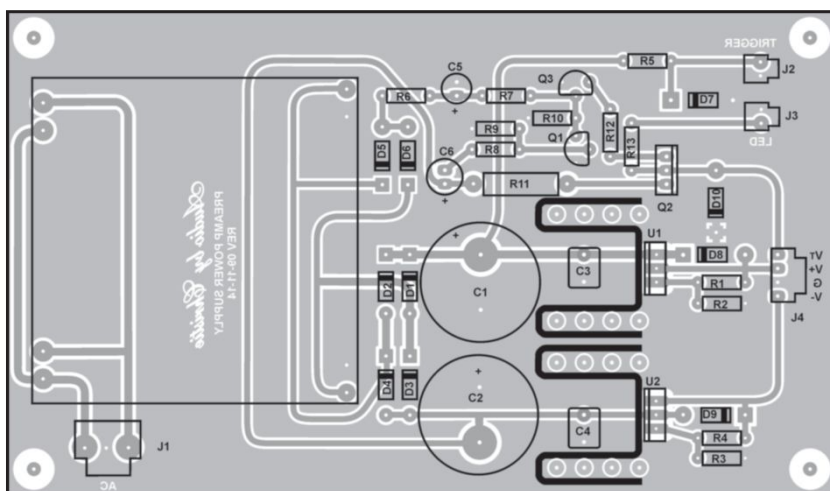


Figure 11: Here is the power supply component layout for the second board.

The input and output RCA phono jacks are mounted on the PCB and provide support for the back of the board. The front of the board is supported by 1" long standoffs. All connections to the board are made with Molex connectors so it is easy to remove, if necessary. All of the ICs and the muting relays are socketed. The holes for the passive components are spaced to match the components in the parts lists. The parts lists for the power supply, the preamplifier board, and the miscellaneous materials can be found in the Supplementary Material section on the audioXpress website (www.audioxpress.com). All ground connects are made via the background, which serves as a ground plane. A number of jumpers are placed to connect "islands" or areas of the back plane that are otherwise isolated.

The complete power supply is implemented on a second board that measures 6" x 3.5" (see **Figure 10**). The power transformer is a PC mounted toroid. The voltage regulator ICs are heatsinked. As mentioned earlier, it is completely separate from the main board and could be used for other projects.

I fashioned the chassis from 1/16" aluminum panels fastened together with 0.5" aluminum angle. It measures 17" x 8" x 3.375" with a 19" dress panel. Placement of the holes for the RCA phono jacks located on the back panel is critical since the jacks are mounted on the printed circuit card. **Figure 11** shows the required layout.

Figure 12 shows the back panel hole layout. The placement of the holes for the AC power connector, fuse holder, trigger jack, and filter switches is not critical other than avoiding the PCBs.

The placement of the holes on the front panel is also not critical. I evenly spaced seven holes for the controls and placed two holes on the lower outer areas of the panel—one for a power LED and one for a stereo input jack (see **Figure 13**). The dimensions shown are for the front subwoofer panel. A 19" standard size rack panel

Project Files

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

Resources

B. Christie, "Build a Powered Subwoofer," *audioXpress*, October 2015.

—, "A 3.1 Channel System," *audioXpress*, April 2016.

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.

is fastened to this subwoofer panel using the controls. As mentioned before, wiring is facilitated by the use of Molex connectors.

Chassis Wiring

Figure 14 shows the chassis wiring. **Photo 1** provides an inside view of the completed preamplifier. To label the front and back panels, I went back to a company that I've used before (Enlarging Arts in Akron, OH). The company makes adhesive-backed decals that are ideal the job. They have durable vinyl film laminated to the front to protect the lettering. They look as good as any commercial product I have seen. The cost was reasonable and I would recommend them to anyone who has a custom-labeling requirement. The custom label finishes off the project and gives it a more professional look (see **Photo 2**).

The System

As a companion to the preamplifier, I built a power amplifier based on TI's LM3886 three-channel power amplifier. With proper power supply it produces about 35 W per channel. I connected it to three small speaker systems, using the Dayton Audio RS100-8 4" full range drivers and a powered subwoofer. The sound is amazing, rivaling that of my home-theater system. It is ideal for my electronics hobby room. 

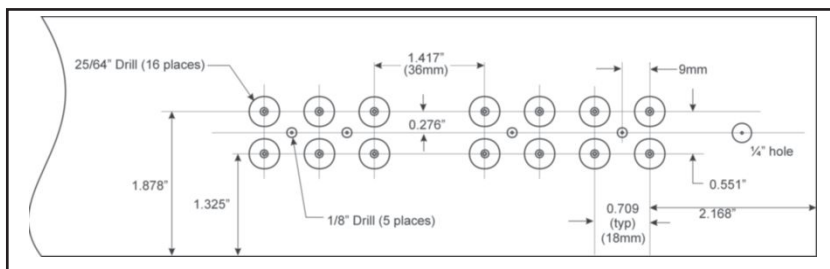


Figure 12: The placement of the holes for the AC power connector, fuse holder, trigger jack, and filter switches is not critical other than avoiding the PCBs.

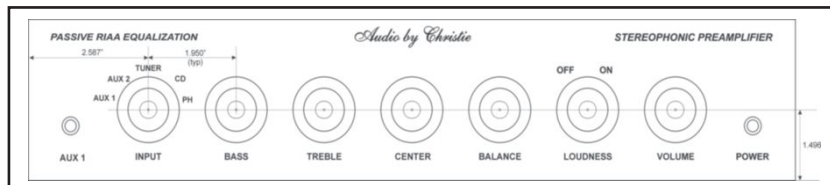


Figure 13: The placement of the holes on the front panel is also not critical.



Photo 2: To complete the project, I had custom labeling made for my retro-style preamplifier.

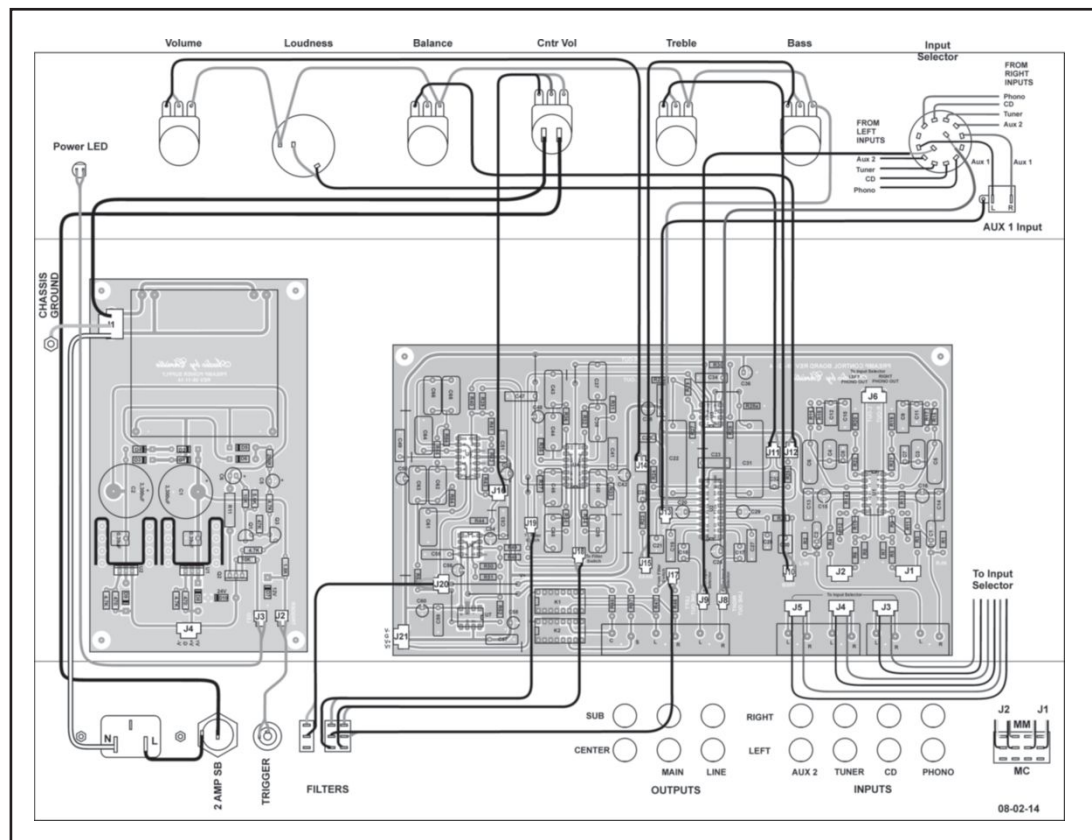


Figure 14: The chassis wiring should be completed as shown.

Optocouplers in a MOSFET Power Amplifier Design

This article presents a novel method of temperature compensation in audio power amplifiers that utilize MOSFETs as the output devices. It is of particular interest for vertical MOSFETs, since these devices must be compensated unless you operate at very low bias currents. I discuss several advantages of using optocouplers to achieve temperature compensation in MOSFET amplifier design and I present simulations using LTspice to illustrate the method.

By
Richard K. Mains
(United States)

Below certain drain current levels, power MOSFETs exhibit a negative gate-source voltage temperature coefficient (TCV_{GS}). This means that if the device heats up, V_{GS} must be decreased in magnitude to maintain the same drain current. If V_{GS} is not decreased and there is insufficient heatsinking to counteract the temperature increase, thermal runaway can occur. Also, if MOSFETs are placed in parallel, one device can heat up and hog current from the other devices.

Thermal Stability Issues

Two general categories of power MOSFETs are lateral and vertical devices. They differ in their internal structures and in their temperature coefficients. The TCV_{GS} for lateral power MOSFETs typically crosses over from negative to positive in the 100 mA to 150 mA bias current range, so biasing them in this range can solve the thermal stability problem. However, the TCV_{GS} for vertical power MOSFETs does not become positive until you reach the 8-to-10-A bias current range, which is

impractically high. You might wonder why in that case anyone would want to use vertical MOSFETs for an audio amplifier. One reason is that vertical MOSFETs are less expensive than lateral devices. In addition, vertical devices can be driven to higher drain current levels than lateral MOSFETs.

The usual approach to temperature compensation is to mount one or more transistors in the bias spreader circuit on the same heatsink as the power MOSFETs. An excellent discussion of this method, of MOSFET amplifiers, and of audio amplifiers in general can be found in Bob Cordell's book, *Designing Audio Power Amplifiers* (McGraw-Hill, 2011).

While this method works, it does have some disadvantages. First, it may be inconvenient to mount transistors from the driver circuit on the heatsink, which may require long traces or wires to make connections, and which can lead to undesirable parasitics and noise pickup. Also, if you parallel MOSFETs, it is not possible to compensate for mismatches in I_d vs. V_{GS} among devices since the

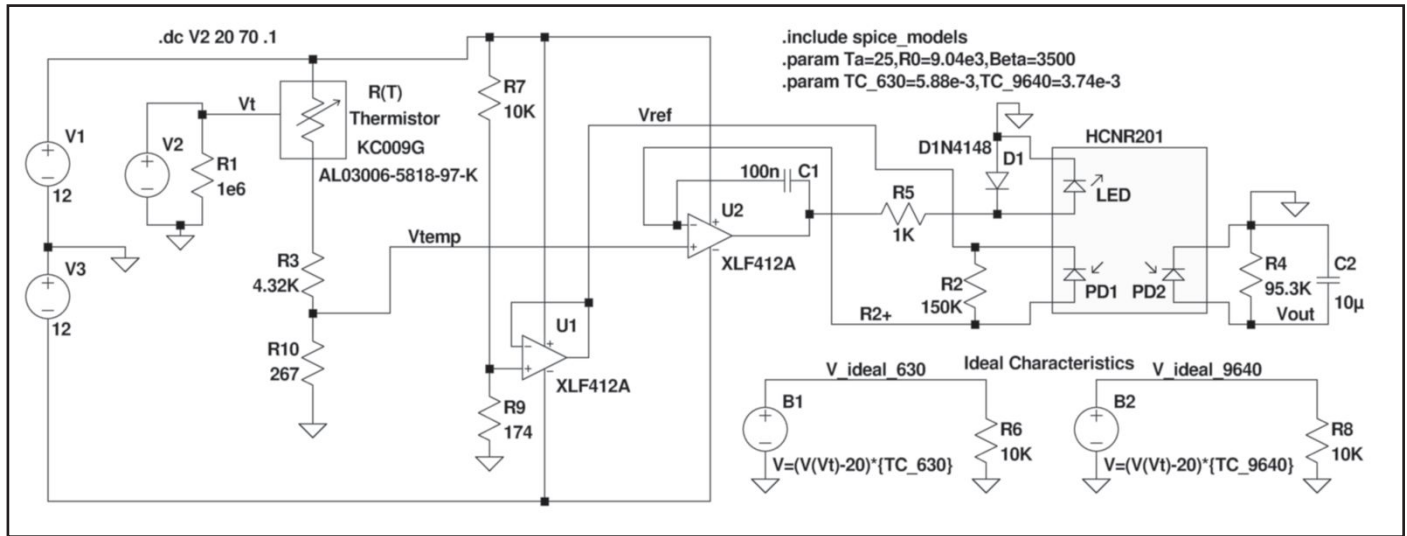


Figure 1: Thermal bias compensation is generated using a thermistor and HCNR201 optocoupler.

same compensation is applied to all devices. The device with the lower threshold V_{GS} can heat up and hog current from the other MOSFETs.

In my amplifiers, I use separate board-mounted heatsinks, and with the traditional method, it is not possible to selectively compensate for different heatsink temperatures. I will show you how to overcome all these disadvantages using optocouplers. However, there is one advantage to the traditional method of bias compensation, namely simplicity—it requires increased circuit complexity to implement the control using optocouplers.

Optocoupler Thermal Compensation Circuits

There are many possible variations of thermal control circuits using optocouplers. **Figure 1** shows a circuit that uses an HCNR201 optocoupler made by Avago Technologies. This IC consists of an input LED that illuminates two closely matched photodiodes, PD1 and PD2. The transfer gain (i.e., the ratio of short-circuit photocurrents in PD1 and PD2 for a given LED current) is specified as $1.0 \pm 5\%$.

Figure 1 is actually a screenshot from LTspice, which is an extremely useful circuit simulator available for free from Linear Technology Corp. I generated all of my circuits and results using LTspice, and you can obtain the files from the Supplementary Materials section on the audioXpress website (www.audioXpress.com). If you are unfamiliar with LTspice, there is an excellent tutorial in Cordell's book and there are help files with the program itself.

There are two main functions shown in **Figure 1** to discuss. The first is how you can generate a signal useful for bias compensation. It is not clear to me what the ideal signal should be, since TCV_{GS} varies with drain current. Another complication is that the

thermistor senses the heatsink temperature, not the MOSFET device temperature. I use a signal that starts at zero at room temperature, say 20°C , and increases linearly with temperature, with slope of TCV_{GS} for the MOSFETs being compensated. **Figure 1** simulates the situation where temperature is monitored using a negative temperature coefficient (NTC) thermistor. To a good approximation, the thermistor resistance is given by:

$$R(T) = R0 \times \exp\left(\beta \left(\frac{1}{(T+273)} - \frac{1}{(Ta+273)} \right)\right)$$

where T is the temperature in degrees Celsius and $Ta = 25^\circ\text{C}$. $R(T) = R0$ at temperature Ta and decreases with temperature. The thermistor has a nominal $R0$ of $10\text{ k}\Omega \pm 10\%$. However, due to self-heating in the circuit shown in **Figure 1**, I use a value of $R0 = 9.04\text{ k}\Omega$. From my measurements, I also use $\beta = 3,500$, which differs somewhat from the value given in the datasheet.

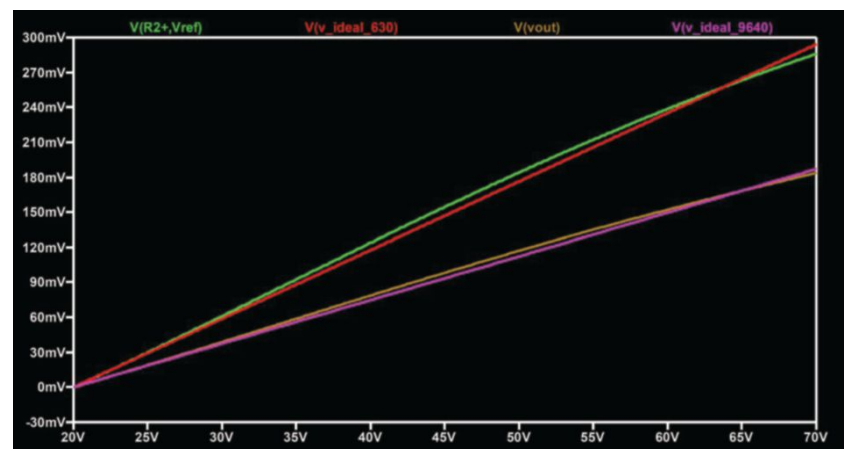


Figure 2: This is an LTspice simulation of the circuit shown in Figure 1.

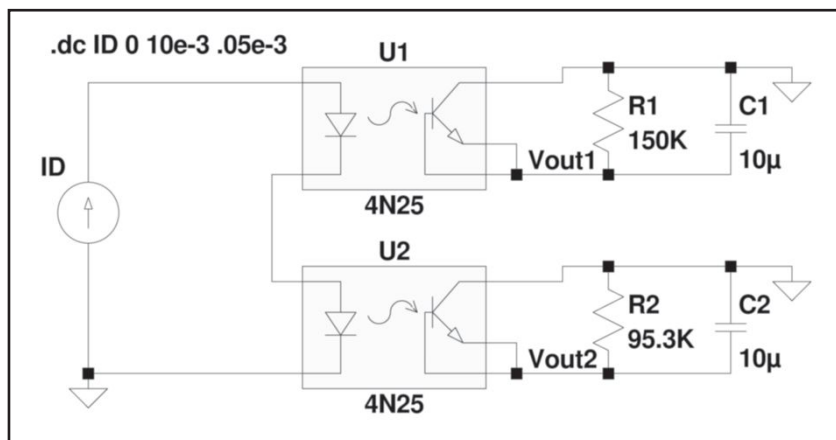


Figure 3: I conducted an LTspice test of 4N25 optocouplers.

Using the data shown in **Figure 1**, I performed a swept DC analysis. The voltage of V2 is stepped from 20 V to 70 V in steps of 0.1 V, and at each voltage a DC solution is performed. The voltage of V2 at node V_T represents the heatsink temperature in degrees Celsius. The thermistor block is a subcircuit, which converts the voltage V_T into a resistance, according to the equation for $R(T)$.

The voltage divider consisting of resistors R3, R10, and $R(T)$ is used to generate a signal at node V_{TEMP} , which is proportional to heatsink temperature V_T . It is not exactly linear, but it can be made nearly so within the important range. This signal is fed to the positive input of op-amp U2, which will attempt to bring its negative input to the same voltage. Imagine that V_{TEMP} is larger than the negative input of U2 at node R2+. This causes U2 to generate a positive voltage at its output, which drives current into the LED of the optocoupler, and in turn produces a voltage across photodiode PD1, with positive polarity at node

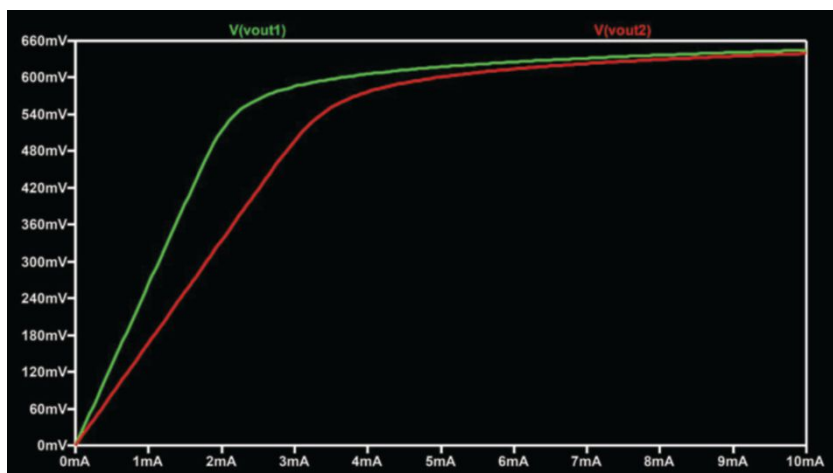


Figure 4: The simulated outputs of the 4N25s shown in Figure 3 are displayed.

R2+. This process continues until the voltage at node R2+ is equal to V_{TEMP} .

The MOSFETs I use are the IRF630 (n-channel) and IRF9640 (p-channel). I measured the TCV_{GS} for these devices at 30 mA bias and found them to be different, with $TC_{630} = 5.88e-3$ and $TC_{9640} = 3.74e-3$ V/°C. (They are actually negative, but I use positive values in the simulations.) **Figure 1** also shows the desired thermal compensations for each device, which I chose to be linear functions with slope TCV_{GS} that go to zero at 20°C. The voltage at V_{TEMP} does not go to zero when $V_T = 20$, so you need to subtract an offset from it.

The resistor divider of R7 and R9 generates this offset, which is buffered at the output of U1 at node V_{REF} . When the voltage at the negative input of U2 at node R2+ is brought to V_{TEMP} , the voltage across resistor R2 is $V_{TEMP} - V_{REF}$. (Resistors R3, R10, R7, and R9 must be carefully chosen to achieve the desired slope and offset for the thermal compensation. Unfortunately, the equations are too complicated to show here, but I will post something in the audioXpress website's Supplementary Material section).

The next important function shown in **Figure 1** is to convert the signal $V_{TEMP} - V_{REF}$ into a voltage that is directly useful for thermal compensation. As mentioned above, this voltage appears across resistor R2 due to the LED current generated by op-amp U2 and the resulting photocurrent in diode PD1. We can't use this voltage directly because it is not isolated. However, PD1 and PD2 are closely matched. So, if we load PD2 with the same resistance as PD1, we can expect that nearly the same voltage will appear across PD2, or across resistor R4, and this voltage is electrically isolated. (I have grounded one side of R4 because SPICE does not like floating circuits; in reality, the output is isolated). Notice that $R2 \neq R4$ in **Figure 1** to show how you can tailor the compensation voltage by changing the ratio of these resistors.

Figure 2 shows a plot of the LTspice simulation of **Figure 1**. The horizontal axis is V2, which represents the heatsink temperature in degrees Celsius. The top two curves relate to compensation for the IRF630, for which the resistor dividers shown in **Figure 1** were designed. The red curve at node v_{ideal_630} is what we want to achieve, and $V(R2+, V_{REF})$ shows the voltage that is actually developed across resistor R2. R2 is purposely different from R4 in **Figure 1**, and the ratio $R4/R2$ is 0.635, which is also the ratio of TC_{9640}/TC_{630} . At this level of loading, PD1 and PD2 act as current generators, so this ratio of $R4/R2$ is used to produce V_{OUT} appropriate for compensation of the IRF9640 device; the relative slopes of the voltages across PD1 and PD2 can be adjusted by varying the

R4/R2 ratio. If instead I want to compensate the IRF630 MOSFET, I set R4 = R2. The bottom two curves in **Figure 2** show that the output voltage across PD2 is close to the desired compensation curve for the IRF9640 device.

To implement thermal compensation, the voltage across PD2 is placed directly in series with the MOSFET gate. This has many advantages. We can eliminate long leads by placing the optocoupler close to the MOSFET driver circuit and the control circuit can be far away from the driver, since it is isolated from it. We can easily tailor the exact form of the compensation by selecting appropriate resistors. And in principle we can have separate compensation for each MOSFET if we want to, by having control circuits for each device; this allows the option of using separate heatsinks.

Note that I placed a 10 μ capacitor, C2, across the output. This is not necessary to carry out the DC simulations, but it serves an important purpose. We want the compensation to be DC or slowly varying with time. The audio signals should appear at the MOSFET gate, but not in V_{OUT} . Since C2 is much larger than the MOSFET capacitance and is in series with it, essentially all of the audio signals appear at the gate;

C2 shunts the audio signal around the optocoupler. Also, the R4-C2 time constant sets the response time of the optocoupler output.

There are real-world considerations related to **Figure 1** that I should mention. Due to the tolerances of the HCNR201 optocoupler transfer gain and the thermistor resistance, resistors R3, R10, and the R4/R2 ratio need to be calibrated to achieve the best accuracy. Also, you should use a good-quality capacitor for C2 to avoid parasitics at high frequencies that might cause oscillation. And the thermistors should be protected from air currents, which can cause erratic behavior of the circuit and make a steady-state solution difficult to obtain.

The circuit shown in **Figure 1** is useful when you want to apply a well-defined thermal compensation to MOSFETs. However, sometimes you want to adjust the gate voltage to achieve a goal where the exact form of the required voltage is not known. One example is if you want to compensate for differences in gate threshold voltages among MOSFETs placed in parallel. Rather than trying to measure the offsets and apply specific corrections, it is simpler to use feedback circuits to equalize the MOSFET temperatures. You don't need the relatively expensive HCNR201 chip for

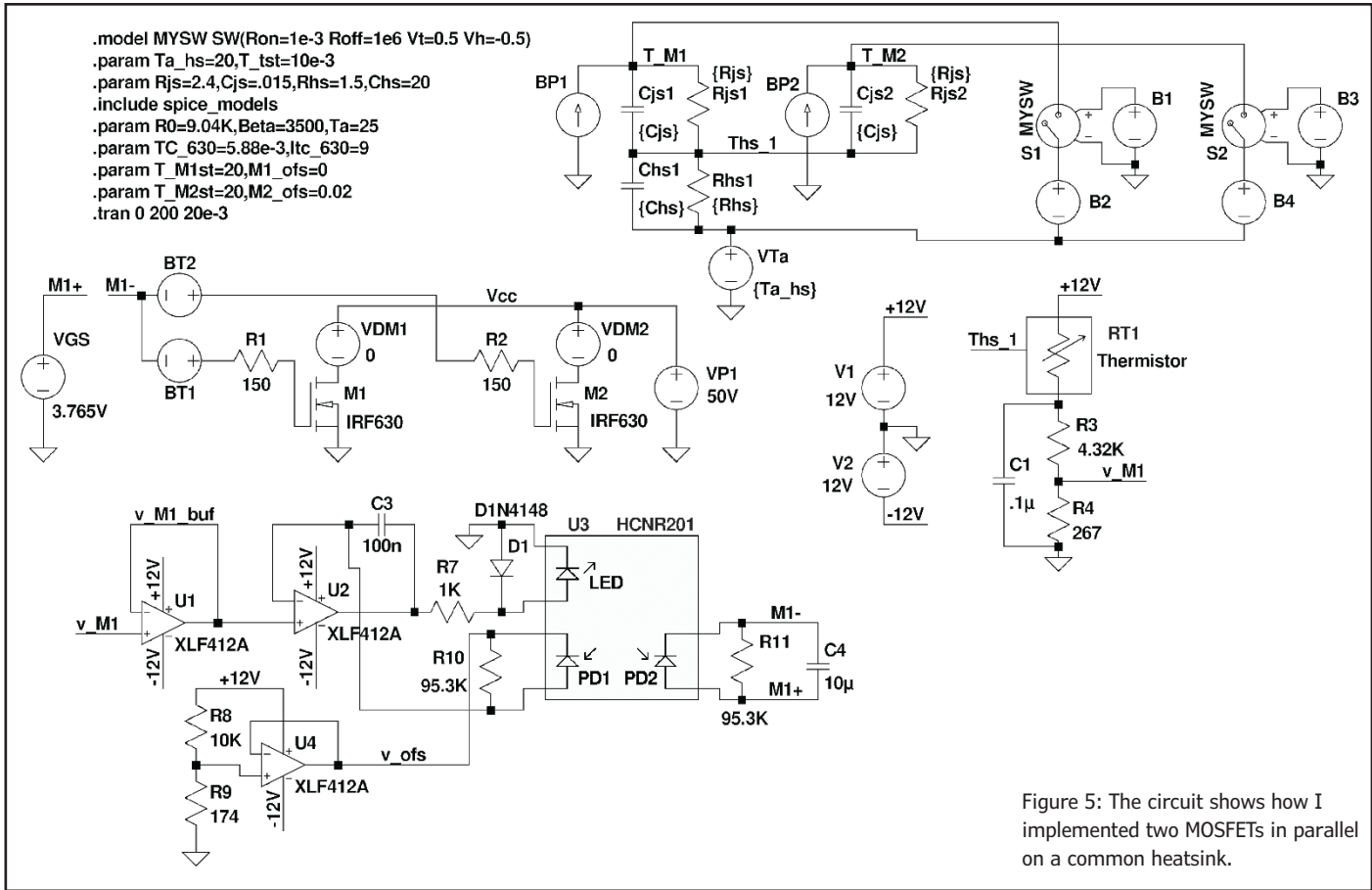


Figure 5: The circuit shows how I implemented two MOSFETs in parallel on a common heatsink.

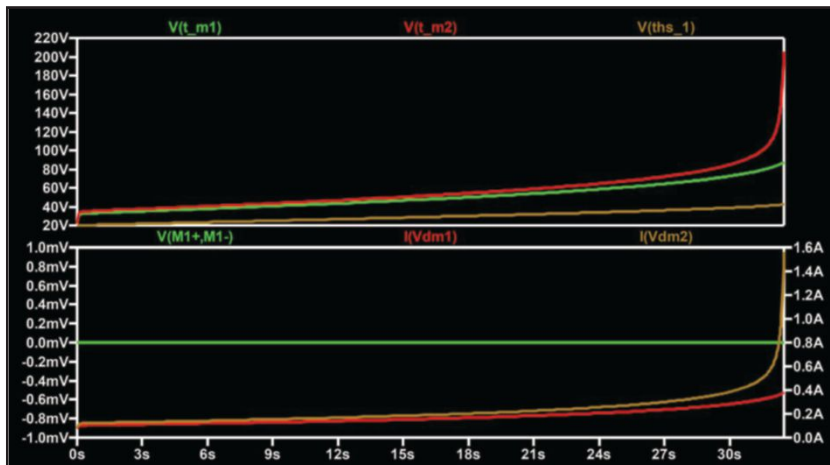


Figure 6: This is a simulation of Figure 5 with thermal compensation disabled.

that. Instead, **Figure 3** illustrates the less expensive 4N25 optocoupler that can be used.

In **Figure 3**, the base-emitter connections of the 4N25s are shorted, configuring the output photo-transistors as diodes. I included two 4N25s to demonstrate the effect of different resistors across the outputs. **Figure 4** shows the results of the swept DC simulation—the horizontal axis is ID and the current through the LEDs of both optocouplers. Since R1 and R2 are different, the output slopes are different in the linear region. Note, however, that beyond a certain current the output of both 4N25s saturates. This behavior also occurs for the HCNR201, although only the linear region was shown in **Figure 2**. If the 4N25 is used in a feedback circuit, it is not as important to stay within its linear output region.

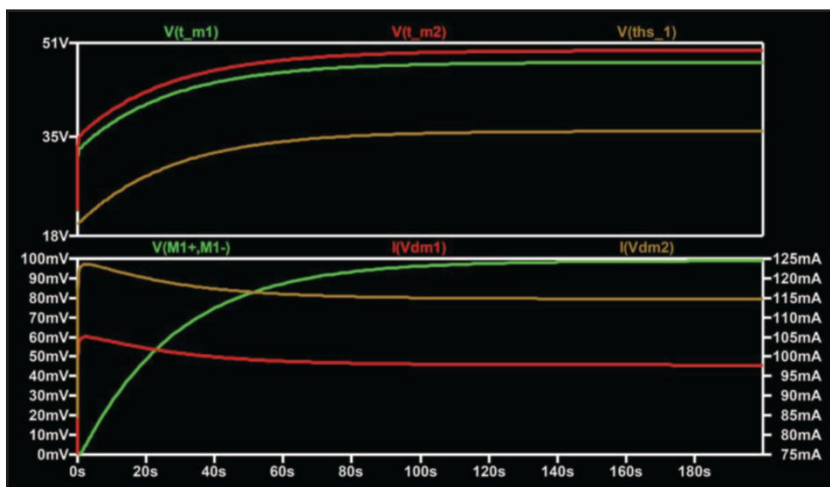


Figure 7: Here is a simulation of the circuit shown in Figure 5 with thermal compensation enabled.

Electrical+Thermal Circuit Simulation

In this section, I show how the ideas presented in **Figures 1–4** can be used to implement complete thermal compensation systems. **Figure 5** shows one implementation, with two IRF630 MOSFETs labeled M1 and M2 that are electrically connected in parallel and mounted on a common heatsink. This could represent the positive half of an amplifier output stage. For the MOSFETs, I made hybrid models as described in Cordell's book, which combine features of the VDMOS and EKV models. Feel free to substitute your own transistor models, if desired.

Voltage generator VP1 applies 50 V between drain and source of the MOSFETs. Voltage generator V_{GS} biases the transistors at a DC operating point. Here I am trying for about 100 mA per device. Voltage sources BT1 and BT2 implement the negative TCV_{GS} of the devices. As the MOSFET temperatures increase, they become more positive, which makes the devices conduct more. The TCV_{GS} is equal to the TC_{630} value in the .param statement at low drain currents, and it decreases linearly with drain current until becoming zero at drain current I_{tc_630} , which I obtained from the datasheet. You can right-click on BT1 and BT2 to see how this is done.

The HCNR201 circuit shown in **Figure 5** is essentially the same as the one shown in **Figure 1**. It generates the thermal compensation signal that is applied to the gates of the MOSFETs. The output of the HCNR201 appears at nodes M1+ and M1-. Notice that there is a break in the gate circuit with the same node labels, which is how I apply the compensation. When $V(M1+, M1-)$ becomes positive it reduces the voltage applied to the gates, which counteracts the effects of BT1 and BT2. The input to the HCNR201 circuit is the voltage at node v_M1 , which is derived from thermistor RT1 (mounted on the heatsink) in series with resistors R3 and R4. Notice that $R10 = R11$ in the circuit. The voltage developed across PD1 is appropriate to compensate the IRF630 device, and since this is the device we are using we want the same voltage to appear across PD2.

The thermal circuit is shown at the top of **Figure 5**. If you right-click on current generators BP1 and BP2, you find that their currents are just the powers dissipated in M1 and M2. In the thermal circuit current represents power in watts, and voltage is temperature in degrees Celsius. $V(T_M1)$ and $V(T_M2)$ represent the internal temperatures of the MOSFETs. R_{JS} and C_{JS} are the junction-to-heatsink thermal resistance in $^{\circ}C/watt$ and thermal capacity in $Joules/^{\circ}C$. I got R_{JS} from the datasheet, and I inferred an approximate value for C_{JS} from curves in the datasheet. The time constant for this junction-to-heatsink thermal impedance is 36 ms.

For the heatsink, I assumed a constant thermal resistance of $R_{HS} = 1.5^\circ\text{C/Watt}$ and thermal capacity of $C_{HS} = 20 \text{ Joules}/^\circ\text{C}$, giving a thermal time constant of 30 seconds. (Heatsink thermal resistance varies with power dissipation. If you have data for a heatsink you want to use, you can replace resistor R_{HS} with a subcircuit that implements resistance in tabular form). $V(\text{Ths}_1)$ is the temperature at the heatsink surface, and voltage generator V_{TA} applies the ambient temperature that the heatsink assumes with no power dissipation, which I chose to be $T_{a_hs} = 20^\circ\text{C}$. Switches S1 and S2 and their controlling voltages B1 and B3 enable the devices to start at known temperatures given by T_{M1st} and T_{M2st} . When the simulation time is less than $T_{tst} = 10 \text{ ms}$ these switches are closed, but for times greater than T_{tst} they are open and allow the temperatures to develop.

In addition to implementing the TCV_{GS} effect, which is a function of device temperature and drain current, BT1 and BT2 also apply constant offsets in the gate circuits, given by parameters M1_ofs and M2_ofs. Since $M2_ofs = 0.02$, device M2 will always have an extra 20 mV applied between its gate and source, which will tend to make it conduct more than M1. It is important to include this effect, since although we try to match transistors that will be placed in parallel, their characteristics will not be identical.

Before we simulate the circuit shown in **Figure 5**, let's see what happens if we disable the thermal compensation generated by the HCNR201 circuit. We can accomplish this by setting $R11 = 1 \mu\Omega$, which shorts the output of PD2 so that $V(M1+, M1-) \approx 0$. **Figure 6** shows the results after making this change. The top curves show the device and the heatsink temperatures. After about 30 seconds the M2 device temperature rapidly increases to a point where it would be destroyed, even before the heatsink is able to respond. The lower curves show that the M2 drain current, $I(\text{VDM2})$, takes off as well. This is an example of thermal runaway, which can happen very quickly. Remember that M2 has an offset relative to M1, which is why it takes off first.

In contrast, **Figure 7** shows the simulation of the circuits shown in **Figure 5** when the value of $R11$ is restored to $95.3 \text{ k}\Omega$. In this case, thermal runaway does not occur. After 200 seconds, the device temperatures of M1 and M2 are 47.6°C and 49.7°C , and the heatsink temperature is 35.9°C . M2, of course, gets hotter than M1 due to its offset of 20 mV. The $V(M1+, M1-)$ curve shows that the thermal compensation generated by the HCNR201 settles down at 99 mV. We can check this value by noting that the heatsink rise from ambient is $35.9 - 20 = 15.9^\circ\text{C}$, so that the thermal compensation should be $15.9 \times TC_{630} = 93.5 \text{ mV}$, close to the value actually obtained.

In the lower curves of **Figure 7**, you can see that the device drain currents initially overshoot their final steady-state values before the thermal compensation has a chance to develop. Also, the final drain current reached by M2 is 114.7 mA, significantly higher than for M1, which reaches 97.7 mA. This is again because of the 20 mV offset of M2. Even though M2 operates at higher current and temperature than M1, thermal stability is maintained.

I mentioned that I use separate board-mounted heatsinks in my amplifiers. As an example of this configuration, **Figure 8** shows a circuit with two IRF630 MOSFETs electrically in parallel but mounted on separate heatsinks. First note that the thermal circuit at the top of **Figure 8** now has two sets of R_{HS}/C_{HS} elements to model the separate heatsinks. To maintain a similar thermal environment, I

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<http://audioxpress.com/page/audioXpress-Supplementary-Material.html>.

Resource

B. Cordell, *Designing Audio Power Amplifiers*, McGraw-Hill, 2011.

Sources

HCNR201 Optocoupler

Avago Technologies | www.avagotech.com

LTspice

Linear Technology Corp. | www.linear.com

have doubled the thermal impedance values relative to the circuit shown in **Figure 5**. Also, note that there are now breaks in both gate circuits where thermal compensation is applied.

The HCNR201 circuit, which generates the thermal compensation for M1, is the same as in **Figure 5**. The circuit with U6, a 4N25 optocoupler, generates the compensation for M2 but operates entirely differently. Rather than setting the M2 compensation based on its heatsink temperature, this circuit attempts to adjust the M2 heatsink temperature, T_{hs_2} , to be the same as T_{hs_1} . By doing this we guarantee that M1 and M2 equally share drain current, and we can

also compensate for offsets in their characteristics.

In this circuit, a buffered version of the signal representing M1's heatsink temperature, v_{M1_buf} , is used as the standard at the negative input of op-amp U5. The signal for M2's heatsink temperature, v_{M2} , is fed to the positive input, and the op-amp will attempt to make the signals the same. Imagine that M2's heatsink is hotter, so that $v_{M2} > v_{M1_buf}$. In that case, U5 will generate a positive output voltage which drives current into the LED of optocoupler U6. This LED current produces a positive voltage at the output base-collector junction of U6, or $V(M2+, M2-)$, which is applied in the gate circuit of M2 and tends to reduce its conduction and bring down its temperature. Resistor R13 limits the maximum current that U5 can produce to about 4.7 mA, since there is about 1 V across the LED when it is forward biased. C5 tends to stabilize the LED voltage. v_{M1_buf} and v_{M2} will not be exactly the same, due to R5; they will differ by the value of the LED current multiplied by R5. Selecting R5 is a tradeoff—a smaller value will make v_{M2} closer to v_{M1_buf} , but it also increases the gain of the circuit and can make it more difficult to reach steady state. I have found that 5 Ω is a good compromise between accuracy and stability. Finally, note that both thermal compensation circuits can only generate positive voltages at their outputs—they can only reduce conduction of the transistors.

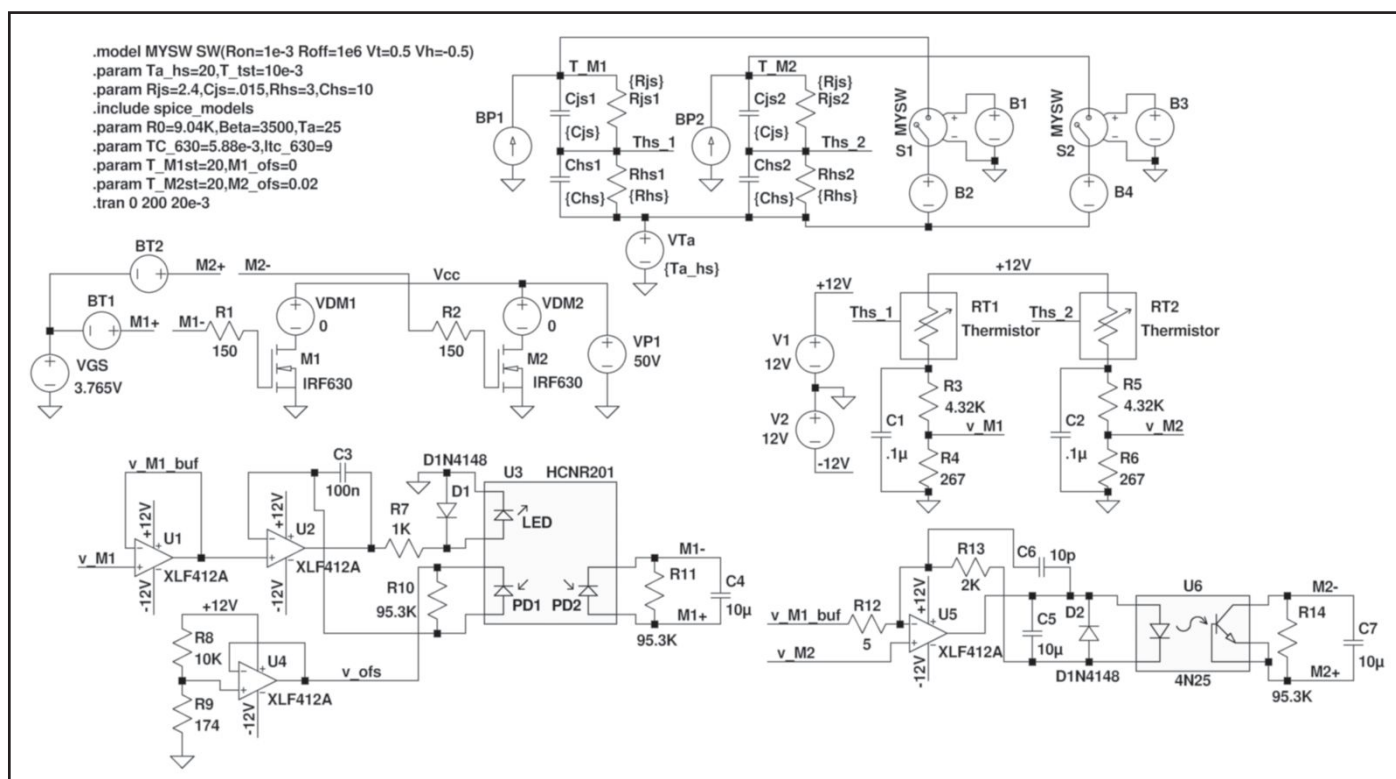


Figure 8: I built a circuit with two MOSFETs electrically in parallel but on separate heatsinks.

There is one potential problem with the 4N25 circuit shown in **Figure 8** that was not obvious to me at first. Before thermal feedback is established (e.g., during the powerup transient or if you are testing the thermal control circuit without powering the MOSFETs), the 4N25 circuit may decide that M2 is hotter than M1. This may occur due to the tolerances in thermistor resistances RT1 and RT2, or to the divider resistors R3–R6 that are selected. In that case, $V(M2+, M2-)$ will reach its maximum value and attempt to cut off transistor M2. To avoid this, divider resistors R3–R6 can be carefully chosen so that under ambient temperature conditions v_{M2} will be slightly less than v_{M1_buf} .

Also, I want to mention one issue I discovered after implementing the method used in **Figure 8**, which is not related to the thermal compensation method itself. If you decide to use separate heatsinks that are closely spaced, you should be aware that there will be interactions between the heatsinks that will alter the thermal resistance from that given in the datasheet, and probably in the wrong direction—it will likely increase. In my case, I saw a significant increase in thermal resistance over what I had expected from the datasheet.

As before, I first show a simulation of the circuit used in **Figure 8** with thermal compensation disabled, which I achieve by setting $R11 = R14 = 1\ \mu\Omega$. **Figure 9** again shows why we need the compensation—thermal runaway occurs quickly in device M2 to destructive levels. In contrast, **Figure 10** shows a simulation with compensation enabled by setting $R11 = R14 = 95.3\ \text{k}\Omega$; nearly equal device temperatures are achieved, 46.5°C for M1 and 47.4°C for M2. Also the drain currents are nearly the same, $98.2\ \text{mA}$ for M1 and $101.6\ \text{mA}$ for M2. In the early part of the simulation, M2's drain current overshoots due to its $20\ \text{mV}$ offset. Even though we compensate for this offset in steady state, it is still a good idea to match devices as closely as possible to minimize this transient overshoot. The heatsink temperature for M1 is 34.7°C , which calls for a compensation of $86.4\ \text{mV}$; the actual value achieved is $V(M1+, M1-) = 91.3\ \text{mV}$. Note from the curves that $V(M2+, M2-) = 110\ \text{mV}$. This is almost exactly $20\ \text{mV}$ above the $V(M1+, M1-)$ value and compensates for the $20\ \text{mV}$ offset of M2 relative to M1.

Conclusion

I hope the ideas in this article provide some useful options for the thermal design of MOSFET amplifiers. I have implemented a compensation scheme similar to that of the circuit shown in **Figure 8** in a version of my amplifier, except that I have three MOSFETs in parallel per side instead of two. So far, the scheme

has worked as the simulations indicate—thermal stability is achieved and the heatsink temperatures differ by at most 3°C or so. However, there is one difference. Rather than reaching a steady-state solution as in the simulations, voltages in the real circuit hover about the operating point, varying slowly above and below it with a period of a few seconds. 

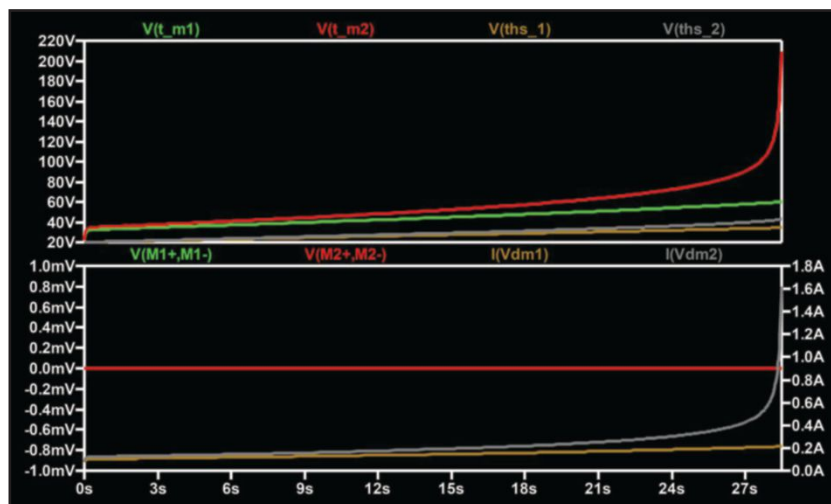


Figure 9: This is a simulation of the circuit shown in Figure 8 with thermal compensation disabled.

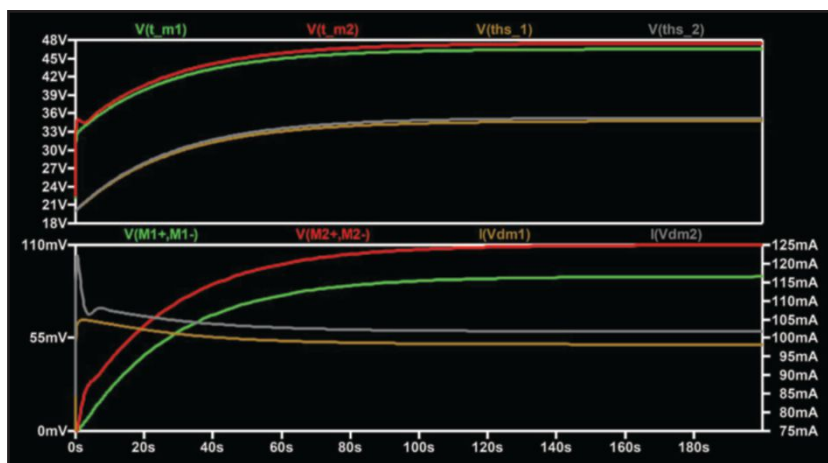


Figure 10: Here is a simulation of the circuit shown in Figure 8 with thermal compensation enabled.

About the Author

Richard K. Mains obtained a Bachelor of Science degree and a Master of Science degree in Electrical Engineering from the Ohio State University in 1972 and 1974, and a PhD in Electrical Engineering from the University of Michigan in 1979. He worked for several years in the area of high-frequency semiconductor device modeling at the University of Michigan. From the early 1990s until 2013, he worked in the area of power electronics circuit design and microprocessor programming at McCleer Power, Inc., located in Jackson, MI. His audio interests include building speakers, developing measurement systems for loudspeakers, and designing power amplifiers and other electronics for music reproduction.

Tremolo Circuits in Guitar Amplifiers

Learn the difference between tremolo and vibrato, and how these effects are created in hollow-state guitar amplifiers.

By
Richard Honeycutt
(United States)

Most of the higher-model classic tube guitar amplifiers were described as having two effects: “vibrato” and reverb. In fact, what they generally had was tremolo and reverb. Vibrato means a rhythmic change in pitch, whereas tremolo means a rhythmic change in amplitude or volume.

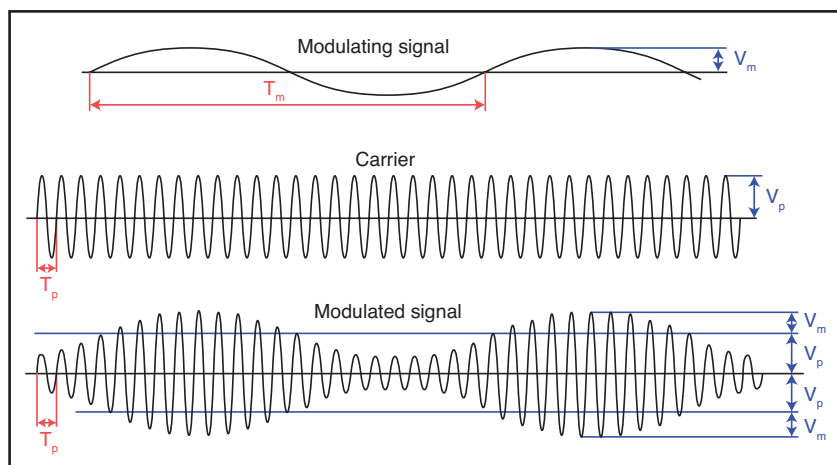


Figure 1: Tremolo involves amplitude modulation. (Image courtesy of Douglas Anderson Electronic Music 1 and 2)

In the 1950s and 1960s, artists such as Duane Eddy (“Rebel Rouser”) and the Everly Brothers (“Let It Be Me”) used tremolo on their guitars as an important part of their trademark sound. But often the controls on the amplifier were labeled “Vibrato.”

True vibrato on a guitar usually requires the use of a Bigsby vibrato tailpiece or what Fender players called a “whammy bar.” Vocal vibrato (think operatic soprano) is a combination of pitch-change and amplitude-change—a combination of pure vibrato and tremolo. Hammond organists had true “mechanical vibrato,” produced by rhythmically changing the speed of the electromechanical tone generator. Those using Leslie speakers had a combination of vibrato and tremolo: The rotating horn produced pitch changes via the Doppler Effect, and the sound level varied rhythmically due to the horns repeatedly pointing toward, then away from, the listeners.

Beginning in the 1970s, some guitarists have used Leslie speakers to provide an effect very similar to that provided by flangers. The First Edition’s “Just Dropped In” (recorded in 1968) was one of the best-known songs to utilize Leslie vibrato/tremolo as a vocal effect. Unlike a choir, in which individual singers’ vibrato

varies independently, using a Leslie on a whole vocal track modulated all the singers together, resulting in a very unique and striking effect whose desirability is a matter of taste.

Tremolo Vs. Vibrato

Another way to explain the difference between tremolo and vibrato is to describe tremolo as amplitude modulation (see **Figure 1**), and vibrato as frequency modulation (see **Figure 2**). Tremolo circuits used in guitar amplifiers usually consist of a variable-frequency oscillator whose frequency of operation ("tremolo rate") is well under 10 Hz, either driving an optoisolator or varying the control-grid bias of the output tubes.

Optoisolators used in classic amplifiers usually comprised a neon lamp illuminating a cadmium sulfide

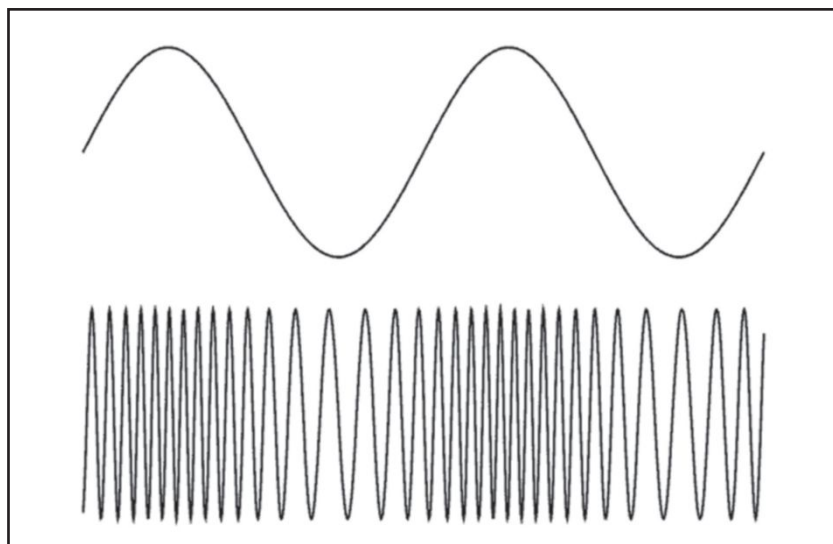


Figure 2: Vibrato involves frequency modulation. (Image courtesy of Angel Fire)

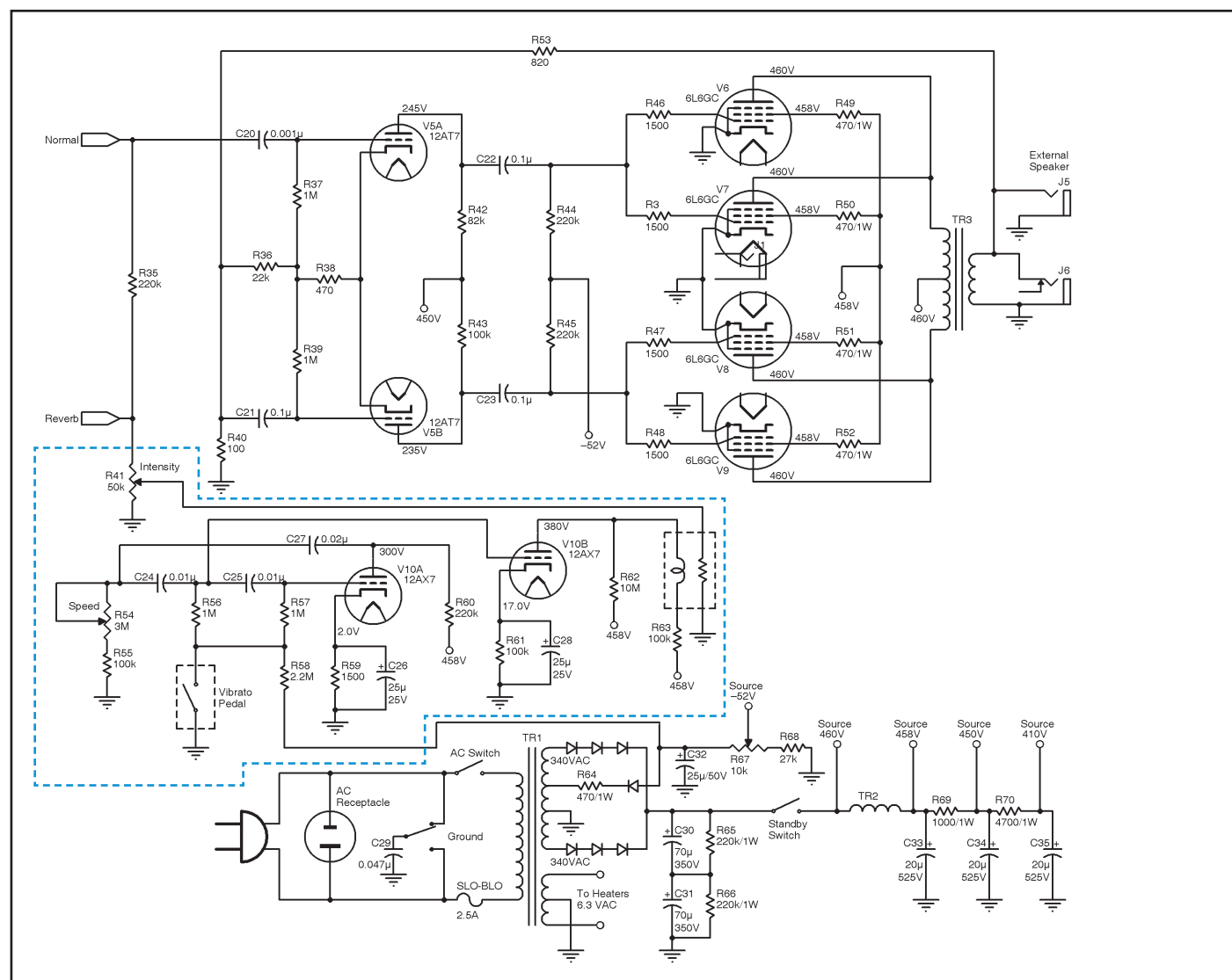


Figure 3: This partial schematic of a Fender Bandmaster amplifier has the tremolo circuit outlined by a dashed blue line.

or cadmium selenide light-dependent resistor (LDR). The neon lamp was placed in the plate circuit of a triode whose grid was driven by the oscillator's output signal. The resistance of the LDR varies more-or-less inversely with the light level, and the neon lamp's brightness is proportional to the current in the lamp. However, since the internal resistance of a neon lamp is extremely low after the voltage across it has exceeded its "striking voltage" (approximately 90 V), control of current depends upon the circuit in series with the lamp. Maximum current for a small neon lamp used in optoisolators is typically about 1.5 to 2 mA.

There are several types of hollow-state oscillators, but the one almost universally used in tremolo circuits is called a "phase-shift oscillator." **Figure 3** shows a partial schematic of a Fender Bandmaster amplifier, with the tremolo circuit outlined in blue. The left-hand triode is the oscillator. The circuit includes three RC networks: the 0.02 μF capacitor, the SPEED control, and the 100 k Ω resistor; and each of the 0.01 μF capacitor-1 M Ω resistor combinations.

At some frequency, the total phase shift provided by these three networks will add up to 180°, and there will be positive feedback, causing oscillation. The SPEED control affects the frequency at which this occurs, within limitations provided by the other components in the phase-shift network. (A common complaint about the Fender tremolo circuit is that guitarists are not able to slow the tremolo as much as they would like, and replacing the 0.01 μF capacitors with 0.02 μF units is a solution.)

Most Fender amplifiers using the optoisolator-style tremolo have pretty much the exact same oscillator as the Bandmaster, and the rest of the tremolo circuit is nearly the same. A good reference or helping you to design a phase-shift oscillator can be found on the Aiken Amplification website (see Resources).

In the Bandmaster circuit, the right-hand triode drives the optoisolator, which is connected so as to vary the amplitude of the "vibrato-channel" signal fed to the 12AT7 phase inverter. Once the triode's plate current is low enough that 90 V appears across the neon lamp, the lamp will begin to glow and then the voltage across the lamp will drop to about 60 V.

At this point, the tremolo oscillator wave applied to the grid modulates the plate resistance of the tube and that resistance, plus the 100 k Ω resistor in series with the neon lamp will control the brightness of the neon lamp. That undulating brightness, in turn, controls the resistance of the optoisolator's LDRs in an inverse fashion: The brighter the neon lamp at any instant, the lower the resistance of the LDR, and the more the signal is attenuated. Thus, the guitar signal being amplified is amplitude-modulated by the tremolo oscillator.

As an aside, in trying to understand the operation of a neon-LDR optoisolator, it helps to remember that a triode is a voltage-controlled current source. The grid-to-cathode voltage controls the plate current. If a neon lamp is connected between the B+ supply and the plate, with or without a resistor in series, the current in the neon lamp will be controlled by the tube, even though the voltage across the neon lamp remains constant at about 60 V once conduction has begun. Thus, it is not helpful to think in terms of the voltage gain of a triode used as a neon lamp driver, but rather in terms of the control of plate current.

Tremolo Circuit's Sound Character

The sound character of any particular tremolo circuit depends upon the tremolo waveform, the way in which the optoisolator tube is driven, and where in the amplifier circuit the amplitude modulation is applied.

Figure 4 and **Figure 5** show the waveforms produced

Figure 4: This is the tremolo waveform produced by a Fender Bandmaster amplifier.

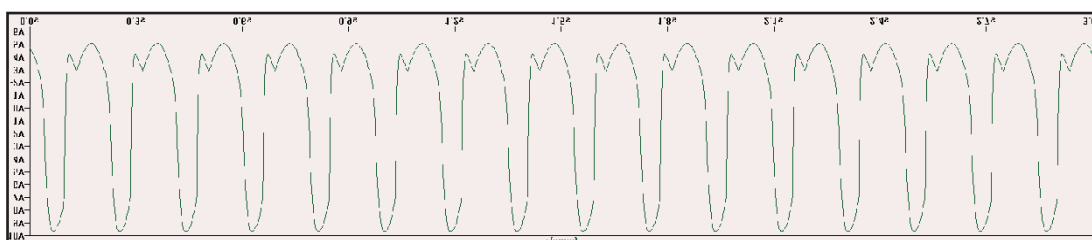
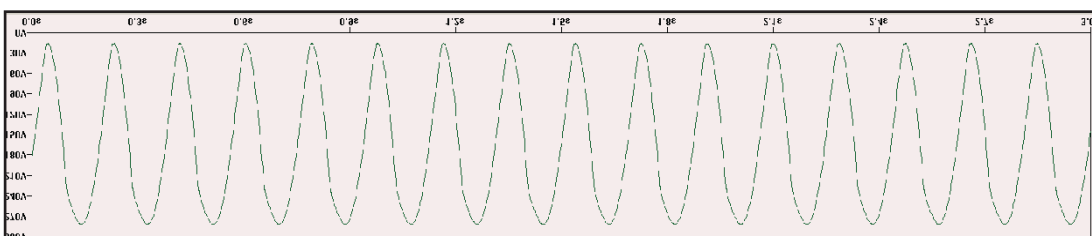


Figure 5: As you can see, the tremolo waveform of a Silvertone 1484 amplifier is more like a sine wave than is that of the Bandmaster.





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by the tremolo oscillators of a Fender Bandmaster amplifier and a Silvertone model 1484 amplifier, respectively. The two amplifiers have very different tremolo sounds, and the difference in waveforms is part of the reason. Another part of the reason is the way the tremolo intensity is controlled.

In the Silvertone, the amplitude of the oscillator waveform is varied before being applied to the tube stage driving the neon lamp, whereas in the Bandmaster, the control is on the output side of the optoisolator. In the Bandmaster, that lets the tube drive stage run full on, while in the Silvertone, the drive voltage to the neon lamp is controlled by the tremolo depth control. The signal amplitude and impedance at the point where the tremolo acts can also affect

the sound of the tremolo.

Other Ways to Provide Tremolo

Another way to provide tremolo is to feed the wave from the tremolo oscillator to the bias network of the output tubes, varying the gain of the output stage. This approach is shown in the schematic of the Fender Vibroverb shown in **Figure 6**. The Princeton Reverb and several other Fender amplifiers besides the Vibroverb use this “bias wiggling” approach.

Yet another approach to tremolo involves applying the tremolo wave to the screen grid of a voltage amplifier, or even a power tube, as in one of the small tube amplifiers using a single-ended 6AQ5 or a similar pentode as the power output tube.

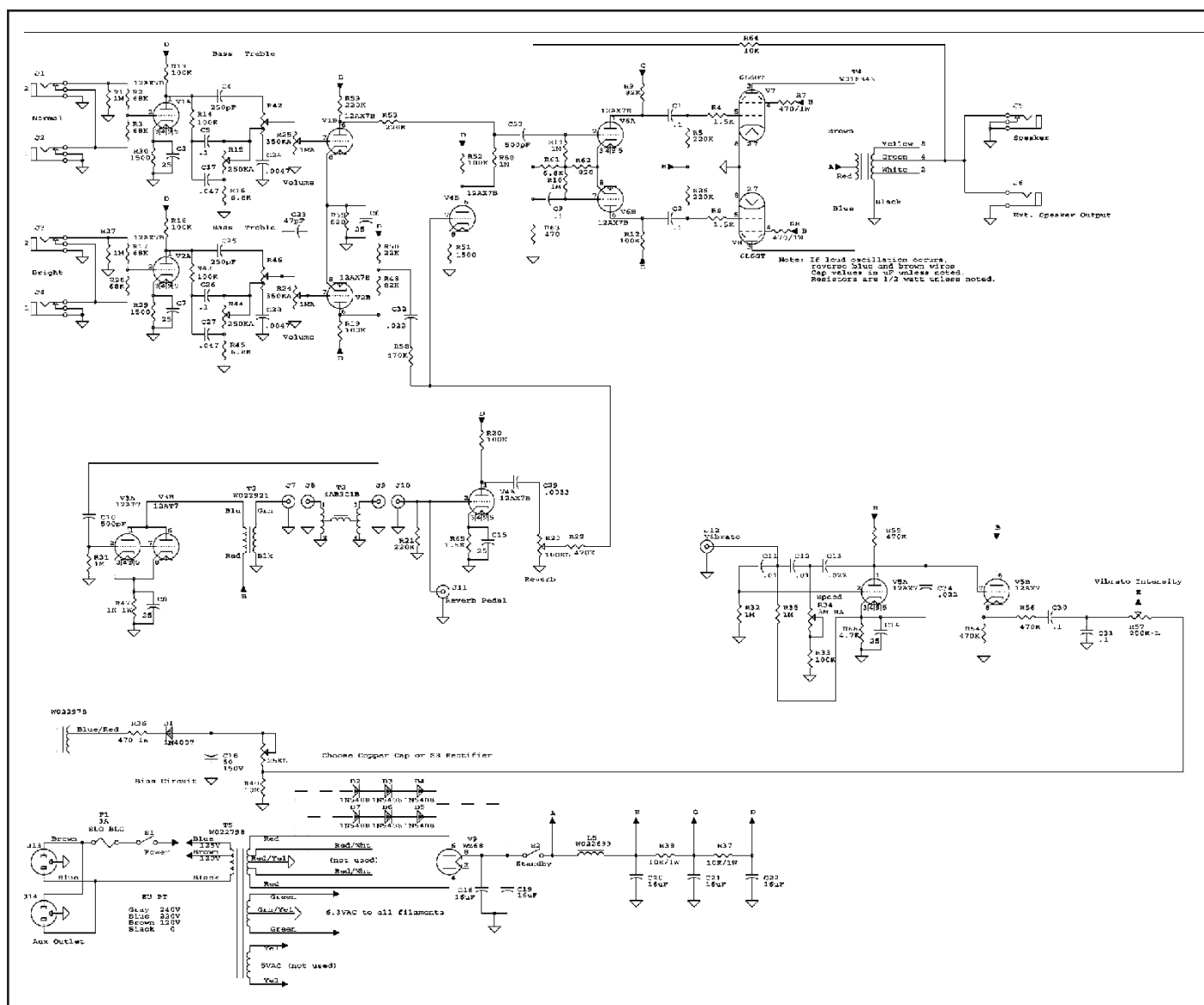
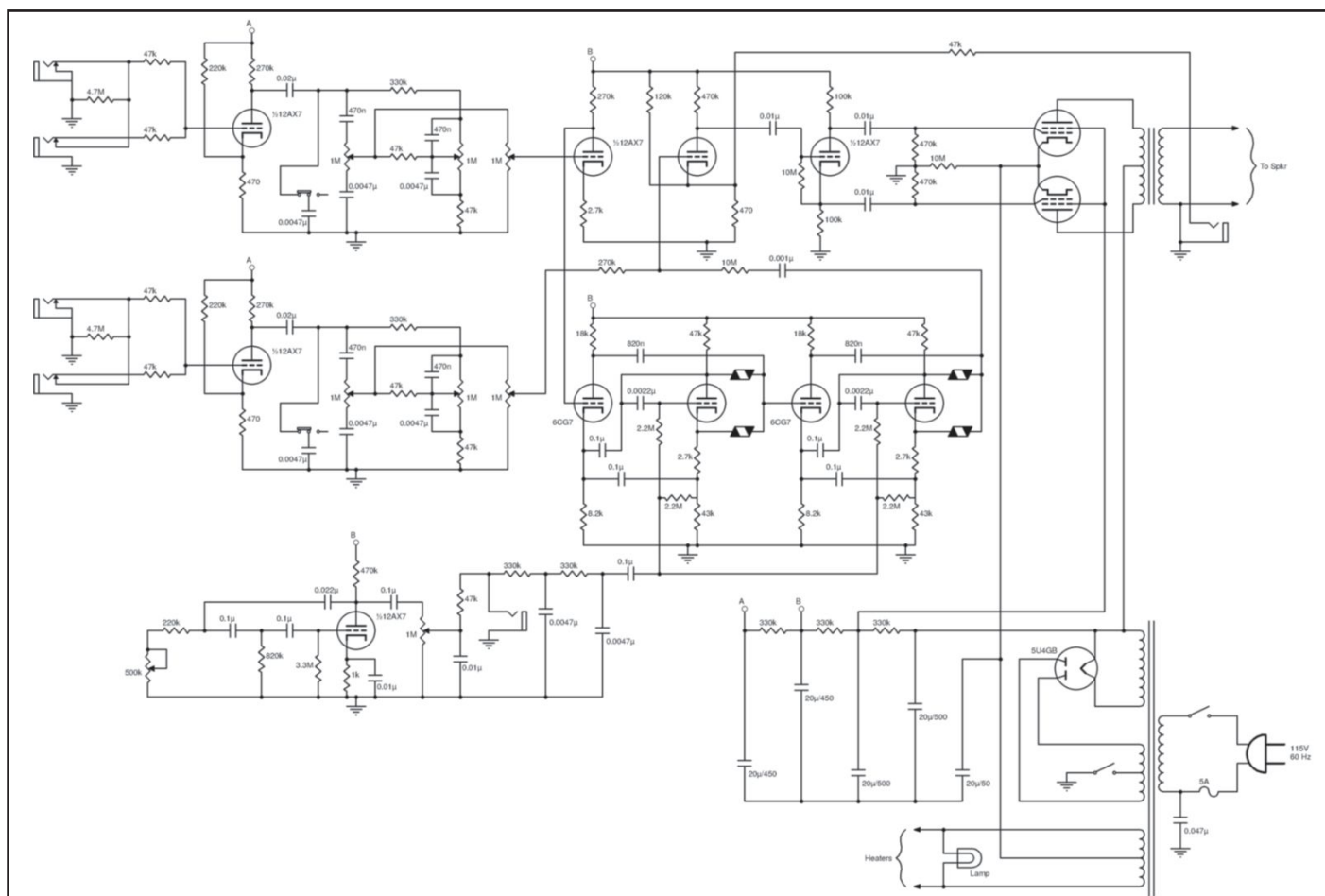


Figure 6: The Fender Vibroverb 6G16 amp applies the tremolo oscillator wave to the output-tube bias point.



Some amplifiers apply the output of the tremolo oscillator to a phase inverter, producing mirror-image outputs. The guitar signal is then split into low- and high-frequency parts that are handled by different triodes. The mirror-image tremolo wave then varies the amplitude of the lows and highs inversely to each other. This has sometimes been mistaken for true vibrato, but is more similar to a stereo chorus effect.

Figure 7 shows a schematic of the Magnatone 260. The two 6CG7 sections appearing near the right center of the schematic function as the two-stage phase modulator. Each section has a varistor connected to the plate, and another to the cathode. Since the cathode and plate signals are 180° out of phase, and have approximately equal voltages in this circuit, whichever varistor has the lowest resistance passes more signal current, making the phase of the signal at that varistor dominant. The resistance of a varistor depends upon the voltage across it, and the voltage is controlled by the oscillator waveform applied to the 6CG7 triode section. By varying the phase of the signal, we vary its frequency, since instantaneous frequency is the time derivative of instantaneous phase.

The Magnatone vibrato circuit is expensive to build because it requires more parts, and while some people really love its sound, others prefer tremolo, perhaps because they are accustomed to it. 

Resources

"Douglas Anderson Electronic Music 1 and 2," <https://daelectronicmusic.wordpress.com/concepts/analog-synthesis/amplitude-modulation>.



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

January 3–4, 2017

ALMA International Symposium & Expo (AISE) 2017

Tuscany Suites

Las Vegas, NV

www.almainternational.org

[www.almainternational.org/](http://www.almainternational.org/alma_events/)

[org/alma_events/](http://www.almainternational.org/alma_events/)

[alma_international_symposium__expo_aise_2017](http://www.almainternational.org/alma_events/)



The Association of Loudspeaker Manufacturing and Acoustics (ALMA) International Symposium & Expo (AISE) is scheduled for two days prior to CES 2017, the Consumer Technology Association show. AISE promises the best in cutting-edge education, networking, exhibits, and fun! It will also offer more business content.

As always, AISE combines a targeted audience of electro-acoustic and audio professionals with more than 40 hours of content and networking in two packed days. The 2017 edition will feature a new floor plan, assuring maximum exposure for exhibitors and ease of access to all convention functions. The theme for AISE 2017 will be "The Future of Mobile Audio," which is currently the most vibrant and growth-oriented segment of the audio industry.

The 2017 event will also feature new education content for and by students and educators, along with ALMA's world-class technical content. Also, look for an important announcement by ALMA's Standards Committee led by John Busenitz and Andrew Grunloh. AISE attendees who purchase an All Access Pass at the early bird price receive admission to the entire event, food, and all the breakout sessions for less than some shows charge for just one seminar. Educators and students can also make crucial connections for internships, career opportunities, research funding, in addition to gaining crucial information.

January 5–8, 2017

CES Show 2017

Las Vegas, NV

www.cesweb.org

In 2017, CES—the world's leading innovation event promoted by the Consumer Technology Association (CTA)—commemorates 50 years as the launch pad for breakthrough technology.

Since its start in 1967, CES has showcased an estimated 700,000 products. These include world-changing innovations such as the VCR (1970), the DVD (1996), the digital radio (2000), and Blu-ray recording standards released (2002), as well as more recent technologies—consumer 3-D printers, tablets, and 4K Ultra HD television.

CES has also driven the market growth in pioneering technologies such as autonomous cars and drones. The 50th anniversary CES will celebrate the evolution of the industry.

CES 2017 will have more than 3,800 exhibitors, covering 2.4 million net ft² and more than 175,000 attendees. A global show, CES expects more than 50,000 international professionals representing 150 countries.

CES continues as a proving ground for innovators and has expanded to feature industries not traditionally connected to consumer tech. The show touches every major industry as leaders from the entire innovation ecosystem come together to discover and experience the latest trends and network with peers. CES is where titans of tech mix with next-generation disruptors and where esteemed and sought-after global leaders take the keynote stage to tout disruptive technology and envision new trends and markets. CES remains a trade-only event and is not open to the general public.



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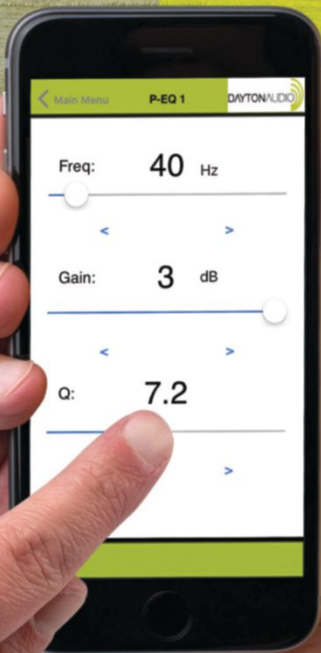
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