

audio**x**press

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Fresh From the Bench

Real Room Correction Primacoustic London 12 Kit in Review

By Ron Tipton

Fresh From the Bench

Vienna Suite Pro High-Precision Stereo & Surround Audio Plug-Ins

By Fernando Rodrigues

You Can DIY!

True Bass in a Small Space A Subwoofer Project with Dayton Audio's Ultimax 18"

By Thomas Perazella



You Can DIY!

Build a Low-Power, Class-A Power Amplifier (Part 2)

By George Ntanavaras

Questions & Answers

Interview with Nancy Moon at MOON Amplification

By Shannon Becker

Audio Electronics

Colorful Amplification for Ultra-High Impedance Sources

By Addie and Whisker

Hollow-State Electronics The Oscilloscope

By Richard Honeycutt



PARTS CONNEXION

The Authority on Hi-Fi DIY



music hall **dB** (dB) decibel Headphones



**75%
Off**

NOW

MSRP Retail
\$199.95 USD

\$49.95 USD

www.partsconnexion.com

Sales:



905-631-5777

Toll-Free - US/Canada

1-866-681-9602

905-681-9602



Inquiries:

info@partsconnexion.com

5403 Harvester Rd, Unit 1, Burlington, Ontario, Canada L7L 5J7



**HAMMOND
MANUFACTURING®**



MANUFACTURING & DESIGNING TRANSFORMERS & ENCLOSURES SINCE 1917

Chassis, Plates & Covers



Tube Output Transformers

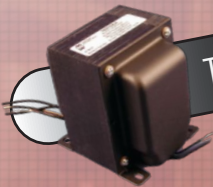
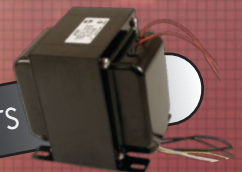
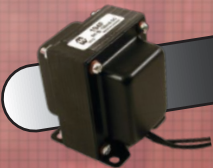


Plate & Filament Transformers



Open & Enclosed Chokes



USA

**Tel: (716) 630-7030
Fax: (716) 630-7042**

Canada

**Tel: (519) 822-2960
Fax: (519) 822-0715**

UK

**Tel: 01256 812812
Fax: 01256 332249**

Australia

**Tel: 8-8240-2244
Fax: 8-8240-2255**

www.hammondmfg.com



February 2017

ISSN 1548-0628

www.audioXpress.com

audioXpress (US ISSN 1548-0628) is published monthly, at \$50 per year for the US, at \$65 per year for Canada, and at \$75 per year Foreign/ROW, by KCK Media Corp., at 111 Founders Plaza, Suite 904, East Hartford, CT 06108, US Periodical Postage paid at East Hartford, CT, and additional offices.

Head Office:

KCK Media

111 Founders Plaza, Suite 904

East Hartford, CT 06108, US

Phone: 860-289-0800

Fax: 860-461-0450

Subscription Management:

audioXpress

P.O. Box 462256

Escondido, CA 92046, US

Phone: 800-269-6301

E-mail: audioXpress@pcspublink.com

Internet: www.audioXpress.com

Postmaster: send address changes to:

audioXpress

P.O. Box 462256

Escondido, CA 92046, US

Advertising:

Strategic Media Marketing, Inc.

2 Main Street

Gloucester, MA 01930, US

Phone: 978-281-7708

Fax: 978-281-7706

E-mail: audioXpress@smmarketing.us

Advertising rates and terms available on request.

Editorial Inquiries:

Send editorial correspondence and manuscripts to:

audioXpress, Editorial Department

111 Founders Plaza, Suite 904

East Hartford, CT 06108, US

E-mail: editor@audioXpress.com

Legal Notice:

Each design published in *audioXpress* is the intellectual property of its author and is offered to readers for their personal use only.

Any commercial use of such ideas or designs without prior written permission is an infringement of the copyright protection of the work of each author.

© KCK Media Corp. 2017
Printed in the US

Cards on the Table



Sometimes, technology adoption poses considerable dilemmas for manufacturers. In terms of product development and marketing cycle, a critical decision for a manufacturer remains when to get the message out: When the planning to adopt a new technology starts or when they're ready to deliver?

In the past, I have written about the audio industry being increasingly dependent upon key technologies and standards that it doesn't control and are defined by the IT, telecom, or consumer electronics industries. Then, the audio industry tries to "mold" things according to its own needs. The problem is that, by the time some manufacturers effectively feel they are under control... there's already a new standard in the industry. And there's little we can do about it, except plan ahead to avoid obsolescence—which inevitably will come—and be as open about it possible. That end-users complain about this state of things is perfectly normal. Harder to understand is that some manufacturers avoid that reality. I find it curious that certain companies adopt the "it's good enough for what it is" attitude in this day and age, as though consumers didn't have access to all the information.

With a few weeks left before CES and the NAMM show, while writing this editorial, I cannot help wondering what the decisions from companies will be regarding key innovations changing the technology landscape, given the benefits and competitive arguments, or the simple fact that they need to move ahead. Thinking specifically about converters and audio recording interfaces, 2017 will certainly be an interesting time. Following a relatively stable period where the industry learned to depend upon Firewire and USB 2.0, in the last few years, things started to change dramatically with USB 3.0, Thunderbolt 1, Thunderbolt 2, and the arrival of the first Audio-over-IP (AoIP) protocols for audio networking interfaces, such as AVB, Dante, Ravenna, and several proprietary solutions. To make things more confusing, many manufacturers completely ignored the changing landscape and continued to introduce USB 2.0 solutions because "that's what everyone is familiar with and it works."

In 2016, everything changed. USB 3.0 and USB 3.1 Gen1 were replaced by USB 3.1 Gen2, adopting the Type-C connector. Thunderbolt 3 was unveiled, using the same new USB Type-C connector and supporting USB, but it is only now becoming available. AVB started to look increasingly as a failure and is almost forgotten. Audinate's Dante dominates in all areas of applications, including live and studio multitrack recording, while the industry is united in a common effort to make the new AES67 interoperability standard a reality, allowing all major AoIP solutions to "at least exchange audio." The new AES70 standard, originated from OCA Alliance efforts, created an architecture for system discovery, control, and connection management for media networks and promises to push AoIP even further.

For 2017, this makes the landscape quite interesting but also challenging. We now know about Thunderbolt 3 and USB-C. We know that AES67 is a reality that doesn't replace any of the AoIP protocols (which means they can continue to coexist). And, we start to have signs that the new AES standards—AES67 and AES70—will effectively complement each other to create completely interchangeable solutions over Ethernet—entire audio systems or specific end-point applications. Oh, and since Thunderbolt 3 carries USB 3.1 and 10 Gigabit Ethernet, it's even possible to have all types of signals on the same cable. So support for all those solutions in the same equipment is not only possible but also recommended.

The question now becomes, not if but when. And which audio manufacturers decide to move first. We understand the product cycle and we understand that we won't see many new products with the new Type C connector at NAMM 2017, but will manufacturers be able to provide the road map that is more necessary than ever? Or will we continue to see new USB 2.0 DACs and audio interfaces be introduced in 2017, as though nothing else has happened?

João Martins
Editor-in-Chief

The Team

President:	KC Prescott	Associate Editor:	Shannon Becker
Controller:	Chuck Fellows	Graphics:	Grace Chen
Editor-in-Chief:	João Martins	Advertising Coordinator:	Nathaniel Black

Technical Editor: Jan Didden

Regular Contributors: Bruce Brown, Bill Christie, Bob Cordell, Joseph D'Appolito, Vance Dickason, Jan Didden, Scott Dorsey, Gary Galo, Gerhard Haas, Chuck Hansen, Richard Honeycutt, Charlie Hughes, Mike Klasco, G. R. Koonce, Ward Maas, Oliver Masciarotte, Nelson Pass, Bill Reeve, Fernando Rodrigues, Steve Tatarunis, Ron Tipton, Stuart Yaniger

OUR NETWORK

audioxpress

VOICE  COIL

 circuit cellar

LIS LOUDSPEAKER
INDUSTRY
SOURCEBOOK

SUPPORTING COMPANIES

ACO Pacific, Inc.	23	Jensen Transformers	35
Acoustic Sciences Corp. (www.acousticssciences.com)	43	KAB Electro Acoustics	23
All Electronics Corp.	41	Linear Systems	37
Avel Lindberg, Inc.	31	Menlo Scientific, Ltd.	67
AXPONA	25	OPPO Digital, Inc.	17
Celestion	11	Parts ConneXion	2
Front Panel Express, LLC	5	Parts Express	68
Hammond Manufacturing Co., Inc.	3	Primacoustic	35
ICEpower	13	Wavecor, Ltd.	29

NOT A SUPPORTING COMPANY YET?

Contact Peter Wostrel (audioxpress@smmarketing.us, Phone 978-281-7708, Fax 978-281-7706) to reserve your own space for the next edition of our magazine.

Highlight your company by sponsoring an article or column.

COLUMNISTS

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

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

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

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

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

The Easiest Way to Design Custom Front Panels & Enclosures

You design it
to your specifications using
our FREE CAD software,
Front Panel Designer



We machine it
and ship to you a
professionally finished product,
no minimum quantity required

- Cost effective prototypes and production runs with no setup charges
- Powder-coated and anodized finishes in various colors
- Select from aluminum, acrylic or provide your own material
- Standard lead time in 5 days or express manufacturing in 3 or 1 days

FRONT PANEL
EXPRESS

FrontPanelExpress.com



● Features

18 Vienna Suite Pro High-Precision Stereo and Surround Audio Plug-Ins Processing for the Orchestral World

By Fernando Rodrigues

Learn more about these essential audio tools for individual track processing and mixing or for the final mastering process, supporting from stereo up to 12.2 surround formats from Vienna Symphonic Library.

26 Primacoustic London 12 Kit Installation and Performance Tests

By Ron Tipton

Ron Tipton puts this acoustic panel kit from Primacoustic to the test in his own familiar listening environment and shares the results. And since this London 12 kit was of the "paintable" variety he even explores his artistic vein.

32 True Bass in a Small Space

By Thomas Perazella

Using a complete kit from Parts Express and Dayton Audio's Ultimate 18" driver,

Tom Perazella shows us how a medium-sized box can replace two 15" drivers in two larger sealed boxes.

52 Colorful Amplification for Ultra-High Impedance Sources

By Addie and Whiskers

An adventure in building a preamplifier for a high impedance piezoelectric pickup goes in an unexpected direction with the finding of a handful of military-specified vintage components.

56 Build a Low-Power, Class-A Power Amplifier (Part 2) Construction and Performance Tests

By George Ntanavaras

In the second part of this article detailing our author's efforts to improve his home stereo system, we learn more about his construction efforts and the outcome of his test results.



● Columns

IT'S ABOUT THE SOUND

- 8 **Streaming Music Experiences (Part 2)**
Streaming Multichannel Audio from Apple TV
By Ron Tipton

STANDARDS REVIEW

- 14 **Bluetooth 5 and Audio Applications**
By João Martins

SOUND CONTROL

- 42 **Room Acoustics and Other Bookcase Necessities**
By Richard Honeycutt

QUESTIONS & ANSWERS

- 48 **Love of Music Leads to New Business Venture**
An Interview with Nancy Moon of MOON Amplification
By Shannon Becker

HOLLOW-STATE ELECTRONICS

- 62 **The Oscilloscope**
By Richard Honeycutt

● Departments

- 4 From the Editor's Desk
5 Client Index
66 Industry Calendar

● Websites

audioxpress.com
voicecoilmagazine.com
cc-webshop.com
loudspeakerindustrysourcebook.com



@audioxp_editor



audioxpresscommunity



linkedin.com/company/audioxpress

Streaming Music Experiences (Part 2)

Streaming Multichannel Audio from Apple TV



Photo 1: The Kanex Digital Audio Adapter is shown with its short HDMI cable plugged into the Apple TV version 4. The Kanex Adapter comes with a wall power supply not shown here.

Are we able to actually enjoy the best sound reproduction with streaming services over the Internet, given the bandwidth restrictions and variable data rates? In this series of experiments with real world services, we tested a 5.1 surround soundtrack of a movie streamed to the latest Apple TV, going through an adapter from Kanex.

By
Ron Tipton
(United States)

A new product from Kanex, the Digital Audio Adapter, enables stereo or 5.1 surround sound listening when used with an Apple TV version 4 (TV4). The Audio Adapter has a short, built-in HDMI cable that plugs into the Apple module (see **Photo 1**). It also provides 5.1 surround sound via standard optical audio, which is no longer included in the latest Apple TV. Because it “piggy-backs” on the Apple unit, it has full access to all the Apple iTunes Store content, but more on this later.

Figure 1 shows the signal flow—wireless connection is discouraged. Running CAT5 or CAT6 cables can be a problem in existing buildings but it’s highly recommended for reliable operation. Bandwidth is reduced by cable splitters, so using separate cables from your router to each listening room is best.

Plex Media Server

I have found that a media server program is useful to keep track of my streamed files. After some web searching, I decided Plex was a good choice. It’s free but there is also Plex Plus, which offers additional features at (I think) a very modest price.

Plex, available for Windows and Mac, keeps track of your Media in four library sub-folders—Music, Movies, TV Shows, and Photos. And it’s best to keep your content in the right folder because Plex looks up (on the web) and displays metadata for your files. Although there is documentation, I found the setup procedure to be confusing until I experimented with it. The problem is knowing what part of the file structure I needed to do and what part Plex does for me.

This is more-or-less explained in the documentation, but I’ll illustrate my point with a Music folder example. In Windows, I create this file structure on my C: drive:

```
\Media\Music\Neil Diamond\Melody Road
```

To add a single track to the Melody Road folder, I hovered my mouse over LIBRARIES on the left-side column and clicked the “+” symbol that appears. I clicked the Music icon and then Next. I browsed for my music folder that contains melody road.wav and clicked Add. (This does not add the folder!) I clicked Next a couple more times and then, finally, clicked

Add Library. (This does add it.) Plex looked up the album information online and displayed the screen shown in **Photo 2**.

Adding a single track is a special case because it can be placed anywhere on your computer—the C: drive, a USB stick, or a CD. But you have to be careful because the added file is just associated with the Melody Road folder but not moved into it. Safe for the C: drive but removing the USB drive or CD would result in a file not found error message. To add more than one track or an entire album, each track must be named in the file structure. For example:

```
\Melody Road
  Track 1 – name.wav
  Track 2 – name.wav
  Track 3 – name.wav
  etc.
```

Take a look at the Plex documentation, included in the Supplementary Material, for more file structure details.

Set up the cabling as shown in **Figure 1** and power on all of the units. Launch Plex and make sure you have a network connection by clicking on Settings. Click Music in the left side column. In my example, I clicked the Melody Road icon and then clicked the Melody Road line. This will load the play screen shown in **Photo 3**. I clicked the Plex Play button. Then, using the Apple remote control, I clicked its Play/Pause button. I also could have navigated to the TV screen Play button and clicked it. This starts the playback.

The “Melody Road” track is stereo and it sounds fine. But how does the surround sound? Using DVD Audio Extractor, I saved my CATS DVD 5.1 soundtrack to a .wav file—about two hours and 4.3 GB at 48 kHz and 16 bits. It too sounds fine but, again, I’m interested in what, if any, distortion is introduced in the process. But, the transmission path is fairly short—Plex computer to the router and then from the router to the Apple TV4 and the Kanex module. A better test would be streaming from a music provider.

iTunes Store

One place to find streaming surround sound is in the soundtrack of a musical movie. I looked at the iTunes Store and purchased a copy of *Moulin Rouge*, which has a Dolby Digital 5.1 soundtrack. The Store displayed in **Photo 4** with a Purchased caption and the TV screen, shown in **Photo 5**, immediately displayed the same album cover under PURCHASED. Navigating to the cover icon with the TV4 remote control, selecting it, and clicking PLAY started the playback.

The picture and sound quality are both excellent, especially in 5.1 surround. Stereo sound was also fine. However, you have to remember to change the Apple TV4 Audio and Video Settings to Stereo or Surround for the Kanex module to properly extract the audio from the HDMI stream. The Kanex audio

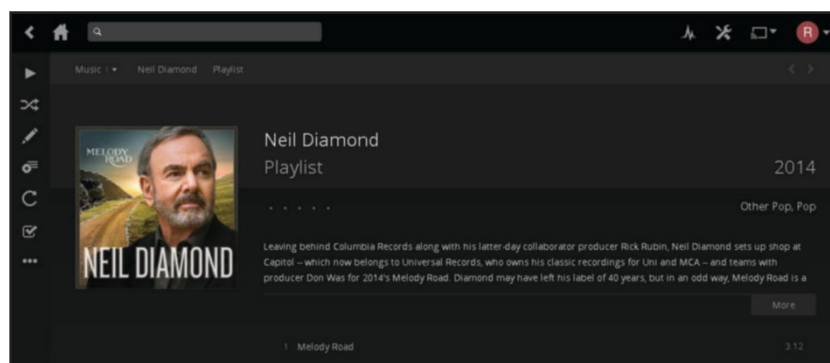


Photo 2: When the Plex media file structure is properly set up, the metadata is accessed online. This displays the album cover and artist information on the computer’s screen.

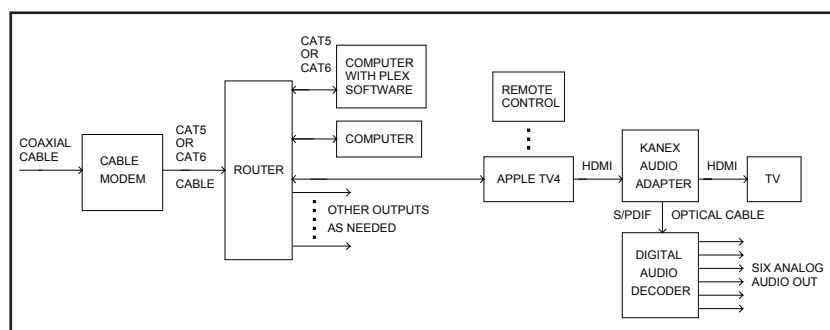


Figure 1: This block diagram of my Apple TV4 with the Kanex Digital Audio Adapter has hard wiring throughout. The incoming cable connection is from Xfinity and shows a consistent speed of 18 to 20 Mbps, which is adequate for HD video according to information on the Apple website. A pdf copy is included in the Supplementary Material as iTunes-downloads.pdf.

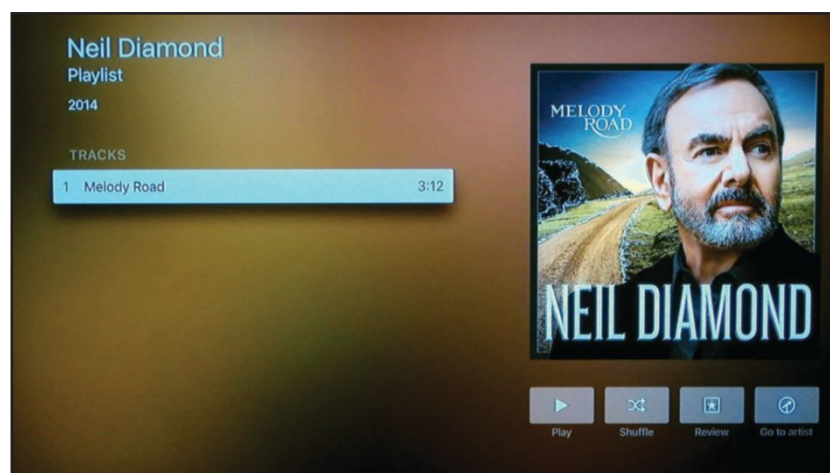


Photo 3: Clicking the Play button on the TV screen with the Apple TV remote control starts playback. Only one track is displayed because I didn’t add the entire album for this test.

output is optical (TOSLINK) so you need an optical-to-analog decoder. I found the decoder shown in **Photo 6**, the J-Tech Digital model JTD-DA-5.1-ANALOG. It decodes Dolby Digital (AC-3), DTS, and pass-through PCM. There are similar looking 5.1

Photo 4: After you have made a purchase from the Apple iTunes Store, it is displayed in your Purchased area. You can download the video MP4 file to your hard drive by clicking the small cloud icon in the upper right corner.

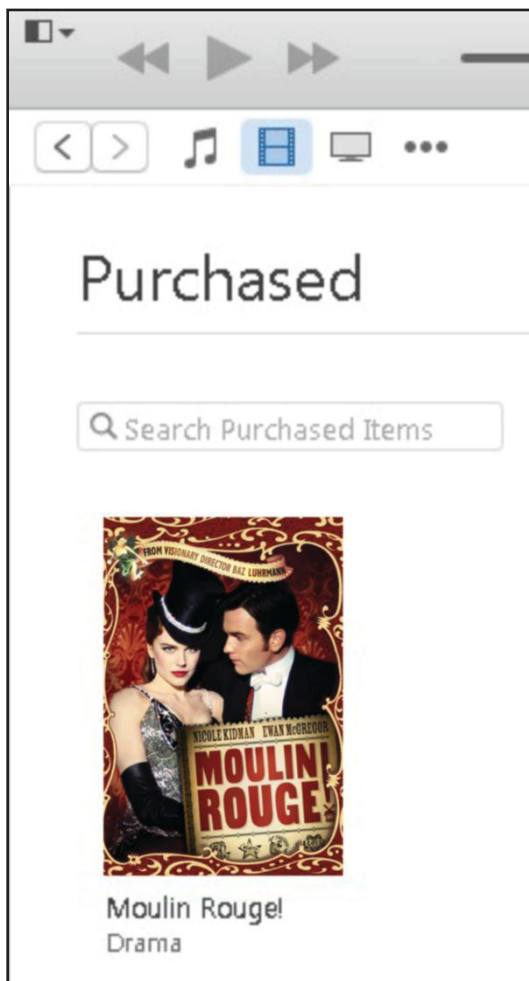


Photo 5: The purchase is also immediately displayed on your TV screen. You start streaming play by navigating to the cover icon with the Apple TV remote control and clicking.

decoders that may work but I haven't evaluated any of them. However, I have read that some just simulate the six-channel output with stereo. Also, if you have an "elderly" AVR without HDMI but with an optical multi-channel audio input, it will also decode your music.

If purchased, you can download the film to your hard drive or stream it as many times as you want, or both. I began with streaming.

I recorded the first 20 minutes of the front left and right channels on several successive days to have comparison files. I wanted to record each playback from the start to make the time alignment easier when extracting measurement segments. The segments do not compare because each 20-minute capture differs in length by several seconds. This is probably caused by bit-rate jitter in the fairly high bandwidth needed to stream video.

Photo 7 shows a time misalignment of 300 mS just 18 seconds from the start of two captures. This does not affect watching the movie—the audio and video are always in sync and you will never notice if the film's duration is a few seconds longer or shorter than a previous watching.

However, if you do several downloads, rather than streaming each time, comparisons are meaningless because they will always perfectly compare. The download is an MP4 video file with a name extension of .m4v.

The Kanex module has a fairly low audio output level, but a \$60 Kanex plus a \$60 J-Tech decoder is an excellent way to enjoy surround music if you don't have an AVR. Connect its digital optical output to a computer sound card and run MusicScope. **Photo 8** shows a True Peak Level (TPL) of -20.8 dB and an RMS of -37.4 dB for a typical 20-minute segment. This does not affect listening to the soundtrack—it's fine. But it does affect a recording because it must be amplified by about 18 dB to make useful



Photo 6: This J-Tech Digital decoder works properly for the Dolby Digital 5.1 soundtrack. It is available from several distributors and from J-Tech Digital in Stafford, TX.

Sound thinking in coaxial design

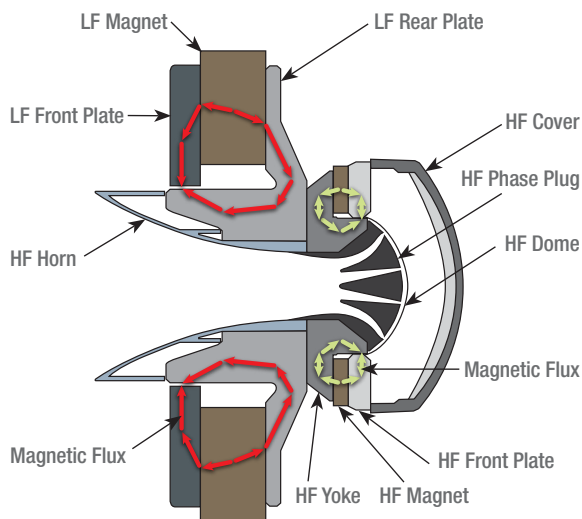
How a common magnet motor delivers a better sounding, lightweight, full-range speaker

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.

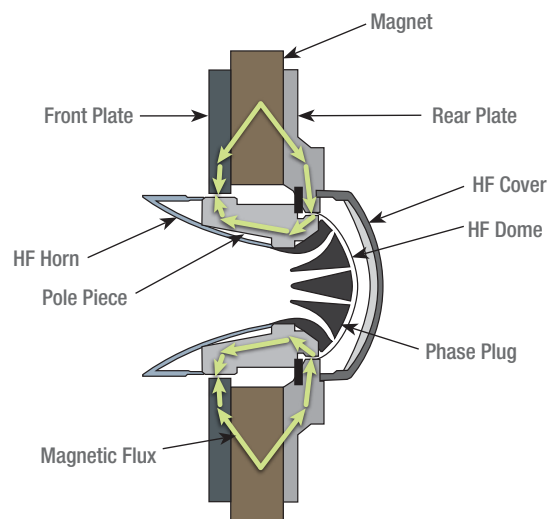
The magnetic flux is optimally distributed between the two elements (calculated using FEA), to achieve the best balance between HF and LF response (see diagrams below). Additionally, the use of a single magnet assembly also means lighter weight and a more compact profile, compared to more conventional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



celestion.com

CELESTION

measurements. I used two recording sound cards: one for the analog recording from the 5.1 decoder and another card for the digital output from the Kanex Adapter. The gain of each card is typical and all volume sliders were set to maximum. The two recorded files had the same TPL and RMS.

Dynamic Range

I took several three-minute segments from the two-hour soundtrack, measured the dynamic range of each and then averaged the readings. The result is graphed in **Figure 2**, and the dynamic range is excellent—nearly a straight 100 dB line.

This agreed with my listening tests—from very soft to very loud. In fact, you need a very quiet listening room to hear some of the dialog because at the same volume setting some of the music will reach at least 90 dB SPL

As in previous articles, I measured the dynamic range with my dynamics.exe program. It runs fine in Windows XP and 7. The executable, source code in “C,” libraries and documentation are included in the Supplementary Material as dynamics.zip.

Netflix

You can also stream Netflix movies and those

Photo 7: A comparison of stereo tracks from two file captures shows a 300 mS time variation between the tracks just 18 seconds from the start. None of my captures of the entire two-plus hours film have the same time duration.

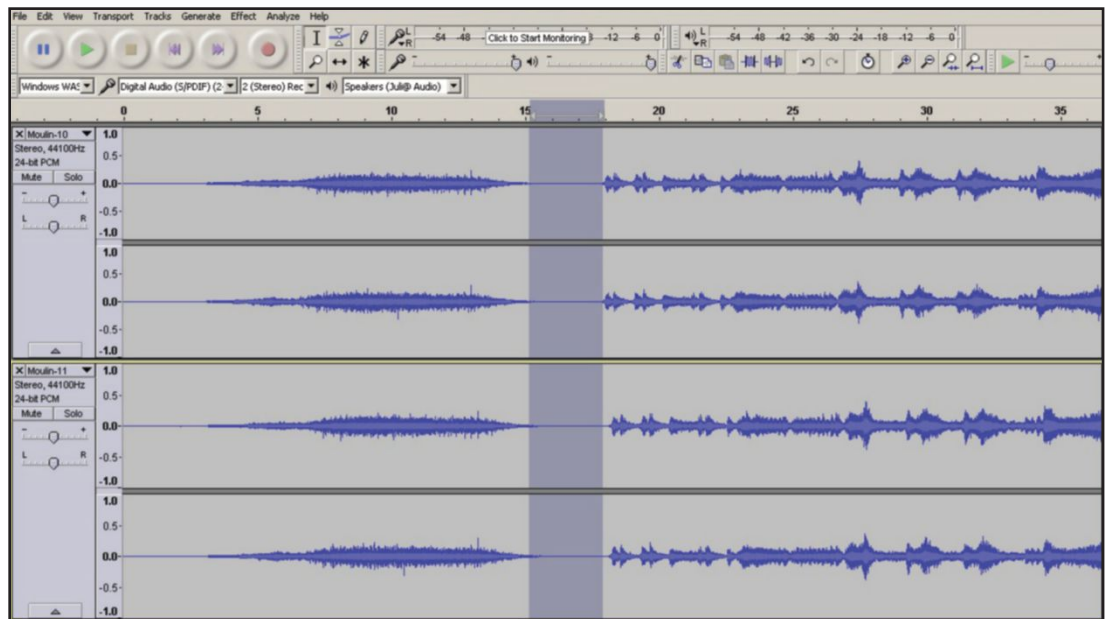
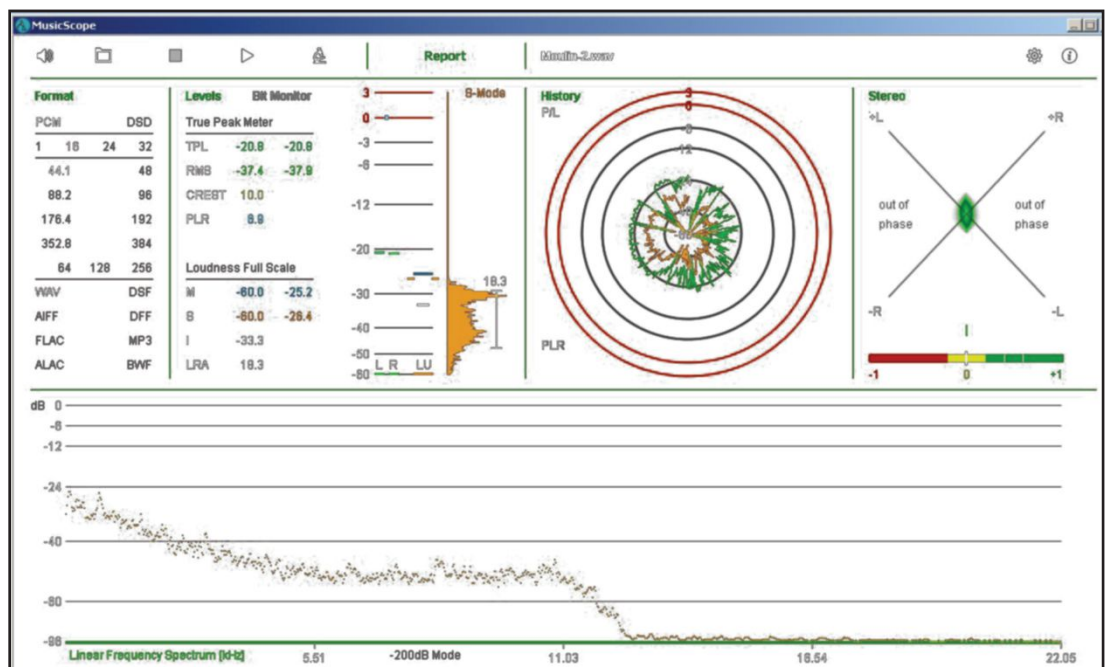


Photo 8: This MusicScope screenshot of a typical 20-minute *Moulin Rouge* stereo capture shows a true peak level (TPL) of -20.8 dB and an RMS value of -38.4 dB. The recording sound card's gain is typical and all volume sliders were set to maximum.



Project Files

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

Sources

Apple TV version 4

Apple | www.apple.com

DVD Audio Extractor

Computer Application Studio | www.dvdae.com

J-Tech Digital model JTD-DA-5.1-ANALOG decoder

J-Tech Digital, Inc. | www.jtechdigital.com

Kanex module

Kanex | www.kanex.com

Plex Media Server

Plex | www.plex.tv

MusicScope

XiVero GmbH | www.xivero.com (Windows & Mac)

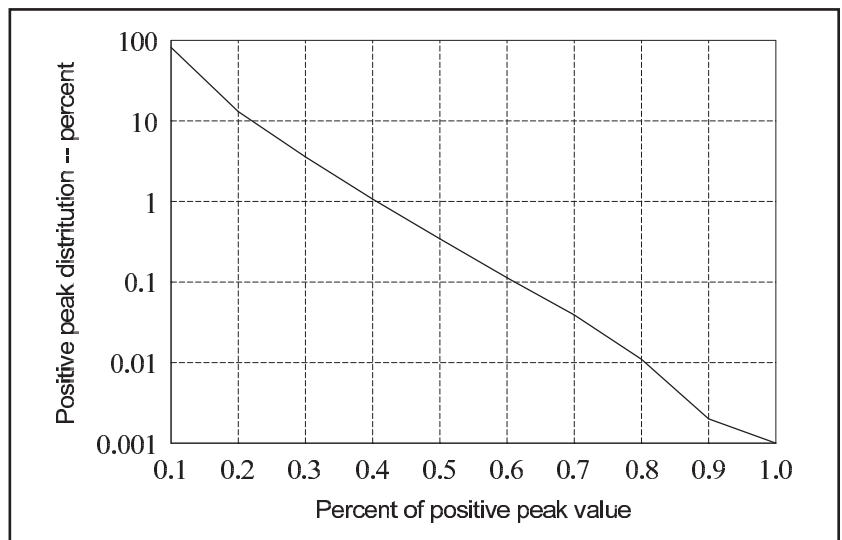


Figure 2: The dynamic range shown here is an averaged value for several segments of the *Moulin Rouge* left front soundtrack. The nearly straight line has no hint of compression.

from other services to your Apple TV4. I don't have a Netflix account so I didn't try it, but the setup seems straightforward. I included the setup and troubleshooting guide as *Netflix-and-Apple-TV.pdf* in the Supplementary Material.

Final Thought

The Kanex Audio Adapter is a useful addition to your Apple TV4 for both stereo and surround listening. It is available from several online distributors as well as directly from Kanex. 



NEW 2017

INTEGRATED CLASS D AMPLIFICATION SOLUTIONS

	100 AS1/2 1 x 100W + 2 x 100W
	300 AS1/A1 1 x 300W + 2 x 300W
	700 AS1/2 1 x 700W + 2 x 700W
	1200 AS1/2 1 x 1200W + 2 x 1200W

**JOIN US AT CES & NAMM 2017 OR
CONTACT US AT CESNAMM2017@ICEPOWER.DK**



ICEPOWER INTRODUCES

New AS series of Integrated Power Solutions for Professional and Consumer Audio Applications

**SUPERIOR SOUND QUALITY
UNIVERSAL MAINS
REGULATED AUXILIARY SUPPLIES
ERP COMPLIANT STANDBY
PRE APPROVED FOR SAFETY & EMC**

Bluetooth 5 and Audio Applications



On December 7, 2016, the Bluetooth Special Interest Group (SIG) officially adopted Bluetooth 5 as the latest version of the Bluetooth core specification. Key updates to Bluetooth 5 include longer range, faster speed, and larger broadcast message capacity, as well as improved interoperability and coexistence with other wireless technologies. This sets the stage for transformative wireless connectivity.

By
João Martins

(Editor-in-Chief)

As the Bluetooth SIG stated, “The increased speed of Bluetooth 5 lays the groundwork for the next generation of Bluetooth audio, and the increased range will deliver reliable connections that make full-home, building, and outdoor use cases a reality. All this while maintaining its low-energy functionality and flexibility for developers to meet the needs of their device or application.”

Bluetooth 5 continues to advance the Internet of Things (IoT) experience by enabling simple and effortless interactions across a vast range of connected devices. The increased speed of Bluetooth 5 lays the groundwork for the next generation of Bluetooth audio, and the increased range will deliver reliable IoT connections that make full-home, building, and outdoor uses a reality.

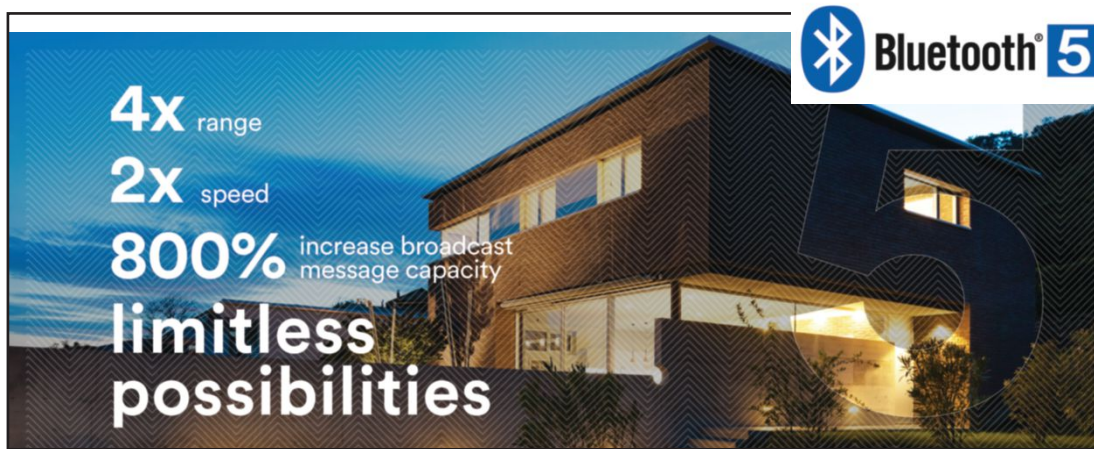
“Bluetooth is revolutionizing how people experience the IoT. Bluetooth 5 continues to drive this revolution by delivering reliable IoT connections and mobilizing the adoption of beacons, which in turn will decrease connection barriers and enable a seamless IoT experience,” says Mark Powell, executive director of the Bluetooth SIG. “This means whole-home and building coverage, as well as new use cases for outdoor, industrial, and commercial applications will be a reality. With the launch of Bluetooth 5, we continue to evolve to meet the needs of IoT developers and consumers while staying true to what Bluetooth is at its core: the global wireless standard for simple, secure, connectivity.”

Key feature updates include four times the range, two times the speed, and eight times the broadcast message capacity. Longer range powers whole-home and building coverage, for more robust and reliable connections. Higher speed enables more responsive, high-performance devices. Increased broadcast message size increases the data sent for improved and more context relevant solutions.

Bluetooth 5 also includes updates that help reduce potential interference with other wireless technologies to ensure Bluetooth devices can coexist within the increasingly complex global IoT environment. Bluetooth 5 delivers all of this while maintaining its low-energy functionality and flexibility for developers to meet the needs of their device or application.

The adoption of Bluetooth 5 comes at a crucial time of industry growth, with ABI Research predicting 48 billion Internet-enabled devices to be installed by 2021, of which nearly one-third will include Bluetooth, according to Ryan Martin, senior analyst at ABI Research.

“The global wireless connectivity market is growing rapidly, with an anticipated 10 billion annual IC shipments by 2021,” says industry analyst Andrew Zignani with ABI Research. “The introduction of Bluetooth 5 will create new opportunities in various verticals of the IoT market by reducing complexity and cost and giving manufacturers greater flexibility in targeting multiple applications and use cases.”



Bluetooth 5 has huge implications for all application scenarios, including audio. Extending range will deliver robust, reliable connections that make full-home, building and outdoor uses a reality. Higher speeds will send data faster and optimize responsiveness. Increasing broadcast capacity will propel the next generation of “connectionless” services such as beacons and location-relevant information and navigation.

Consumers can expect to see products built with Bluetooth 5 within two to six months of its December release. The Core v5.0 Specification document is now available for download on the Bluetooth SIG website (www.bluetooth.com/bluetooth5), as well as an interesting white paper on Bluetooth range, “Debunking the Myth: Bluetooth Range.”

Bluetooth 5 in Perspective

A few days after the Bluetooth 5 specification was published, audioXpress asked for comments from the industry to better understand its impact for audio applications and wireless headphones and earbuds, particularly looking at how the update addresses known problems, such as left/right (stereo) channel synchronization.

We received comments from Mark Powell, Executive Director, Bluetooth SIG; industry analyst and consultant Nick Hunn, CTO at Wifore Consulting, Ltd. and Hearing Aid Working Group Chair of the Bluetooth SIG; Michael Abrams, MD, Chief Executive Officer, FreeLinc Technologies, Inc.; and Jonny McClintock, Director of aptX Sales and Marketing, Qualcomm Technologies International, Ltd. We are sharing those comments here. In the statement from Powell, we learn that the Bluetooth SIG is planning a future update to the specification that was published, specifically addressing audio enhancements.

Mark Powell, Executive Director, Bluetooth SIG

Every new Bluetooth specification release incorporates advancements and improvements to a wide variety of core features. The goal being that all products and applications using the latest specification, including wireless earbuds, will provide a better end-user experience. Ultimately, implementation is key when it comes to creating the best Bluetooth audio experiences. It is clear to us that the future of wireless audio needs low energy and interoperability to work. Bluetooth 5 lays the foundation for the next generation of Bluetooth audio over low energy. The SIG is working closely with our member companies on an audio update

that will address more audio specific enhancements - details on this update will be coming in the future.

Nick Hunn, CTO at WiFore

It's difficult to fully comprehend how much wireless technology has changed our lives. Just 25 years ago, digital mobile phones didn't exist. It was a world before laptops and only a few tens of people around the world had used the precursor of Wi-Fi. Connectivity was largely tethered to the phone on your desk or in your hall and the web didn't exist. Today, wireless is pervasive and billions expect instant connectivity wherever they are.

We can't see the wireless spectrum. If we could, we would be worried. What would have looked like a deserted rural lane in 1990 would now appear as a multi-lane highway close to gridlock. Particularly in the unlicensed, 2.4 GHz spectrum, which is used by Bluetooth, Wi-Fi, and Zigbee, we can envisage things grinding to a halt. That's why Bluetooth 5 is so important.

Bluetooth 5 is the first major Bluetooth release, which is not based on a specific application feature (e.g., high speed or low power). Instead, it provides a package of features that are aimed at robustness. These don't just enhance performance, with longer range, higher throughput and greater payloads, but also concentrate on better reliability and coexistence with other wireless technologies, both in the same band and in adjacent spectrum. It builds on 15 years of experience of being the world's most successful personal wireless standard, to create the foundation for increased performance and user experience over the next 15 years.

It's a very brave decision to concentrate on solving problems before they appear, but one that is vital to future applications. Behind the scenes, work is going on aimed at more specific application areas. We will shortly see the release of a Bluetooth mesh standard, followed by a second generation of audio, which supports the new ways in which we will use voice. Both of these applications will work better because of the features of Bluetooth 5.

Bluetooth started out in mobile phones, but

today it has spread far wider. From children's toys to smart home to medical devices, wearables and factory automation, retail beacons, wireless headsets, and hearing aids, Bluetooth has become the wireless connection of choice. We will not stop using wireless. However, the wireless spectrum is not a flexible commodity. We cannot increase it, so we have to use it more efficiently. Bluetooth 5 embodies that recognition and lays the groundwork for an on-going generation of amazing, new applications.

Michael Abrams, MD, Chief Executive Officer, FreeLinc Technologies, Inc.

When considering the new Bluetooth 5 specifications, it's important for us to remember that at its core, Bluetooth is simply a wireless LAN platform that was adapted to offer wireless Personal Area Network (PAN) solutions. By providing a longer range with higher data rates and by leveraging recent improvements in DSP to avoid the increases in power consumption, Bluetooth 5 appears to be a shift back toward those roots as a wireless LAN platform... while still providing the wireless PAN capabilities. Bluetooth 5 appears to fill gaps in the Wi-Fi ecosystem by offering a "mobile" wireless LAN platform for the Industrial IoT and M2M markets. It may also create value in the Mobile Consumer ecosystem by offering a more seamless integration of consumer devices with geolocation and CRM services. However, in an attempt to do many things, we always have to wonder whether it's possible to do any of them very well.

Bluetooth 5 does not appear to be offering new solutions for the security and the reliability problems that have plagued its predecessors. And depending on how these new capabilities are deployed, Bluetooth 5 may worsen the spectrum crowding in already-congested areas, may decrease reliability for the end-users, and may increase the Negative Networking Effects already being felt along the Bluetooth supply chain.

As with previous releases of Bluetooth, there seem to be rumors spreading that a more reliable point-multipoint networking experience is just on the horizon, but Apple's apparent difficulties synchronizing even the left and right channels of its AirPods, using its own W1 chip suggests otherwise. Given the previous iterations of 4.0 will remain the WPAN solution in Bluetooth 5, improvements in the point-multipoint networking performance seem unlikely.

In conclusion, it's always important to recognize what you do well, and to leave to others what you don't. I applaud Bluetooth 5's strategic shift back toward its roots as a wireless LAN solution, while continuing to support the wireless PAN ecosystem it helped to create. Hopefully, others will see this as an opportune time to add new and complementary wireless PAN solutions, and to step in where Bluetooth has begun to step back.

Jonny McClintock, Director of aptX Sales and Marketing, Qualcomm Technologies International, Ltd.

As someone who has been in the wireless technology industry for more than 20 years, I'm no stranger to the fact that consumers haven't always been receptive to Bluetooth for music streaming. The technology suffered from a bad reputation, known for delivering a lower quality—and an often frustrating—music streaming experience. But over the past few years, this perception has dramatically changed, and consumers seem to be just as comfortable wirelessly streaming music from their phones as they are making calls on them. So why are they suddenly more willing to cut the cord?

In short, Bluetooth has dramatically evolved. The original limitations for music streaming were partly due to the implementation of the SBC codec. Manufacturers initially used its lower "Bit Pool" value, which resulted in a sub-optimal audio performance on most devices. As a result, Bluetooth simply could not be used for high-quality streaming, and music-loving consumers were unwilling to make the move to wireless. What the industry needed was an alternative technology that could deliver consistent CD-quality audio.

But new technologies such as Qualcomm aptX audio, combined with improved silicon platforms and updates to the A2DP profile, have narrowed the gap on cables to a point where no discernible difference can be heard between wired and wireless audio. This means Bluetooth is no longer the acoustic compromise it once was. When combined with the right technology, it can provide excellent performance, battery life, and ease of use. All big wins in consumers' eyes, as these capabilities are available in a wide variety of high-performance audio equipment such as headphones, soundbars, speakers, and automotive and home entertainment systems.

Bluetooth 5.x enhances the Bluetooth Low Energy (BLE) packet structures to enable new audio use cases not previously possible over BLE. The features of enhanced data rate means you can now get music over a BLE link and enabling some improved use cases (e.g., broadcast audio) and reduced lip sync issues when linked to a video source. The continuation of improved consumer experience is something that we at Qualcomm have taken very seriously as witnessed by the widespread adoption of aptX and the recently released aptX HD to address the burgeoning Hi Res music market. Our road map will support and leverage the new features of the standard as and when it's ratified. 

The Core v5.0 Specification document is available for download from the Bluetooth SIG website (www.bluetooth.com/bluetooth5). Any further comments regarding audio applications with Bluetooth 5 and its impact in the industry are welcome and may be submitted by email to editor@audioxpress.com

oppo

Music Without Limits

Sonica DAC will change the way you enjoy digital music in your home. Connect to a computer over USB, or sync up with your favorite devices over AirPlay or Bluetooth. Then enjoy incredible streaming from Spotify or Tidal, or select from the songs on your smartphone with the convenient Sonica App.

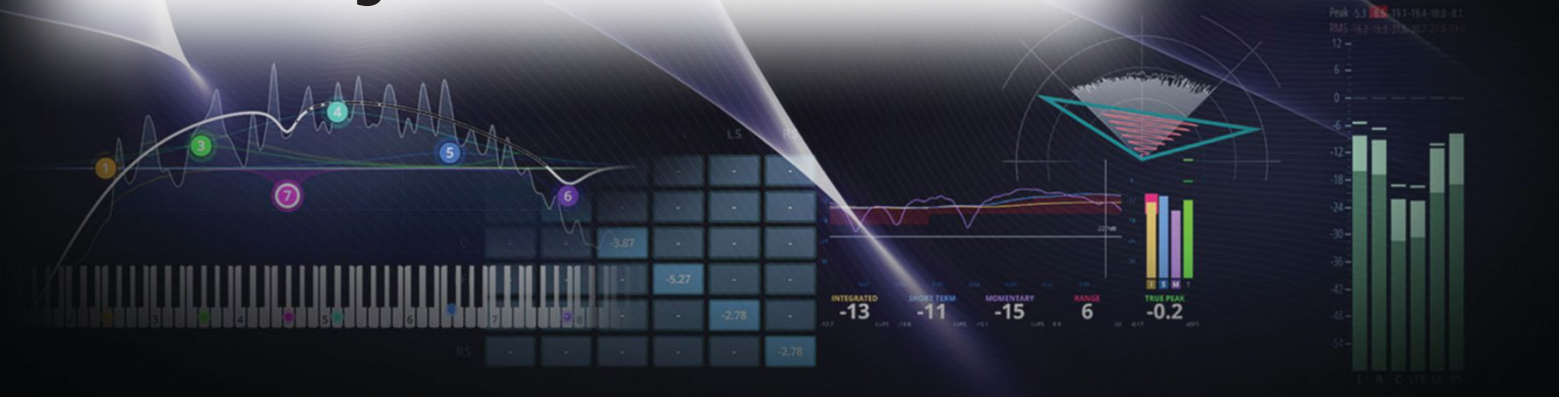


   @oppodigital

OPPO Digital, Inc. | www.oppodigital.com | Menlo Park, CA

Vienna Suite Pro High-Precision Stereo and Surround Audio Plug-Ins

Processing for the Orchestral World



A software development from the famous Austrian company Vienna Symphonic Library (VSL), Vienna Suite Pro includes a collection of 14 64-bit high-precision plug-ins for a variety of audio-processing tasks—from dynamics and filter processing plug-ins to analysis plug-ins. This is a review of these essential audio tools for individual track processing and mixing or for the final mastering process, supporting from stereo up to 12.2 surround formats.



By
Fernando Rodrigues
(Portugal)

The Vienna Suite was created by the famous Vienna Symphonic Library (VSL) purposely for use with its orchestral libraries. Of course, it doesn't have to be reduced to that role and we've tried it with other material, mainly orchestral, but not from VSL.

I have been a user of the Vienna Suite since it first appeared. It always pleased me because of its clear sound, clean interface, and ease of use. If there ever was a prize for what a true modern plug-in suite should be, Vienna Suite would certainly be a good candidate. Therefore, I immediately contacted VSL as soon as it launched Vienna Suite Pro. As I expected, VS Pro is a 2.0 version of the old Vienna Suite. Some plug-ins have been merged (the EQs and the reverbs), others have been refined, and some more have been added, to make the bundle more attractive and comprehensive.

All in all, the new version of the Vienna Bundle Pro may be of great added value for any user of the Vienna Symphonic orchestral libraries, and that said,

also to any user of orchestral libraries. It comes with plenty of presets specifically designed for the different instruments and sections of the symphonic orchestra, sometimes identified with the names of specific VSL libraries. In spite of having been designed with those in mind, I found them good starting points to use in real situations with different libraries and also with real orchestral recordings.

What's in the Bundle?

VSL streamlined its suite to make it more suitable to current trends. That doesn't mean it assumed any compromise in terms of sound, although some concessions have been made in terms of the graphical user interface (GUI). In fact, the graphics in the Pro versions seem more in the vein of the current "flat" trend, so recently debated, mainly because of Windows 10. But although the GUIs became "flatter," they are still polished and became scalable and high-resolution (retina ready). The GUI is now vector-

based, which allows for extreme resizes (from 75% to 200%). The contextual help—that used to appear in small yellow labels when we drag the mouse over the control—is now displayed in the bottom of the GUI window.

The entire suite is now also surround ready, which is understandable if we think that VSL libraries are among the most used tools among composers, especially screen soundtrack composers. Some plug-ins also gained a common feature called Group Section, allowing us to assign the available channels (which, in a surround environment production, can go up to eight) to different groups, which can be processed differently. So, it's like having eight processors in one. This is one of the most important innovations in this Pro version of the package.

On the other hand, VSL merged things that could be redundant and created other plug-ins that may be suitable, especially in surround environments. That's how the two reverbs are now just one, and the two equalizers are now also just one. The Equalizer and the Multiband Compressor/Limiter also went through a big remaking, significantly expanding their processing capacity.

The new plug-ins are basically “utilities,” like the Loudness Meter, Matrix Mixer, Surround Balance, and Surround Pan. These are added to several other utilities that were already present in the previous Suite (e.g., the Analyzer and the Goniometer), which may be a sign of the extreme care and concern VSL has for what may become the final result of our mix.

So, has anything else changed under the hood? That's what we are going to find out.

An Approach That Respects the Original Sound

When I reviewed the “old” Vienna Suite, years ago, I admired the approach followed by VSL with a big respect for the sound “as is.” The processors can be characterized among the most transparent that I have had the opportunity to try. Some might not favor this, but from my perspective, a suite like this one is always welcome, even if we sometimes wish for something with more “color” or more presence. It's easy to understand why transparency is the key word here—if we think that the people who created these tools are the creators of libraries that intend to recreate the sound and the language of the symphonic orchestra, with all its richness, diversity and nuances. In fact, the processors feature many presets directly addressed to instruments and ensembles present in the different VSL libraries. I found that they sounded very well, even with other libraries. The presets revealed that they were well-thought-out for the particular sounds for which

they were intended, even with different orchestral libraries, and how important user actions are to the final sound. Of course, it is with the VSL original libraries, which I also tried, that they really shine.

I privileged the compressor, the equalizer, and the limiter in the channels, with some occasional use of the Imager and the Exciter, just to hear what



Analyzer Pro is one of the “utilities” included in the bundle. The Analyzer in the previous bundle was one of the more revered plug-ins of the suite because of its comprehensive set of tools and clean interface. The Analyzer Pro refines the approach.



Compressors and EQs are the basis of any basic set of processing tools. With some refinements, both remained in the current version of the suite, and for a good reason: With a clean approach to processing and great respect for the sonic material, these are ideal for orchestral works.



The EQ was deeply revised, with more bands and several filter types, but the sonic approach, remained and the presets are basically the same.

they were able to do (in good or in bad ways). I created extra busses to try the Hybrid Reverb and the Multiband Pro, and I used the analysis plug-ins

(e.g., Analyzer, Goniometer and Loudness Pro), and the Master Channel, together with another instance of the Limiter, just as a measure of caution. I also applied the analysis tools to some of the tracks, as these are helpful (something I already knew from the previous suite).

At Work

All of the plug-ins mentioned earlier were used in productions with a considerable number of tracks. Yet, the CPU load was still low, giving me the confidence for more ambitious work.

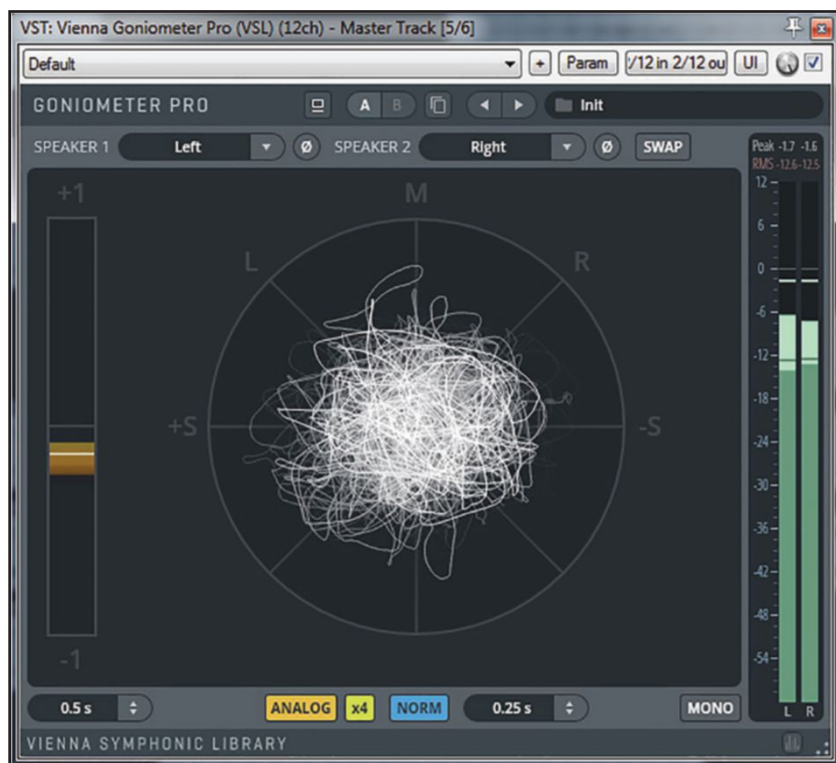
The compressor, at least at a first look, didn't change much. I tried the old and the new with similar parameters, and the results were pretty similar. The presets appear to be the same, too, except that the final gain (Make Up) is usually reduced by as much as -6 dB in the Pro version. I didn't find a reason for this, since, in my tests, the overall gain of the two seems almost identical. As a matter of fact, the Pro version is even able to achieve a bigger gain reduction. With the Make Up at the same value, the final gain after the Compressor Pro is still a little lower than the one with the previous versions. Other than that, in terms of sonic color, they seem very similar.

The equalizer is a totally different beast, and it underwent a big revision for this edition. VSL added up to eight bands and seven filter types. That's why we no longer have a Master Equalizer—the current one is powerful enough to take the role, if needed. And it does more with less. With the current power available in the computers, I could run many more instances in the mix tracks. As mentioned, the EQ preserves the transparency, yet is much more versatile. We can sculpt the sound with high precision, and get it exactly where we want.

I only did a small comparison with the previous edition, using it in just a couple of tracks with simple parameters and four bands (the old Equalizer “only” goes up to five). All in all, I ended up with the impression that the new EQ enables a somewhat better presence and definition. Yet, the main quality of the tool, the one that I appreciated the most, is still there—the transparency. Faithfulness to the sound remains the main characteristic, as it does throughout the entire package.

The “Bigger” Tools

The new Equalizer is one of the “bigger” tools in this suite, together with the Multiband compressor/limiter and the Hybrid Reverb. VSL also created the fabulous MultiImpulse Response (MIR) technology for those seeking more sophisticated convolution tools, but this Hybrid Reverb does a good job in the creation of the right ambiance for the sound.



The Goniometer looks good, is very responsive. Now it can be resized until it's really huge!

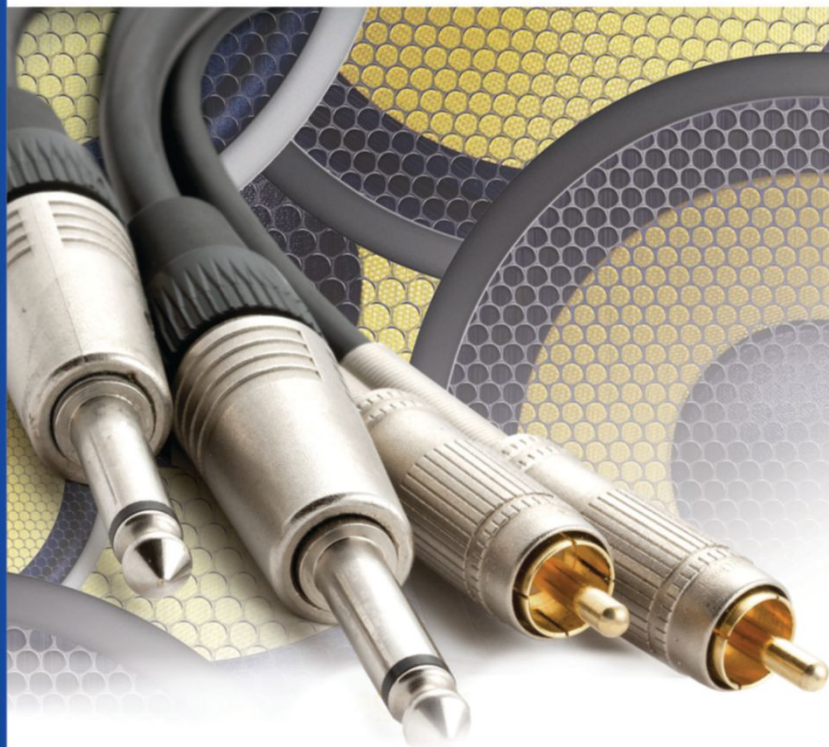


The Hybrid Reverb fused together the two reverbs present in the previous package. While it sounds good and performs well, we would like to have (way) more impulses.

About the Author

Fernando Rodrigues began studying music and technology in the 1970s. His goal was to marry his two passions: music and computers. As a student, he helped assemble the electronics music studio at the music college in Porto. Later, he directed the technology department at one of Portugal's major distributors while pursuing a career teaching musical analysis and composition techniques. He now concentrates on research and writing about music and technology, sharing his own perspectives about music and sound.

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY



Get great articles, projects, tips, and techniques for innovative solutions in sound, acoustics, and electronics.

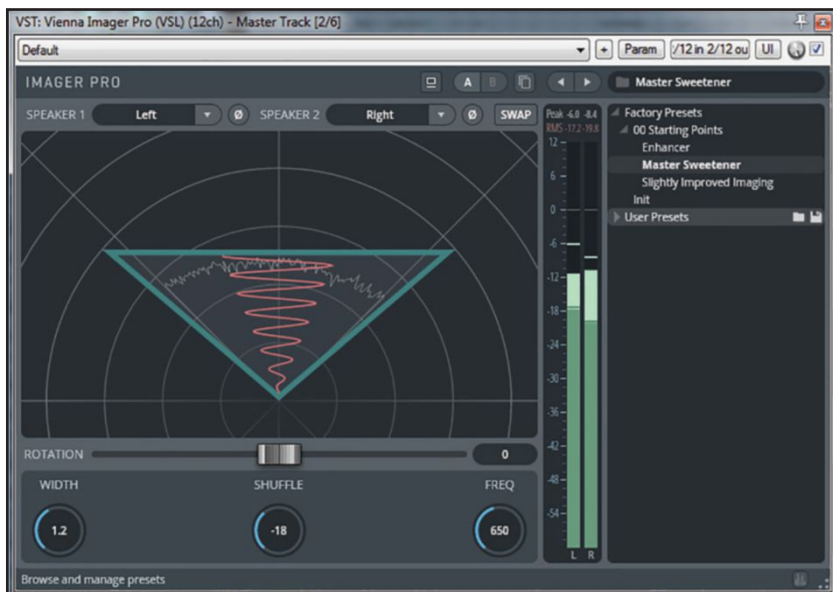
Subscribe at
audioxpress.com

Also available on:



Industry Professionals can qualify for a FREE subscription to Voice Coil. Visit: voicecoilmagazine.com





One of the “new” utilities included in this package was a pleasant surprise. In a classical approach, it’s not usual to mess with the stereo signal. However, this Imager Pro is really powerful and allows users to do much more than we would recommend.



The Vienna Limiter Pro is impressive. It can go really loud and remain clean. And it can be a lifesaver.



In this picture, you can see one of the additions VSL did to the GUI—the Group Section (located in the extreme left side of the GUI). This may be helpful for those working in surround. With the Analyzer Pro, we can have a graphic analysis assigned to each Group and have all of them displayed at the same time (with different colors).

I tried the Multiband center/left (C/L) and the Hybrid Reverb with some of the more complex projects on which I’m currently working. The reverb, in particular, was crucial to my intentions for particular ambiances. Maybe I’m spoiled by the huge amount available in Altiverb, but I think VSL could expand the number of impulse responses.

To be fair, we now have much more than was available in the previous version, even with the Convolution Reverb and the Hybrid Reverb combined. However, many of the files are of the same room, only from a different position (marked as L, C, R, for example, which means they are ready for surround use). This is because of the different architecture of the convolution engine in Hybrid Reverb Pro.

In the end, we are limited to just a few environments. Certainly, it is true that the plug-in is “hybrid,” which means we can still create ambiances with the algorithmic part. Therefore, we can do more than we could with just a convolution only reverb. But, in my opinion, the relatively small number of impulses is something VSL needs to address. Still, this hybrid reverb enabled me to achieve good results with a combination of the two engines.

The Multiband is simultaneously simple and powerful. We don’t have a fixed number of bands. What we have is a line for the threshold and another for the gain, where we can create “nodes” and move those nodes to the value we want. Besides this, we have just a few more parameters—overall threshold (which simultaneously adjusts all the values defined in the curve), Attack, Release, Spread, and the final Gain. We also have some more specialized controls that define whether the amount of compression affects other tracks in the same group (for surround), extent to which the channel compression will be globally linked and the compression strengthened. Usually, the first of these two last parameters is set at zero and the strength at 100%. Working with this was a total breeze.

The “Utilities”

Why did I put the utilities in quotes? Well, because the utilities are sometimes neglected. Yet, the ones presented in this suite are really deserving of their utility classification. The old Analyzer was praised because of its simplicity and precision. The Goniometer is also precise and responsive. This new edition introduces a few more, including two dedicated to surround work. In total, we have seven plug-ins in the suite that we can classify as “utilities”—Analyzer, Goniometer, Powerpan (these existed in the previous version), Imager, Loudness, Matrix Mixer, the Surround Pan, and Surround Balance.

The Analyzer Pro takes advantage of the new Group Section to allow the definition of up to eight groups. Each group has its own analyzer graph, and we can have all of them displayed simultaneously, if we want. By default, all channels are assigned to Group 1. It also has two modes—Analog and Digital. The difference between them lies in the way they analyze the audio, with the digital based in Fast Fourier Transform (FFT) with the weighting filter, and the analog in a Analog Reference Pitch (and, we suppose, analog filters).

The Goniometer Pro was optimized (according to VSL) and takes advantage of a much bigger display. It also gained a GLOW trailing feature to enable us to have a better idea of the signal evolution.

The Powerpan Pro was also improved over the previous version, with tools to handle surround like a different speaker pair selector, and four tools for signal balancing, all available directly in the main window—Gain, Delay, Spectral, and Phase.

Imager Pro is one of the new tools. It is destined to enhance the stereo impression of a signal pair or to correct faulty stereo signals. Anyone who already worked with Waves PS22 will be immediately at home



Another powerful and flexible tool with a different approach. Like all the other plug-ins in the suite, Multiband Pro also offers a Group Section, allowing linking the attenuation of channels into groups. Internally, it operates like a dynamic equalizer.

with this one. The working principle is basically the same. There are just a few, very effective controls.

Loudness measures the perceived volume of the sound, working in a way similar to the human hearing. It provides indication of compliance with the EBU R-128 and other loudness standards, and it is also useful to help calibrate audio sources. Another welcome addition to the Pro package.

Matrix Mixer, the Surround Pan, and the Balance are three utilities designed with surround productions in mind. The first helps to solve problems like feeding of L/R channel content into the low-frequency effects

For Over 35 Years

ACOustics begins with **ACO**™



Community/Industrial
Noise Monitors/Alarms
Indoors and Out

Measurement
Microphones
Polarized and Electret



Windscreens
3 to 18 inch & Custom
Weather/UV Resistant



Microphone Systems
Standalone
IEPE Powered
Phantom Power
Custom



ACO Pacific, Inc

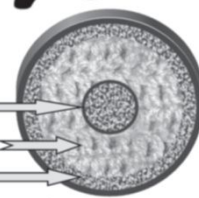
www.acopacific.com sales@acopacific.com

Tel: +1-650-595-8588

Spiral Air™
Interconnects

*Coax
Is Cool
Again!*

Cardas Golden Strand
Air Filled Cotton Dielectric
Unique Spiral Litz Shield
ETI Tellurium Copper Connectors



Affordable Litz Interconnects and Bulk Wire
Tonearm - Phono - Line - Loudspeaker

KAB

Preserving The Sounds
Of A Lifetime

www.kabusa.com



(LFE) channel, solving phase problems of individual channels, enabling phase adjustment of a single channel in a surround setup, or even create pseudo-surround by sending phase-inverted L/R channels to the opposite side of the sound field.

Surround Pan Pro and Surround Balance Pro are exactly what their names imply. Of the two, perhaps the later will be of more utility, given the fact that the former is usually present in most digital audio workstation (DAW) software. However, Surround Pan Pro enables users to position a stereo source to any surround format, and this is not common.

Conclusion

The Vienna Suite was already a favorite of mine. It is a well-thought-out GUI and faithfulness to the audio is one of the characteristics that attracted me to it since the beginning. Sure, sometimes it's useful to have colorful processors, but the market is full of such options, including some we reviewed. On the other hand, there aren't that many fully transparent processing options.

Vienna Suite Pro remains faithful to the principle of its predecessor package, while rounding the bundle with some useful additions. The GUI redesign is also welcome, since it now allows for much-needed window resizing and supporting higher resolutions in large-sized monitors.

All in all, this bundle is a must to work with orchestral libraries, soundtrack work, or any other kind of music with orchestra and large ensembles. For those working with surround formats, the package adds extra utilities, and well-thought-out additions without quality compromises.

While this software package may look very pro oriented (and it is), it remains compatible with popular 32-bit applications, like Sound Forge Pro. This means that VSL provides a high-quality software-processing package for a wide user base, even if VSL usually targets the more demanding users. 

Vienna Suite Pro—Sound Processing Plug-In Suite

Available in VST, VST3, AU, and AAX Native formats.

System Requirements:

- Windows 7/8/10 (latest Service Pack, 32/64-bit), Intel Core 2 Duo/ AMD Athlon 64 X2 (or better).
- Mac OS X 10.8 (latest update) or higher, Intel Core 2 Duo

Uses eLicenser USB key and eLicenser Control Center activation software

Vienna Symphonic Library | www.vsl.co.at

Software from Vienna

Vienna Symphonic Library is a research-driven music software and sample library developer based in Vienna, Austria. The company was founded by Herb Tucmandl in October 2000. Since then, more than 2 million samples of nearly every instrument of the symphonic orchestra, choir and more have been recorded and published. With over a terabyte of high-resolution sound, it is by far the largest sample database ever created by a single company.

Using orchestral libraries such as the Vienna Symphonic Library, music professionals can accurately and authentically recreate orchestral pieces on their computers. This enables composers to produce not only a printed score, but also an audible and convincing rendering of their music. By rendering their orchestrations, composers can more easily influence producers, directors, publishers, and concert organizers to perform or record a new opus, without necessarily having to book expensive orchestras, concert halls, or recording studios for a demo. Furthermore, any changes of instrumentation, tempo, length, or key can be dealt with, quickly and easily. Even pieces that have until now been considered “unplayable” can be realized with this new technology.

The list of Vienna Symphonic Library users includes musicians, composers, arrangers, and studio owners on all continents. Well-known TV and film music composers such as Danny Elfman, A.R. Rahman, Alan Silvestri, Alexandre Desplat, and many others use their “Vienna Libraries” (as some like to



The famous Vienna Konzerthaus (Great Hall)

call them) to create early versions of their movie scores. Increasingly, composers are also combining virtual orchestra sounds with live musicians in the final mix. TV series such as *CSI*, *The Mentalist*, and *Dexter*, and many commercials are almost exclusively created “inside the box.” Tight production schedules and limited budgets usually don't allow for recordings of real orchestras, so the sample-based recording isn't just a mock-up, it's the final cue.



Herb Tucmandl, the founder of the Vienna Symphonic Library (VSL)



AXPONA

AUDIO EXPO NORTH AMERICA

APRIL 21-23, 2017



DISCOVER YOUR SOUND.

THE North American High-End Audio Show.

The world's best hi-fi systems will be playing in over 125 listening rooms. Experience thousands of products including loudspeakers, audio components, vinyl and headphones, all in one place. If you have a passion for sound, AXPONA is the place to be!

THE WESTIN O'HARE ROSEMONT ROSEMONT/CHICAGO, IL



For show and ticket Info, visit www.axpona.com

The Chicago area is known for its vibrant music scene, great food, museums, and shopping. There is so much to do ... downtown, and all around the city and suburbs. AXPONA takes place at the Westin O'Hare, minutes away from the airport with easy access to downtown attractions. When you're at AXPONA you won't have to go far to enjoy all that Chicago has to offer.





Fresh From the Bench

Primacoustic London 12 Kit

Installation and Performance Tests

Primacoustic's London 12 Kit contains acoustic panels that tackle problems such as primary reflections, flutter echo, standing waves and excessive bass. In this article, the kit to the test in an actual listening room.



Photo 1: The London 12 Kit contains 22 high-density fiberglass panels covered with an acoustically transparent fabric to eliminate fiberglass shedding. Mounting hardware is included for drywall and paneled rooms. (The lady is not included.)

By
Ron Tipton
(United States)

Recently I received a London 12 Kit from Primacoustic so I could evaluate it in my music listening room. This kit, recommended for an up to 120 ft² room, fits my 136 ft² space fairly well. The kit contains 22 high-density fiberglass (6 lb per cubic feet) sound absorption panels covered with an acoustically transparent fabric to prevent the panels from shedding (see **Photo 1**).

The kit arrived in two cartons—one at 72 lb and the other 50 lb. It took the better part of an hour to do the unpacking. I didn't weigh it, but I suspect the packing was about half the weight. Heavy-gauge corner covers were used to protect all of the larger panels and an abundance of cardboard and Styrofoam was added to protect the panel surfaces. The debris nearly filled my pickup truck when I took it to the recycle bin.

The Panels

Primacoustic has a unique panel mounting technique that is designed to do minimum damage to your walls. The company provides the stamped-metal "impalers" shown in **Photo 2**, which you screw

to your walls. The panels are held in place by the sharp points and are easily removed if you want to relocate them. All the necessary hardware is provided for drywall and paneled rooms.

The absorption panels are available in three standard colors: gray, beige, and black. The covering fabric is dyed to these colors before the application to the fiberglass to preserve its acoustic transparency. And, Primacoustic has recently introduced a new color: Absolute White, which is paintable. The kit I received was of this variety, and I closely examined the panels. They are actually painted white, which does have some effect on the acoustic transparency. But, according to the company representative I discussed this with, the effect is minimal if the repainting is done according to the instructions—a copy is included in the Supplementary Material (found on the *audioXpress* website).

Primacoustic's preferred method is compressed-air spray. I don't have the proper equipment for this, but more importantly, I don't have the space to do it. I don't object to white but I did, at least, want to add some splashes of color. I decided that

a very dry roller might do what I had in mind. I used a 3" wide roller with a flat, blue-gray water-based paint that was a nice contrast to my room's colors. Rolling most of the paint off on newsprint left the roller very dry, and it was easy to pick up more paint from the newsprint. I put an abstract pattern on one of the 12" x 12" panels and let it thoroughly dry. Then, with my USB microscope, I examined one of the original 12" x 12" panels and my painted one.

I tried several magnifications but $\times 7$ seemed to give the best results. **Photo 3a** shows a 0.71" x 0.53" area of an original panel. **Photo 3b** shows the same area of my painted panel. Both images show that about half of the acoustically transparent pores are painted over. My dry roller basically painted the "high points" of the original white paint. This satisfied me that my painting didn't make any further change to the acoustics.

I put an abstract design on the other 11 smaller panels and then got bold by adding some splashes of "Toasted Honey" to about half of them. You can see the results in the room photos shown in **Photos 4–7**.

Panel Location

Everything in my room, except for the Primacoustic panels, was as you see it in the photos. This made it difficult to follow the company's panel location guidelines. The guidelines recommend a LEDE placement, meaning "live end and dead end." In the case of a listening room, the live end is at the sound source with the listener at the dead end. Therefore, most of the absorption should be at the listening end.

My one totally unobstructed corner was right behind my chair so one of the 2' x 4' panels was corner mounted with four of the 45° impalers as a corner bass trap. The other 2' x 4' panel was placed on the end wall spaced 3.5' from the wall on special impalers. This is also a bass trap arrangement because it's spaced from the wall. (The other end wall corner is actually a door so I didn't corner mount the second large panel.)

I used the "mirror" procedure for locating most of the 1' x 4' panels. Sitting in my chair, I asked a friend to run a flat hand mirror along each wall. When I saw a reflection of one of the column stereo speakers, the reflection became a panel location. The other panels were located in an attempt to absorb secondary reflections.

This is perhaps a good place to mention that this kit's acoustic treatment is for a stereo listening room where you want 20% to 30% of the wall area covered by absorbers. A surround sound room is usually recommended to have 40% to 60% coverage



Photo 2: The stamped metal impaler plates measure about 3" x 4". Attach them to your wall with the points facing up. I found using a large screwdriver or small pry bar to bend the points closer to 45° made it easier to attach the panels. Place a plate against the wall, mark the hole locations, and drill two 0.25" holes (the drill bit is provided), insert the plastic expansion plugs and affix the plate with two of the provided screws.

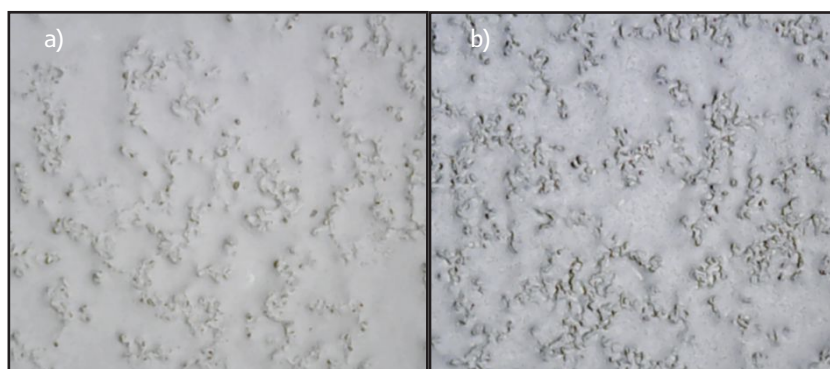


Photo 3: (a) This 0.71" x 0.53" area of an as-received 12" x 12" panel shows the factory's white paint covering about half the acoustically transparent pores. I was assured this did not diminish the panel's performance. (b) A same-size area of one of my dry-roller painted 12" x 12" panels shows my paint covering just the high-points of the original white paint, leaving about half the pores open.



Photo 4: This is the front wall of my listening room. The stereo speakers, center speaker, and subwoofer are part of the 7.1 setup, but just the stereo speakers were used for testing. The blue curtain does not cover an actual window as described in the text.



Photo 5: Moving counter-clockwise, this photo shows more of the Primacoustic panels, a computer used as a music server, and the equipment rack containing, among other things, eight power amps.



Photo 6 : This shows the alcove I created by removing a walk-in closet wall. One of the 2' x 4' panels is corner mounted and the other 2' x 4' panel is mounted 3.5" from the wall with special impalers. Both mountings provide bass trapping. Also shown is the right height speaker, part of the 7.1 setup, and three of my painted panels. (Perhaps I have a future in abstract art.)

because of the multiple sound sources. Although my room is set up for 7.1, I do listen to a lot of stereo music. But I've found that even the smaller coverage is helpful for surround sound-tracks.

Although it appears to, the blue curtain does not cover a real window (see **Photo 4**). The exterior window was removed and paneled to match the outside wall. The inter-wall space is filled with thermal fiberglass insulation and the inside window

replaced to retain the insulation. The curtain is made of a fairly heavy material but it's not velour, which is often used for sound absorbing curtains and drapes. Overall, the area's absorption is low but not reflective.

Acoustical Room Measurement

Earlier this year, I removed a walk-in closet wall to add an alcove to the listening end. It was a good



Photo 7: There was not much space left on this wall for absorbers so I did what I could, which from my measurements appears to be adequate. I don't know how the irregular edges of the shelves, and the CD and DVD cases affect the acoustics.

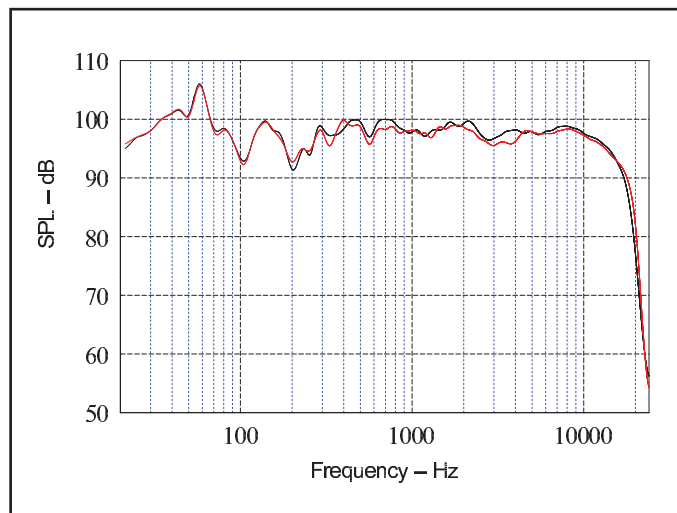


Figure 1: This graph of the average left channel frequency response before and after the addition of the Primacoustic panels shows little difference because of the 2" panels' limited control of the room modes. The black line is before the panel addition and the red line is after.

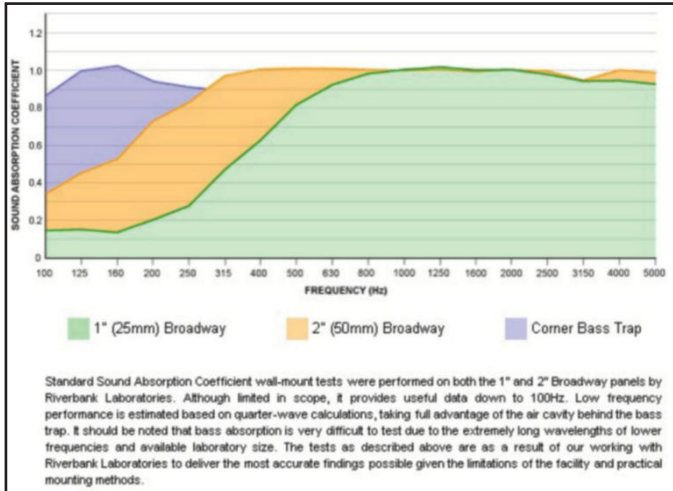


Figure 2: This graph shows the absorption of its 1" and 2" thick Broadway panels and of the 2" panels used as bass traps. (Image courtesy of Primaacoustic)

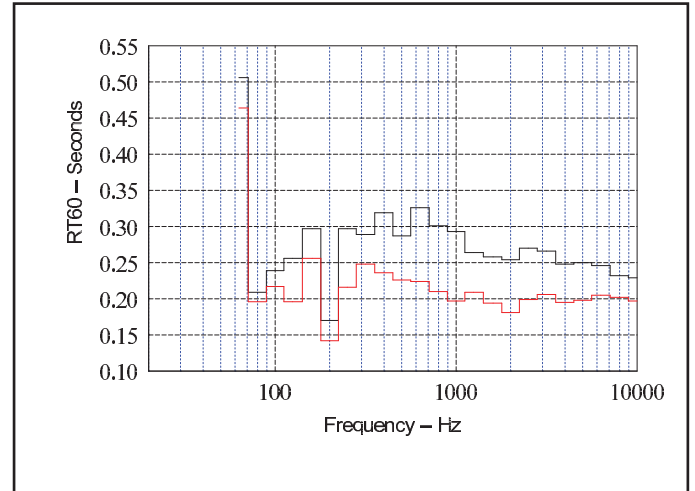


Figure 3: The before- and after-treatment reverberation times (RT60) show an after treatment improvement that correlates with the prediction based on room volume and typical ceiling, wall, and floor absorption. The black line is before the panel addition and the red line is after.

idea because it makes the room seem much more spacious. However, it also voided all my previous acoustic measurements. I carefully measured and recorded the position of my chair and used Room EQ Wizard (REW) to measure six locations

around a 3' circle around my chair as a before-treatment baseline. I repeated the measurements after mounting the panels. REW is an amazingly comprehensive measurement tool, especially so because it's free.

Balanced Drive Die Cast Mid/woofer Transducers

Available in sizes 4", 5", 6.5", 8"

wavecor



Balanced Drive
technology

www.WAVECOR.com

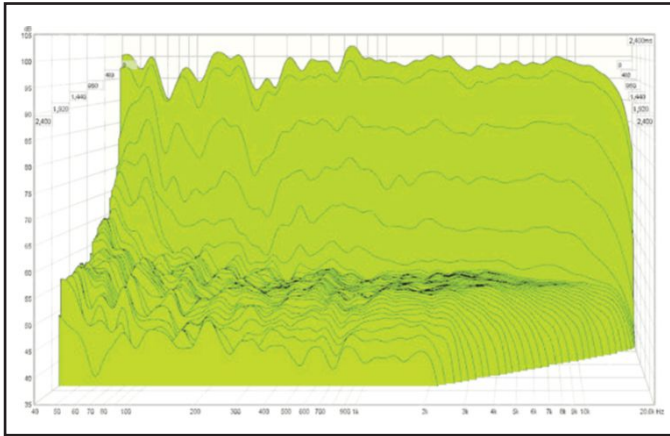


Figure 4: This before-treatment waterfall plot shows a less smooth surface after the initial decay of 35 dB. This adds a bit of harshness to the sound. Sort of like the after-taste of a food or beverage lasting a bit too long.

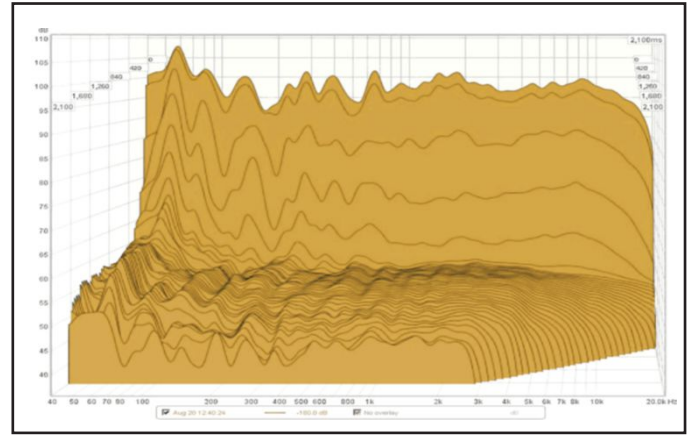


Figure 5: The after-treatment waterfall plot has a smoother surface than that shown in Figure 4 after the initial decay. Also the initial decay is faster and the total time span of the plot is less. This is consistent with the RT60 curves shown in Figure 3.

Frequency Response

As I expected, the before-and-after frequency response curves shown in **Figure 1** are about the same because much of the deviation from flat occurs below 250 Hz. These are the left channel averages of the six measurement positions. But the right channel shows nearly the same thing. The bass traps, using the 2" thick fiberglass, have good absorption down to 100 Hz, but the 2" panels themselves are effective to only about 250 Hz as shown in **Figure 2**. A 3" or thicker bass trap would extend the absorption

to a lower frequency and these are available from Primacoustic as its GoTrap, FullTrap, and MaxTrap, which control bass down to 40 Hz. Of course, being thicker they also take more room space.

I think this quote written by Nyal Mellor, Acoustic Frontiers, and Jeff Hedback, HdAcoustics, in their article, "Acoustical Measurement Standards for Stereo Listening Rooms," explains frequency response in a small room better than I can: "The behavior of a small room in the frequency domain is broken down into a least two parts—above the so called 'transition frequency' and below it. Below the transition frequency the response is dominated by the effect of room modes. Above 250 Hz the response is dominated by the spectrum of the early reflections." (The complete article is included in the Supplementary Material.)

After re-examining the regions above and below 250 Hz shown in **Figure 1**, it may be unrealistic to think the absorbers are going to have a major effect on the room modes. Simply put, too much energy is being bounced about at low frequencies.

Time Response

The Primacoustic panels reveal their merit by how they improve the time domain response. **Figure 3** shows the before-and-after RT60 reverberation times. The before treatment average is 0.28 seconds, with an after treatment average of 0.22 seconds. These numbers were calculated by averaging the left and right times and then taking the average over the frequency range of 63 Hz to 10 kHz.

There is an equation for the exact calculation of the reverberation time based on room volume

Project Files

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

Resource

N. Mellor and J. Hedback, "Acoustical Measurement Standards for Stereo Listening Rooms," HdAcoustics and Acoustic Frontiers, 2011, www.acousticfrontiers.com/wp-content/uploads/2011/10/acoustic_measurement_standards.pdf.

Sources

USB Model UTP200X003MP microscope
AmScope | www.amscope.com

RT60.xls
Bob Golds Home | www.bobgolds.com/RT60/RT60.htm

London 12 Kit
Primacoustic | www.primacoustic.com

REW
Room EQ Wizard | www.roomeqwizard.com

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.

and the area of each absorber multiplied by its absorption coefficient. This is a difficult calculation for a complex room with many reflective surfaces (e.g., the room shown in the photos for this article).

Using Bob Golds' RT60.xls spreadsheet with typical values for the ceiling, wall, and floor absorptions, I get a recommended RT60 of 0.21 seconds. Using his averaged table based on room volume only, the RT60 is 0.17 seconds. The after-treatment value is in the right range and the curve is smoother than before treatment. Also, the spreadsheet gives 0.47 seconds at 63 Hz, which agrees with the measurement.

Waterfall Plot

REW also calculates the waterfall plot from the amplitude and phase measurements. This graphic adds amplitude to the two reverberation time dimensions of time and frequency. That is, it shows time slices of the amplitude decay vs. frequency. A smooth amplitude decay across the frequency band sounds ... well, smoother and more pleasing.

Figure 4 shows the before-treatment waterfall. Here we notice the lower frequencies are decaying

more slowly than higher frequencies. This adds a bit of harshness to the sound. In this case, it's not too noticeable but it could be better. **Figure 5** shows the after-treatment change. After the initial rather swift decay, the "top surface" of the waterfall is more leveled. That is, a smoother surface that sounds smoother.

Final Words

The London 12 Kit made a significant improvement to my room's acoustics. The lower frequencies could be further tamed with a larger bass trap which would take some changes to the rear-wall panel positions. The panels easily lift off the impalers so that aspect would not be a problem. However, Primacoustic has some suggestions if you want the panels to not be easily removed. For more information, visit www.primacoustic.com .

Are you a guitar player

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

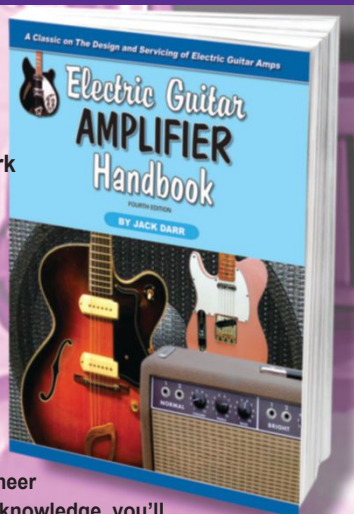
Then this book is for you.

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

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

And much more!

Whether you're an audio engineer or a musician with a thirst for knowledge, you'll find the explanations in this book easy to comprehend as long as you have a general interest in electronics and a passion for DIY.



Get your copy today!

cc-webshop.com

*Ready, willing
and*

AVEL



*offering an extensive range of ready-to-go
toroidal transformers
to please the ear, but won't take you for a ride.*

 **Avel Lindberg Inc.**

47 South End Plaza, New Milford, CT 06776

p: 860.355.4711 / f: 860.354.8597

sales@avellindberg.com • www.avellindberg.com

True Bass in a Small Space

There is a common misconception that in order to get truly deep, clean bass you must have a lot of space for the enclosure. There are three conditions that I think are necessary to achieve “true bass”—wide bandwidth, high dynamic range, and low distortion. In this article, we discover how to achieve true bass in a small area.



Photo 1: Dayton Audio Ultimax 18" subwoofer is purpose-built to move tons of air and create clean, articulate, and fast bass.

By
Thomas Perazella

(United States)

When evaluating the size/bass relationship, the correct situation was summed up by Josef Anton Hofmann whose name represented the “K” in the firm name KLH. There are three interrelated considerations—low bass extension, small box size, and high sensitivity. You can pick any two of the three conditions but will have to sacrifice the other.

This situation was described in detail in my last *audioXpress* speaker article titled “True Bass Rides Again” (April and May 2015). As I approached the end of that project, I became aware of a new driver from Dayton Audio, the Ultimax 18". I mentioned, that it could be a game changer in how we look at low bass reproduction. My guess was that one of those drivers in a medium-sized sealed box could replace two of my 15" drivers in two larger sealed boxes.

Fortunately, I have had the chance to test a kit that Parts Express introduced that contains the Ultimax 18" driver, a 1,200-W switch mode plate amplifier with integral DSP, and an easy-to-assemble sealed box kit to house both. The one I tested from Parts Express came pre-assembled and finished (see **Photo 1**).

The Kit

To make life easy, the kit contains everything you need to implement the new 18" Ultimax into a complete subwoofer. The kit (Part number 300-7200)

contains one Ultimax 18" driver (Part number 295-518), a plate amplifier SPA1200DSP (Part number 300-8000), a modified knock-down sealed subwoofer housing (Part number 300-7079) with cutouts for the Ultimax 18" driver and plate amp with display, and four 1 lb bags of Acousta Stuf Polyfill material (Part number 260-317). The only other things I needed to assemble and complete the kit were glue, clamps, paint, and feet or spikes. This subwoofer is no lightweight with a shipping weight for the assembled unit of 130 lbs. At the time of writing, the price for the kit was \$899.88.

Monster Driver

When I first saw the 18" Ultimax, I thought to myself what a monster. My 15" DVC drivers looked positively small in comparison. Pictures definitely do not do justice to the imposing effect this driver has visually. The 18" cone is a Nomex honeycomb coupled with fiberglass outer skins to produce a very rigid structure that is definitely not run-of-the-mill in appearance (see **Photo 2**).

To me the most impressive specification is the 22 mm Xmax. That, coupled with the large cone, means this driver can move gobs of air in a linear fashion. The frame is cast and two spiders are used to prevent rocking of the cone or voice coil during large excursions (see **Photo 3**).

The driver has two 2 Ω voice coils that can be wired in series or parallel. In this application, they are wired in series for a 4 Ω impedance. The Vas is actually low for a driver of this size at 7.5 ft³ so that a relatively “small” box can be used. However, the sensitivity is stated at 88.6 dB at 2.83 V. Given that the impedance is 4 Ω that actually works out to be 85.6 dB at 1 W. This is a hungry driver. All the specs are available on the Parts Express website (www.parts-express.com).

The Amplifier

Given the driver’s low sensitivity, a substantial amplifier is needed to achieve maximum performance. That amplifier takes the form of the SPA1200DSP. It is a switch mode amplifier for high efficiency with a rating of 1,200 W into the 4 Ω load of the Ultimax 18” driver. That is sufficient power for almost any conceivable musical situation that will be encountered by this subwoofer. It is notable that many low-cost switched mode amplifiers do not reach their rated power output without clipping. When I tested this amplifier, it never failed to produce at least 1,200 W into the driver’s 4 Ω load at any frequency above 15 Hz. Impressive.

All switched mode amplifiers have a level of ultrasonic idle tone at the output. This amplifier had a low 1.8 V level of 360 kHz idle tone, which corresponds to a negligible 0.8 W of power.

Already having several amplifiers that can produce such a prodigious amount of power, that rating did not pique my attention nearly as much as this amplifier’s advanced DSP capabilities. My experiences with several audio products that have DSP have shown what amazingly good results can be achieved by proper application of the controls available through DSP. In this regard, the SPA1200DSP is quite well featured. To get a good overview, you need to look at the block diagram of the amplifier (see **Figure 1**). There are three main functional blocks in the diagram—the signal processing section, the power amplifier section, and the display and manual control section.

DSP at Work

At the heart of the DSP function in this amplifier is an Analog Devices ADSP21479 processor, which is quite a powerful unit for a subwoofer plate amplifier. For Wi-Fi, it uses a Microchip MRF24WG0MA chip that provides 2.4 GHz 802.11 b/g connectivity. Looking at the DSP section, you can see the huge flexibility of signal processing in this amplifier. **Figure 2** shows a flow chart of the signal through the various stages of processing. Signal inputs include a microphone connection for room correction. The amplifier comes

Photo 2: This is a front view of the 18” Ultimax driver.



Photo 3: Here is a side view of the 18” Ultimax driver.

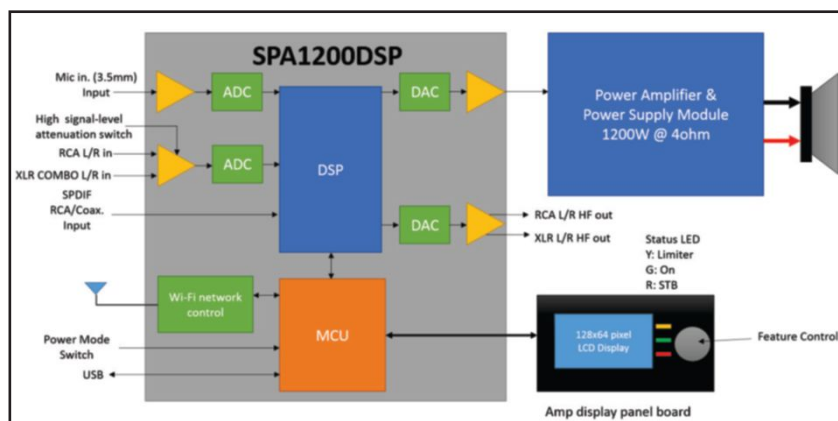


Figure 1: The block diagram of an SPA1200DSP amplifier provides a good overview.

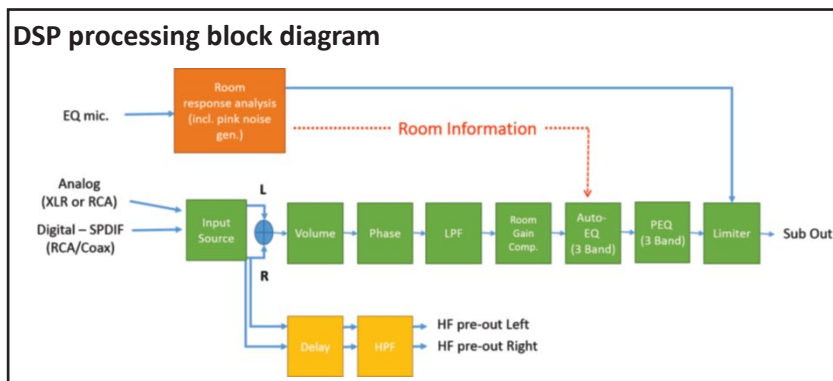


Figure 2: The flow chart shows the SPA1200DSP's signal path as it passes through various stages of processing.

with a microphone and stand adapter. Other analog inputs include both unbalanced RCA jacks and balanced XLR jacks. For analog inputs, a switch is available to select normal level for the unbalanced input or high level for the balanced input. The digital audio input is a coaxial SPDIF connector. There is also a USB connector to connect the amplifier to a PC, enabling the use of the supplied GUI software to effect control of various parameters. For iPhone junkies, there is also a built-in wireless connection that enables you to use your iPhone coupled with

a Dayton Audio SubRemote iPhone Control app to perform the functions. There are general controls for volume, input choice, Wi-Fi configuration, and more. **Table 1** shows the various controls available on the amplifier and their values.

High Pass at Its Best

One function that stood out was the amplifier's control abilities for satellite speakers. In most plate amplifiers, the input analog audio signal, at speaker level, is passed back out to the speakers themselves through a simple 6 dB/octave high pass consisting of a capacitor. If the signal is at line level, it is passed back out to an amplifier that can control the main speakers either unchanged or perhaps through a simple fixed capacitor/resistor 6 dB/octave high-pass filter. Having worked with a lot of subwoofer/satellite speaker combinations, I have come to realize that the greatest benefit of a high-pass filter is the unloading of low frequencies from the satellites.

There is a striking improvement of overall sound when the demands of low-frequency reproduction are removed from the main speakers. Most main speakers do not have sufficient linear volume displacement to handle even moderate levels of very low-frequency content. The result is that the distortion in the bass or bass/mid drivers dramatically increases. To get the most improvement, the speakers must be high passed at frequencies and crossover slopes that prevent excursions of the drivers outside of their linear operating areas. A 6 dB/octave crossover slope will generally not cut it. With that slope, excursion continues to increase even below the crossover point. At least 12 dB/octave is needed to prevent increased excursion below the crossover point.

With this amplifier, the input signal is digitized and fed to the DSP section. Within that section, one of the functions is control of the high-pass parameters that will control the main speakers. Slope and crossover point as well as time delay can be controlled. For all the various controls, you can feed the full frequency spectrum to the mains, but I cannot see a reason to pass the unmodified signal through the amplifier for that condition. You can then pick the parameters that work best for your main speakers, including slopes of 12 dB/octave and above. This is a powerful feature and will greatly help in achieving the best melding of the subwoofer with the mains and reduction of intermodulation distortion in the mains.

Low-Frequency Controls

Gain of the woofer amplifier is set on the main control panel in 1 dB increments. The low-frequency

SPA1200DSP Values		
Level 1 Function	Level 2 Function or Value	Level 3 Function or Value
Volume	0 to -100 dB in 1 dB steps	
Low-pass filter	On/Off	On/Off
	Frequency	31, 40, 50, 63, 80, 100, 125 Hz
	Slope	12 dB, 24 dB/octave
High-pass filter	On/Off	On/Off
	Frequency	31, 40, 50, 63, 80, 100, 125 Hz
	Slope	12 dB, 24 dB/octave
Phase degree	0 to 180 in 1 degree steps	
High-pass delay	0 to 10 ms in 0.1 ms steps	
Room gain	On/Off	On/Off
	Frequency	25, 31, 40 Hz
	Slope	-6, -12 dB/octave
Subwoofer Tune	Sealed, 16 Hz, 25 Hz	
PEQ—all 5 bands	Disable	On/Off
	Frequency	20 to 200 Hz in 1 Hz steps
	Level	-12 to 6 dB in 0.1 dB steps
	Q	0.4 to 11 in 0.1 steps
Limiter	Threshold	-12 to 0 dB in 0.1 dB steps
	Attack	0.1 to 100 ms in 0.1 ms steps
	Release time	5 to 5000 ms in 0.1 ms steps
Input Source	Analog, Digital	

Table 1: Various controls are available on the SPA1200DSP amplifier.

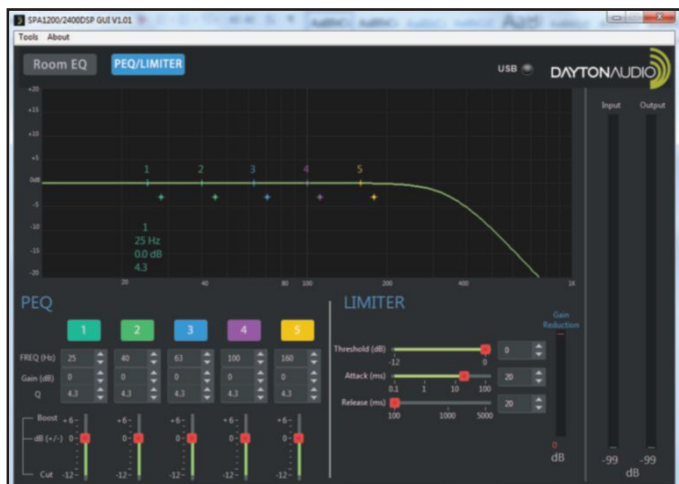


Figure 3: The parametric EQ screen shows the result with no EQ applied.

rollover frequency and slope can be set for the woofer. Phase can be controlled in both directions. An unusual setting is an adjustment for room gain. You can pick a frequency and slope for the adjustment and then add attenuation. A feature with an unusual name is the "Subwoofer Tune." In reality, it is a selectable cutoff filter with three choices. The first is called "Sealed" where the signal is passed unattenuated to the woofer. There are two other selections for cutoffs at 16 Hz and 25 Hz. The slopes are not specified, but this feature is designed to protect woofers from excessive excursions where they are used in vented boxes and would become unloaded below the port resonance frequency.

Parametric EQ

An extremely valuable feature of the DSP is three channels of parametric EQ. My first experience with parametric EQ goes way back to a pair of Orban 672A equalizers that I used to tame some speaker and room resonances. Having the flexibility of parametric EQ was a real eye and ear opener for me. Since then, I have used digital parametric EQ extensively in all my main speaker systems.

The three channels of parametric EQ are accessible through the

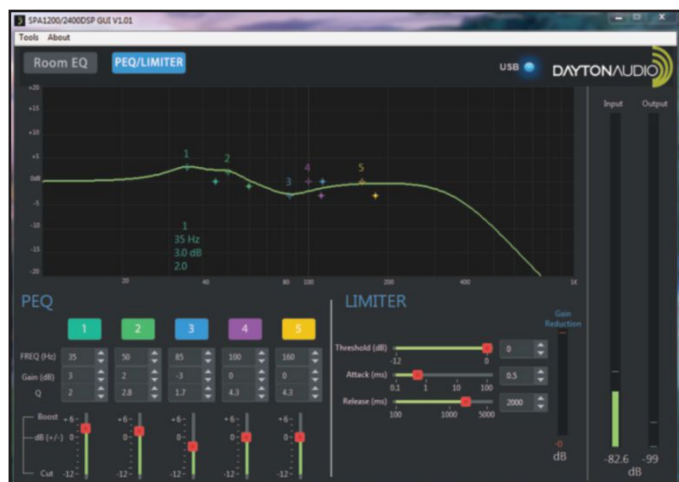


Figure 4: Parametric EQ is shown after I made some manual adjustments.

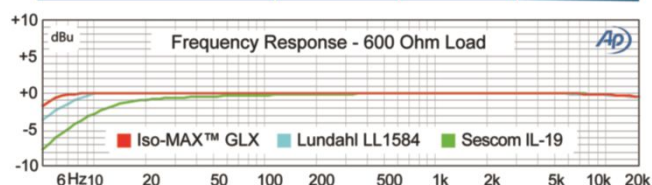
Iso-Max® GLX isolator... dwarfs the others!

When it comes to performance, build quality, warranty and value, the Jensen Iso-Max GLX™ is in a class of it's own!

Jensen Iso-Max... noise elimination done right!



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



Jensen Transformers Inc.
9304 Deering Avenue
Chatsworth, CA 91311
Phone: (818) 374-5857

©2016 Jensen Transformers Inc. All rights reserved.
Specifications and appearance subject to change without notice.
www.jensen-transformers.com

jensenTM
transformers

PRIMACOUSTIC MAKE MAN CAVE SOUND GOOD!



Audiophile
REVIEW.COM

"...spend all the money you want on audiophile electronics but your room acoustics are the single most important factor in how good your audiophile system will sound."



London 10™
room kit - \$499.99

PRIMACOUSTIC
www.primacoustic.com

A division of Radial Engineering Ltd. - 1588 Kebet Way, Port Coquitlam BC V3C 5M5
©2016 Primacoustic. All rights reserved. Specifications and appearance subject to change without notice.



Photo 4: This is my in-room measurement equipment.



Figure 5: The Auto Room EQ screen shows the original and adjusted curves.

GUI and your PC or the iPhone app. I don't have an iPhone so I will only show the PC version. **Figure 3** shows the screen with no EQ applied. With the parametric EQ you can control the center frequency, the gain, and the Q or bandwidth. Values can be directly input into the appropriate fields or arrows can be used to increment or decrement the values. **Figure 4** shows the additional manual adjustments I made in the room after the basic auto room EQ had been done.

Automatic Room EQ

Every room has some nodes that either reinforce or cancel certain frequencies. To some extent, especially with very low frequencies these can be adjusted to improve performance. The SPA1200DSP is supplied with a microphone and a holder so that in-room measurements can be made and then problems ameliorated. The process entails selecting the Room EQ button in the GUI, placing the microphone at the first of three locations and then selecting the Start Position 1 button. Measurements will be made and then you will be asked to move the microphone to position 2 where another set of measurements will be made. This is repeated at a third position of your choice, and all three positions averaged to come up with a final EQ curve.

Table 2: I compared the linear volume of the Ultimix 18 and the DVC 15.

Driver Displacement Calculations						
Driver	Area	Xmax mm	Total Linear Excursion	Individual Linear Displacement Liters	Number of Drivers	Total Displacement Liters
Dayton DVC385-88 DVC 15"	830 cm ²	15.0 mm	3.00	2.5 ltr	2	5.0 ltr
Dayton UM18-22 Ultimix 18"	1,213 cm ²	22.0 mm	4.40	5.3 ltr	1	5.3 ltr

Photo 4 shows all the measurement equipment I used for both the Auto Room EQ and distortion measurements done with the program LAUD from Liberty Audio. You can see the small microphone on the left arm of my listening couch, which was the first position. In the center of the couch is the ACO Pacific 7012 measurement microphone I use to feed LAUD. Both the pre and post measurement curves are displayed in **Figure 5**. The pre EQ curve is shown in grey and the post EQ curve with corrections is shown in blue. Below the curves are the EQ values that were used to modify the response.

Measurements

Normally, the first measurement I make on a subwoofer is a closed mic'd frequency response. To do that LAUD needs to work with an amplifier where the signal and the speaker leads on the amplifier are both at ground potential. Laud measures the level of each test frequency from the amplifier and compares it to the microphone output of SPL. That way if the amplifier has any frequency fluctuations, they are removed from the speaker measurement.

This amplifier has a bridged output so the negative speaker lead is not at ground potential, preventing me from doing true frequency response measurements. However, I have not found any

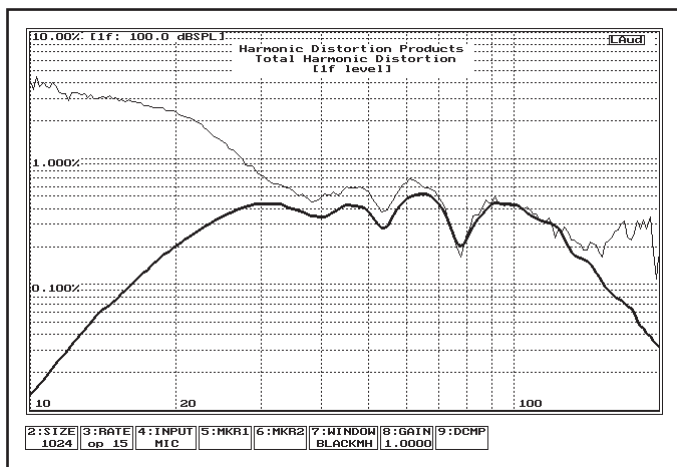


Figure 6: This is the measured Ultimax 18" close microphone distortion.

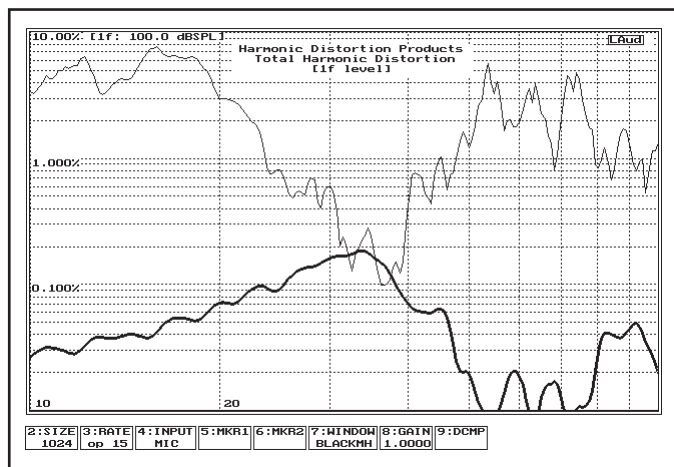


Figure 7: Here is the measured Ultimax 18" listening position distortion.

plate amplifiers that had any significant frequency anomalies that would prevent an assumption of flat output response. During my tests, the amplifier had no low-frequency cuts programmed into the DSP.

The close mic'd harmonic distortion measurement started the series. At the end of my last article, "True Bass Rides Again," I postulated, based on mathematical considerations, that one Ultimax 18 should be able to replace two of my DVC 15s. **Table 2** shows a comparison of the linear volume displacement. The calculations show that the linear volume displacement of one Ultimax 18 is slightly higher than two DVC 15 woofers.

The Ultimax 18's closed microphone distortion was low. In fact, it was the lowest close microphone distortion of any subwoofer I have ever measured. **Figure 6** shows the distortion measurements. As you can see, at 100 dB with the microphone close to the cone, the distortion at 20 Hz was just slightly over 2% and never went above 0.7% from 30 to 200 Hz. This driver has excellent linearity.

Although I do not have an anechoic chamber, I decided to do some distortion measurements at my listening position, which is 14' from the subwoofers at a level of 100 dB. Keep in mind that measurements in a normal room like I have are fraught with problems due to nodes, especially at frequencies over about 45 Hz. The distortion measurements are a percent of the fundamental. Suck outs at a particular fundamental frequency can result in the distortion products registering higher as a percent at that frequency.

If there is a bump at a particular fundamental being measured, the distortion products will seem to be lower as a percentage of that frequency. In

the distortion measurements taken of both the Ultimax and DVC drivers at the listening position, the percent distortion is the upper lighter trace and the fundamental level is the lower heavier trace. You can see the effect I mentioned as the fundamental

A New Part Introduction for 2016

1.8 nV Low Noise Dual P-Channel JFET 4pf Low Capacitance Complement to N-Channel LSK489



LSJ689 Series



- Low Noise < 1.8nV
- Low Capacitance: 4pf max
- Low Offset: 20mV max
- Complement to Dual N-Channel JFET - LSK489
- Ideal for Differential Amplifier Applications when uses with the LSK489
- Ideal for Voltage Controlled Resistor Applications
- TO71, SOIC-8, SOT23-6, ROHS Packages
- Contact the Factory or Visit the LIS Website for Data Sheets/Pricing/Samples

www.linearsystems.com
1-800-359-4023

Exhibiting in Arena, Booth#7710

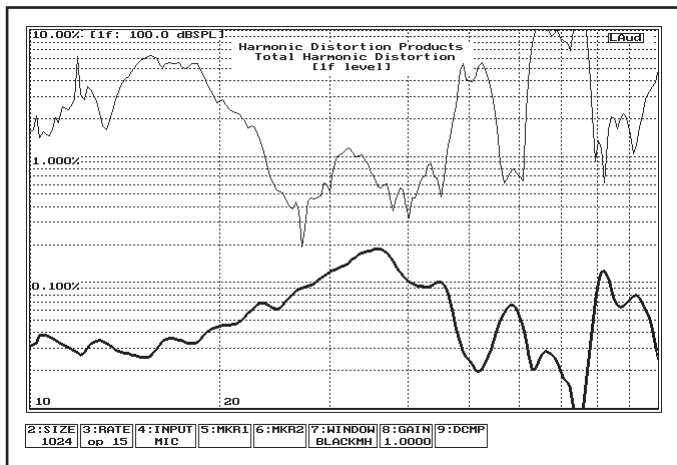


Figure 8: I also measured the DVC 15" listening position distortion.



Photo 5: I placed the Ultimax 18" next to an LP for size comparison.

goes up and down in level. As the fundamental goes down, the distortion percent goes up and vice versa. Although you will have to do a little "reading between the lines," the results are still valuable in showing the very low distortion of both solutions especially at low frequencies, which would wreak havoc with most smaller drivers.

The Ultimax distortion at the listening position never exceeds 7% below 20 Hz and drops to about 0.5% by 30 Hz. For measurements at 100 dB and 14' from the driver, this is great performance (see **Figure 7**). The two DVC 15s show similar results as expected with distortion below 20 Hz never exceeding 6% and dropping to around 0.5% at 30 Hz (see **Figure 8**). These measurements were made with no EQ being applied.

Just for grins, I decided to repeat the measurements with EQ in place. In the case of the

Ultimax, the SPA1200DSP amplifier was limited to 20 Hz for corrections. In my system, I use the DEQX HDP-Express II preamplifier/processor for crossover and EQ functions. It has a lower correction limit of 10 Hz. When repeated, the distortion graphs show much flatter fundamentals with the corrections provided by the SPA1200DSP not quite reaching 10 Hz, but the DEQX reached all the way down to the 10 Hz limit resulting in flatter response. At 10 Hz, I'm not sure anyone could tell the difference, but it is nice to have that capability. To put that in perspective, the cost of the DEQX alone is \$3,495, which is more than four times the cost of the complete Ultimax kit. Yes, as a preamplifier/processor the DEQX is phenomenal, but it shows the value of all the aspects of the Ultimax kit.

Figure 9 shows the distortion results of the Ultimax with the room EQ function of its amplifier.

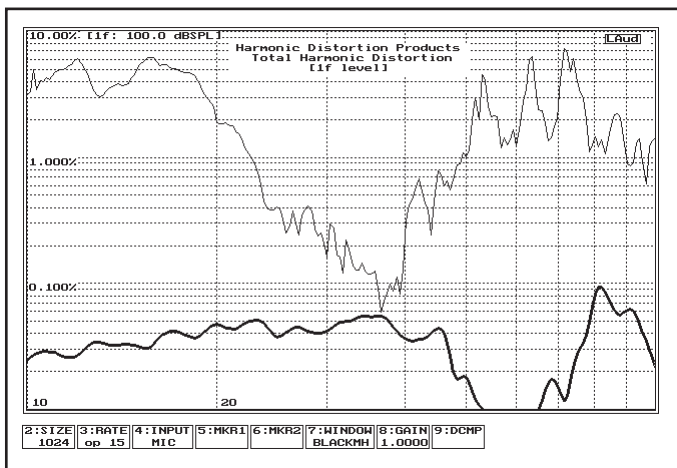


Figure 9: This is the measured Ultimax 18" listening position distortion with EQ.

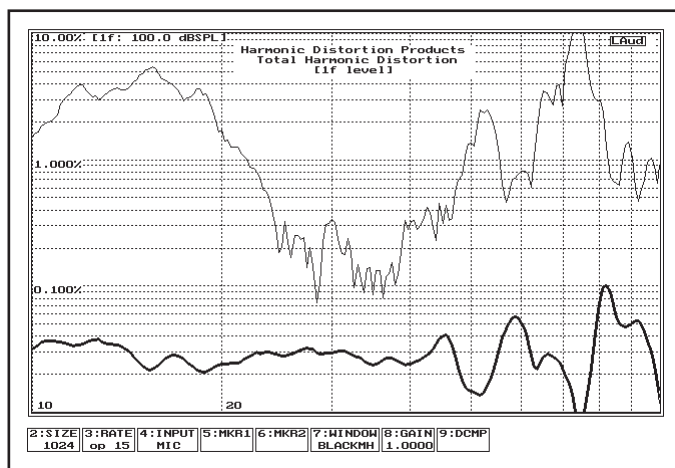


Figure 10: This is the DVC 15" listening position distortion with EQ.

Although the distortion measurements are similar, you can see the fundamentals are much flatter. **Figure 10** shows the distortion results of the two DVC woofers with the DEQX correction. The fundamentals are impressively flat from 10 Hz to around 45 Hz.

Size Comparison

The primary goal of this exercise was to see if a substantial size reduction of the subwoofer could be achieved with the same acoustic performance. **Photo 5** shows the Ultimax subwoofer next to an LP for size comparison. Even though the subwoofer is substantial, you can see that the majority of the front surface area is taken up by the massive driver. In reality, the total volume is actually small in relation to the prodigious bass output.

For comparison, I also placed the Ultimax next to two of the DVC subwoofers I currently use. **Photo 6** shows the size comparison. As you can see, there is a tremendous difference in size between the two DVC subwoofers and the Ultimax. **Photo 7** shows the Ultimax in position behind my mid bass and mid/high speaker sections. Compared to them, the subwoofer virtually disappears. If the outputs are similar in level, distortion, and character, from a size standpoint, using the Ultimax would result in a significant reduction in room space utilization and visual impact.

Measured distortion levels at the same output already confirmed the similarities in the single Ultimax compared to the two DVCs. The final test was to actual listen to some music to determine the character of the different approaches.

Listening

Selecting appropriate source material for testing truly high performing subwoofers can be tricky. There are a lot of sources claiming to contain killer bass that are actually quite anemic. In addition, there are a lot of rumors about how low certain tracks go. Many of those are inaccurate. I have a few selections that I have found contain very low bass and at high amplitudes. I chose four main selections—"Bass I Love You" from Bass Mekanik, "Helicopter" from the Revue du Son test disk, "Saint-Seans Organ Symphony" from the Boston Audio Society test disk, and "Shuttle Launch" recorded by Bob Katz using DPA mikes supplied by a friend, Mike Morgan.

It is important to know this last cut that I received from Morgan is the original two-channel feed. The cut available from Katz's site is a multi-channel mix using other mikes in addition to the DPA microphones. They sound slightly different but both are exceptional.



Photo 6: I placed the Ultimax 18" subwoofer next to equivalent DVC subwoofers for comparison.

These pieces are not suitable for normal speakers as at best the low frequencies will not be reproduced and at worst they can destroy woofers. If you plan to use any of the above samples do so with great caution. To make sure they really contain low bass, I used the application Audicity to determine the true low-frequency content.

I fed the cuts into Audicity and placed time markers at the beginning and end of the lowest



Photo 7: For the listening test, I placed the Ultimax behind my mid bass and mid/high speaker sections.



Bass Frequency Determination							
Music	Times	Segment 1	Segment 2	Segment 3	Segment 4	Segment 5	Average
Bass Mekanik	Finish	12.518	20.94	29.36	37.78	71.465	
"Bass I Love You"	Start	12.368	20.79	29.21	37.63	71.315	
	Duration	0.15	0.15	0.15	0.15	0.15	
	Frequency	6.66666667	6.66666667	6.66666667	6.66666667	6.66666667	6.666667
Revue du Son	Finish	59.649	59.853	60.005	60.161	60.313	
"Helicopter"	Start	59.597	59.802	59.954	60.11	60.263	
	Duration	0.052	0.051	0.051	0.051	0.05	
	Frequency	19.2307692	19.6078431	19.6078431	19.6078431	20	19.61086
Boston Audio Society	Finish	28.9	29.072	29.772	113.838	116.458	
Saint-Saens Organ Symphony	Start	28.841	29.015	29.713	113.78	116.399	
	Duration	0.059	0.057	0.059	0.058	0.059	
	Frequency	16.9491525	17.5438596	16.9491525	17.2413793	16.9491525	17.12654
Mike Morgan	Finish	28.22	29.828	30.29	31.923	32.649	
Shuttle launch	Start	28.097	29.689	30.173	31.809	32.51	
	Duration	0.123	0.139	0.117	0.114	0.139	
	Frequency	8.1300813	7.1942446	8.54700855	8.77192982	7.1942446	7.967502

Table 3: Here are the results from my music frequency tests.

frequency waveforms. For each piece, five samples were taken and then put into a spreadsheet so that frequencies could be calculated for each sample and then an average of the five could be determined. **Table 3** shows the results. With frequencies from 6.7 Hz to 19.6 Hz and with high levels, these pieces will quickly reveal any limitations in subs under test.

In addition to the low bass tests, I used many other musical pieces to determine overall bass performance and integration with the mid and high sections of my speakers. The music choices included various organ recordings, jazz containing string bass, kick bass, piano, rock, and other bass rich pieces. For a list of some of those selections, refer to Table 2 in the second part of my article "True Bass Rides Again," (*audioXpress*, May 2015). During all the tests, the Ultimax was on the left side of my system and the two stacked DVCs were on the right side.

Listening tests were performed by myself and two other listeners—one male and one female. The tests consisted of two sections, the first using only one side of the subwoofers at a time with the mid and treble sections of both speakers to compare

the Ultimax to the DVCs. In all tests, the associated equipment used the settings that resulted in the flattest response at the listening position. The second part of the test was to run the complete system with both subwoofer sections to determine the overall balance of the system in actual listening conditions.

The results were the same with all listeners in all tests. The subwoofers sounded the same regardless of the selections played. When both sections were used the overall balance and impact of the subwoofers was spot on. Since the actual characteristics of the DVC system was extensively covered in my May 2015 *audioXpress* article, I will not repeat the detailed results for each test piece here. The main goal was to determine if the single Ultimax 18" driver in a smaller box could replace a pair of DVC 15" drivers in individually larger boxes. In everyone's opinion, the answer is yes. This was very exciting because it confirmed two different approaches could provide the same acoustical results.

The Bottom Line

Are there any nits to pick? Yes, but they are quite small. The GUI that enables computer control of the amplifier's DSP only works with two functions—the auto room EQ and the parametric EQ. The other functions have to be addressed at the control panel, which is not as convenient. A temporary memory storage location for a setup in progress would be convenient for quick comparisons during changes to

About the Author

Thomas Perazella is a retired IT director. He is a member of the Audio Engineering Society, the Boston Audio Society, and the DC Audio DIY group. He has authored several articles in professional audio journals.

the parameters. You have unlimited setups available but they are stored in files that have to be recalled to replace the current setup in memory.

In addition, the instructions are limited and an expansion would be appreciated. I am one of those rare types who reads all the instructions provided with equipment and find that most could use significant improvement. Parts Express has taken these and other suggestions to heart and it is working on upgrades.

The bottom line is that Hofmann's Iron Law holds true. If you want deep bass in a small enclosure, you have to give up sensitivity. However, if you have a clean high-powered amplifier, you can achieve exceptional low bass in a relatively small enclosure.

This Parts Express kit provides a great driver with a powerful, flexible amplifier and a sturdy enclosure kit resulting in a solid bass performance that would cost you several times more to achieve in a commercial solution. Another advantage of the kit is you can finish it to your liking. So if you are a DIYer with a hankering for true bass, you should give serious consideration to this kit. It will be hard to beat. For more information, visit Parts Express at www.parts-express.com 

Resources

V. Ho, "J.A. Hofmann, 86, entrepreneur, Army researcher," *The Boston Globe*, January 2011, http://archive.boston.com/bostonglobe/obituaries/articles/2011/01/11/ja_hofmann_86_entrepreneur_army_researcher/?page=2.

T. Perazella, "True Bass Rides Again (Part 2): The Power Question," *audioXpress*, May 2015.

——, "True Bass Rides Again (Part 1): Reproducing That Original Sound," *audioXpress*, April 2015.

Sources

7012 Measurement microphone

ACO Pacific | <http://acopacific.com>

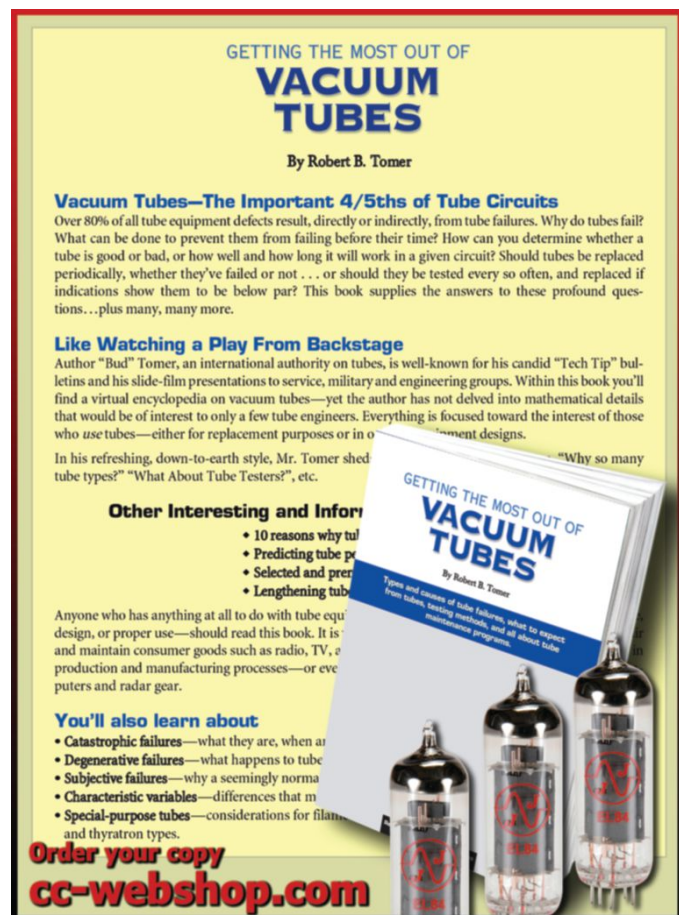
LAUD

Liberty Audio | <http://libertyaudio.com>

Ultimax 18" Kit

Parts Express | www.parts-express.com/dayton-audio-18-ultimax-subwoofer-with-spa1200dsp-amplifier-and-cabinet-package--300-7200

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.



Room Acoustics and Other Bookcase Necessities



As wonderful as the Internet is, professional acousticians still rely on a well-stocked bookcase to provide references on the theory and the data needed to practice their profession. In this article, we discuss one of the must-have books.

By
Richard Honeycutt
(United States)

The stocking of the bookcase begins during undergraduate school and continues throughout graduate school into professional practice. The first books to be acquired are often textbooks, and the familiarity of these textbooks makes them convenient references. Textbooks are often supplemented by handbooks covering specific areas of acoustics in greater detail than a textbook can provide.

A few textbooks have become classics because of their clarity, topical coverage, technical merit, and organization. One such textbook is *Room Acoustics* by Heinrich Kuttruff, first published in 1973. The sixth edition of this comprehensive textbook has just been released by CRC Press (see **Photo 1**).

Kuttruff's Background

Kuttruff is Emeritus Professor and former head of the Institute of Technical Acoustics at Aachen University in Aachen, Germany. Born in 1930 in Engen in Konstanz, Kuttruff is a physicist and expert on acoustics. Kuttruff studied physics at the University of Freiburg and the University of Göttingen and received his doctorate in 1958. In 1968, he became adjunct professor at Göttingen and in 1969 Professor of Electroacoustics at the TH Darmstadt. Beginning in 1972, he was Professor of Technical Acoustics and Director of the Institute of Technical Acoustics at RWTH Aachen University. From 1972 to 1995, he was co-editor

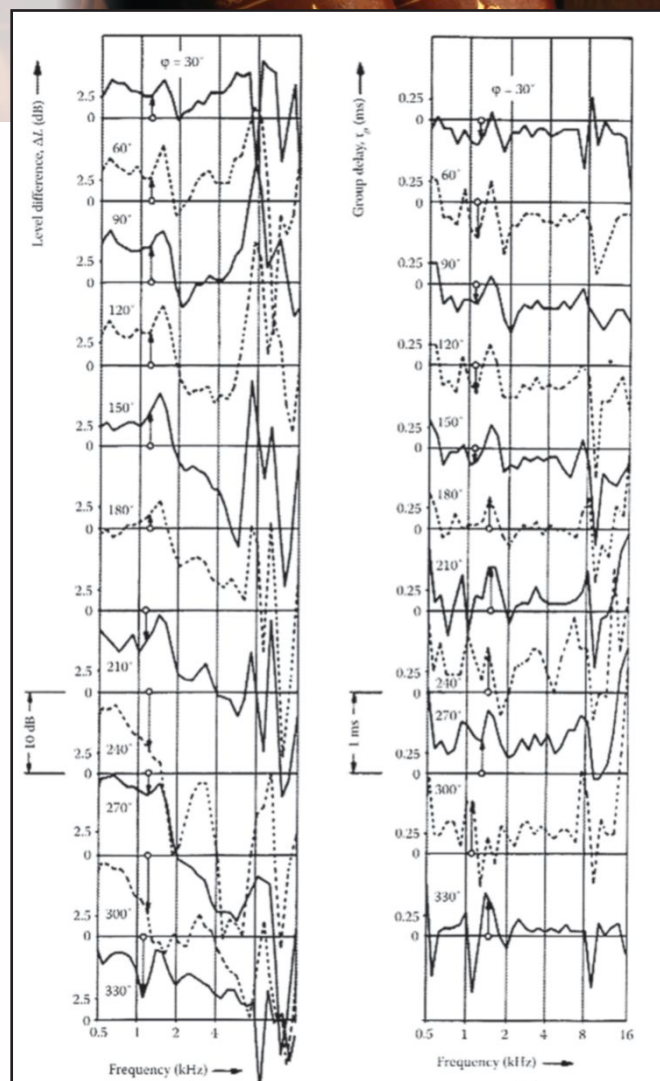


Figure 1: Heinrich Kuttruff presents frequency-response graphs of the head-related transfer function (HRTF) for several directions of sound incidence.

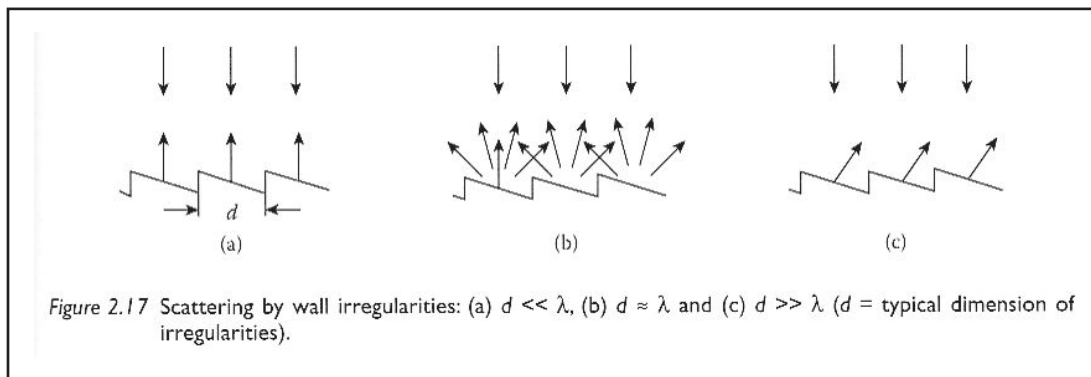


Figure 2: Kuttruff depicts scattering by irregularities in a wall.

of *Acustica* and 1975 to 1995, of *Applied Acoustics*. In 1995, he became professor emeritus.

In 1988 he received the Silver Medal of the French Society for Acoustics. He was a corresponding member of the 1989 Göttingen Academy of Sciences. In 1996, he received the Helmholtz Medal of the German Society of Acoustics; and in 2005, the Rayleigh Medal from the Institute of Acoustics. In 2003, he became an Honorary Fellow of the Institute of Acoustics.

Room Acoustics covers the field of architectural acoustics in depth and breadth. I have listed the major Table of Contents headings and included some comments on the corresponding chapters.

Introduction

Some facts on sound waves, sources and hearing. One indication of this book's technical rigor is that discussion of the acoustical wave equation begins on page 2. After this introductory discussion, other important ideas in the physics of sound and psychoacoustics are presented. Although the mechanism of directional sound perception is discussed in many books, and the head-related transfer function (HRTF) is defined, Kuttruff's is the first textbook that has come my way that illustrates what HRTF's look like (see **Figure 1**).

Reflection and Scattering

Virtually all of the "standard" or older acoustics texts discuss acoustical reflection and absorption. Many of them also address edge diffraction. But Kuttruff is one of the few texts that covers scattering. Although all computerized acoustical modeling programs include an algorithm for scattering, the subject is not well-understood by many people in the acoustical community.

In addition to simple scattering from rough surfaces (see **Figure 2**), Kuttruff also discusses Schroeder phase grating and quadratic residue diffusers, which are two of the most common

methods to introduce scattering in a predictable, controlled way.

Sound Waves in a Room

Kuttruff begins this chapter by discussing geometrical vs. wave approaches to acoustical analysis. He includes the equations for calculating the frequencies of the room modes for a rectangular room, pointing out that such a room is seldom found in reality, since furniture and discontinuities in wall



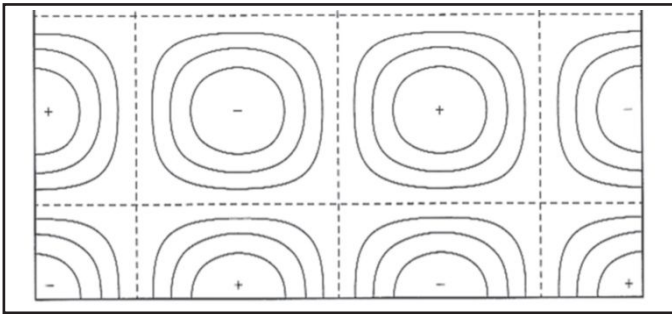


Figure 3: Kuttruff illustrates modal sound pressure distribution in a rectangular room.

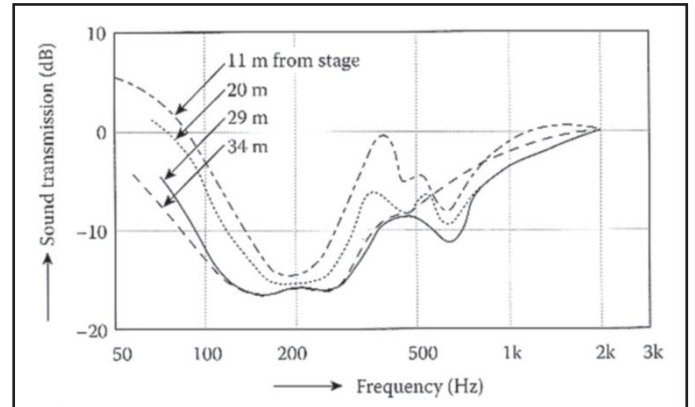


Figure 4: Kuttruff illustrates the seat dip effect, measured at unoccupied seats in the Boston Symphony Hall.

materials prevent real rooms from matching the assumptions included in deriving the equations.

Figure 3 shows an example of the sound pressure contours for one mode in one specific room. Also discussed are the number and density of modes, and the finite-element method of numerically solving the wave equation. Room transfer functions, modal decay, and reverberation round out the topics covered in this chapter. The immense practicality of Kuttruff's approach in our age of computerized acoustical analysis is readily seen if you compare his treatment of room acoustics with the predominantly statistical one appearing in most earlier textbooks.

Geometrical Room Acoustics

A thorough understanding of geometrical acoustics is essential to evaluating a venue for echoes and focusing, and for understanding how most computerized acoustical modeling software works. Kuttruff's coverage is enhanced by references to recent research in various elements of geometrical acoustics.

Reverberation and Steady-State Energy Density

This chapter includes aspects that are not covered in most other textbooks, such as the effects of non-uniform boundary absorption, unequal sound

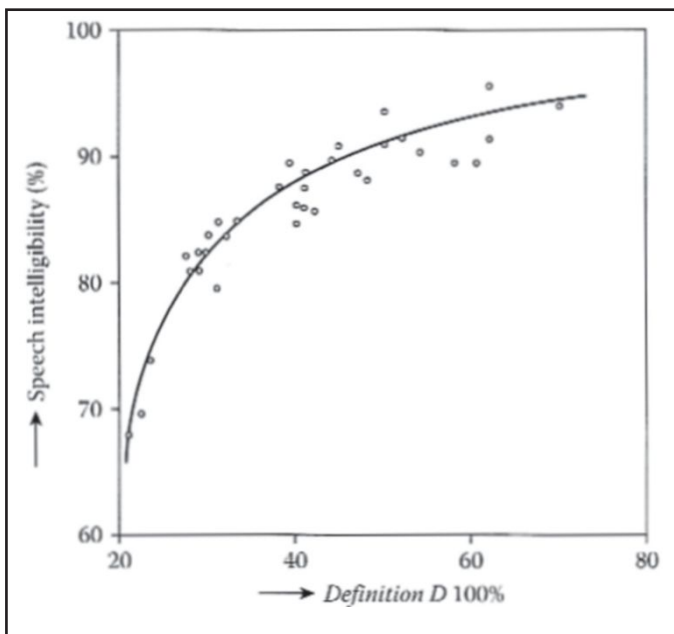


Figure 5: Kuttruff depicts the relationship between speech intelligibility and definition.

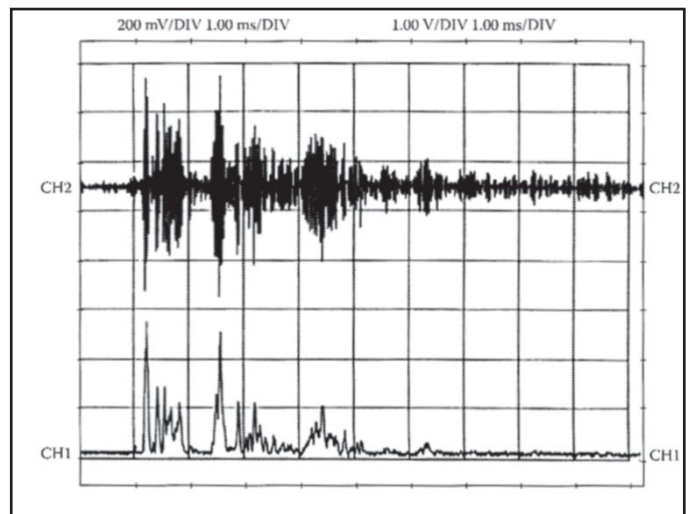


Figure 6: Kuttruff shows a sample room impulse response and its squared envelope.

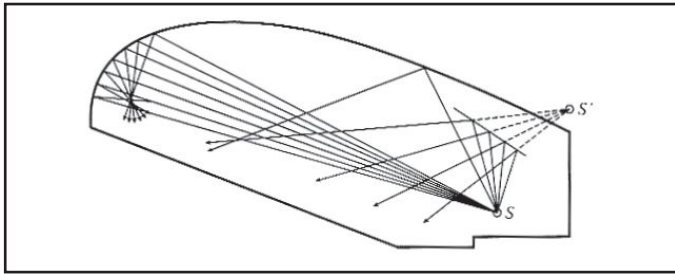


Figure 7: Kuttruff illustrates the effect of room shape upon direct sound arrival at the seats.

path lengths, and imperfect diffuseness. Prior to the work of Trevor Cox and Peter D'Antonio from the late 1970s forward, sound diffusion was usually addressed only in a qualitative fashion. Kuttruff's coverage updates the discussion so as to enable the reader to understand quantitative approaches to the use of diffusion to eliminate acoustical defects.

Sound Absorption and Sound Absorbers

This chapter discusses the mechanisms of acoustical absorption: non-adiabatic effects, viscous effects, thermal relaxation, and diaphragmatic absorption. The application of this understanding to resonant and non-resonant absorbers is described in the form of examples, but not at the level of detail represented by the extensive absorption coefficient tables found in many other texts. Kuttruff provides the best discussion of audience seat-dip effect (see **Figure 4**) that I have seen in a textbook.

Subjective Room Acoustics

Subjective effects of room acoustical parameters have been studied in depth by a number of researchers throughout the last three-quarters of a century. As a result, we have many new metrics with which to describe a venue's acoustical performance. These include echo perception, clarity index, definition, center time, Speech Transmission Index (STI), stage support, source strength, apparent source width, and Inter-Aural Correlation Coefficient (IACC). The increase in number of metrics has not necessarily

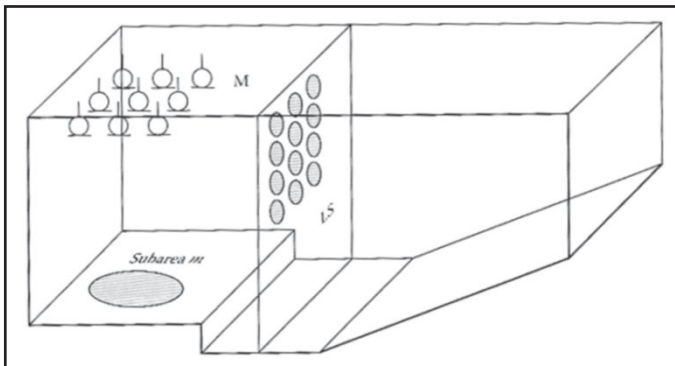


Figure 8: Kuttruff illustrates the principle of wavefront synthesis, using an array of microphones above the performers, feeding a processed signal to an array of speakers.

Sign up now for The Audio Voice FREE weekly e-newsletter!

Check out our innovative content and be the first to know what's happening in the audio industry.

Read full articles online from audioXpress and Voice Coil authors and learn what's new.

One click is all it takes to have this e-newsletter delivered to your inbox.



audioxpress.com





Resources

Acoustical Society of America, www.abdi-ecommerce10.com/asa.

H. Kuttruff, *Room Acoustics*, 6th Edition, CRC Press, October 13, 2016.

been accompanied by a corresponding increase in accessible reference material that would facilitate an acoustician's designing a venue to optimize these performance areas, or help professionals to understand the interrelations among the new metrics.

Kuttruff's chapter on subjective room acoustics remedies this long-standing omission in the literature. **Figure 5** illustrates the way in which definition correlates with speech intelligibility or STI.

Measuring Techniques in Room Acoustics

Not all acoustics textbook authors address measurement methods, but Kuttruff does. He begins by examining the measurement, examination, and analysis of impulse response (see **Figure 6**). Then he discusses measurements of directional distribution and diffuseness of the sound, and sound absorption (both the impedance tube and reverberation chamber methods). He concludes the chapter by describing the measurement of scattering coefficient.

Design Considerations and Design Procedures

The design of auditoria and concert halls is usually performed by a team including one or more acoustical consultants and architects. As Kuttruff describes, the design includes multiple factors: noise, room shape (see **Figure 7**), and the room finishes that control reverberation time.

The use of variable absorption in multipurpose halls is discussed, as are the methods of checking a design by scale modeling and computer simulation, including auralizations (virtual reality).

Electroacoustic Systems

While this book is not intended to train the reader to be a sound-system designer, it does recognize the need for today's acoustician to be conversant with the basic design elements of simple direct-radiator, horn, and line-array loudspeakers. With this in mind, the operating principles of these types of speakers are presented in a clear, non-mathematical fashion, and the basic principles of speaker placement are laid out.

In addition, the somewhat more abstruse techniques of wavefront synthesis (see **Figure 8**), and electronic reverberation enhancement are introduced.

Summary

While no textbook on acoustics stands alone as a complete library, Kuttruff's *Room Acoustics*, 6th edition does a fine job of presenting the basic elements of acoustics, particularly architectural acoustics. It goes well beyond most of its predecessors in addressing topics that have entered the acoustical world since the 1980s.

Where mathematics are needed to present a complete understanding of principles, mathematical treatment is used, but only in service of understanding the science and art of acoustics. The level of presentation is appropriate for the serious student whose education includes undergraduate-level calculus-based physics.

Even for those with less math background, much of the meat of the text is accessible. I have no hesitation in recommending it as a basis for your personal technical library in acoustics.

Other Books Worth Consideration

Additional books that you may want for your acoustics library could include specialized ones for your particular areas of interest, such as architectural acoustics, speaker design, sound-system design, industrial and community noise prevention, and the like, as well as handbooks on acoustical measurements and the other subjects just mentioned.

(A handbook tends to include mainly practical "how-to" information and data, with less emphasis on the physics behind the topics discussed.) And for a really comprehensive library, you may want some of the classic books by Vern Knudsen, Harry Olson, Leo Beranek, and Alton Everest. These books can sometimes be found on Internet sites, and a few of them have been reprinted by the Acoustical Society of America. 

This column is sponsored by: **Acoustic Sciences Corp.**

ASC was created in 1984 by Art Noxon to supply his revolutionary invention, the TubeTrap, for distribution by Monster Cable. Over the years, ASC evolved into a premier specialty acoustic product design and manufacturing operation. All products are handmade in the US. As the world's first factory-built corner-loaded bass trap/treble diffuser, the TubeTrap became one of the iconic founding products of what would become today's acoustic products industry.

acousticssciences.com | 800-ASC-TUBE (272-8823) | info@acousticssciences.com

The "Must Have" reference for
loudspeaker engineering professionals.

Home, Car, or Home Theater!

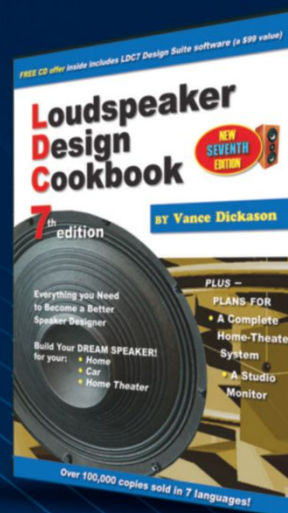
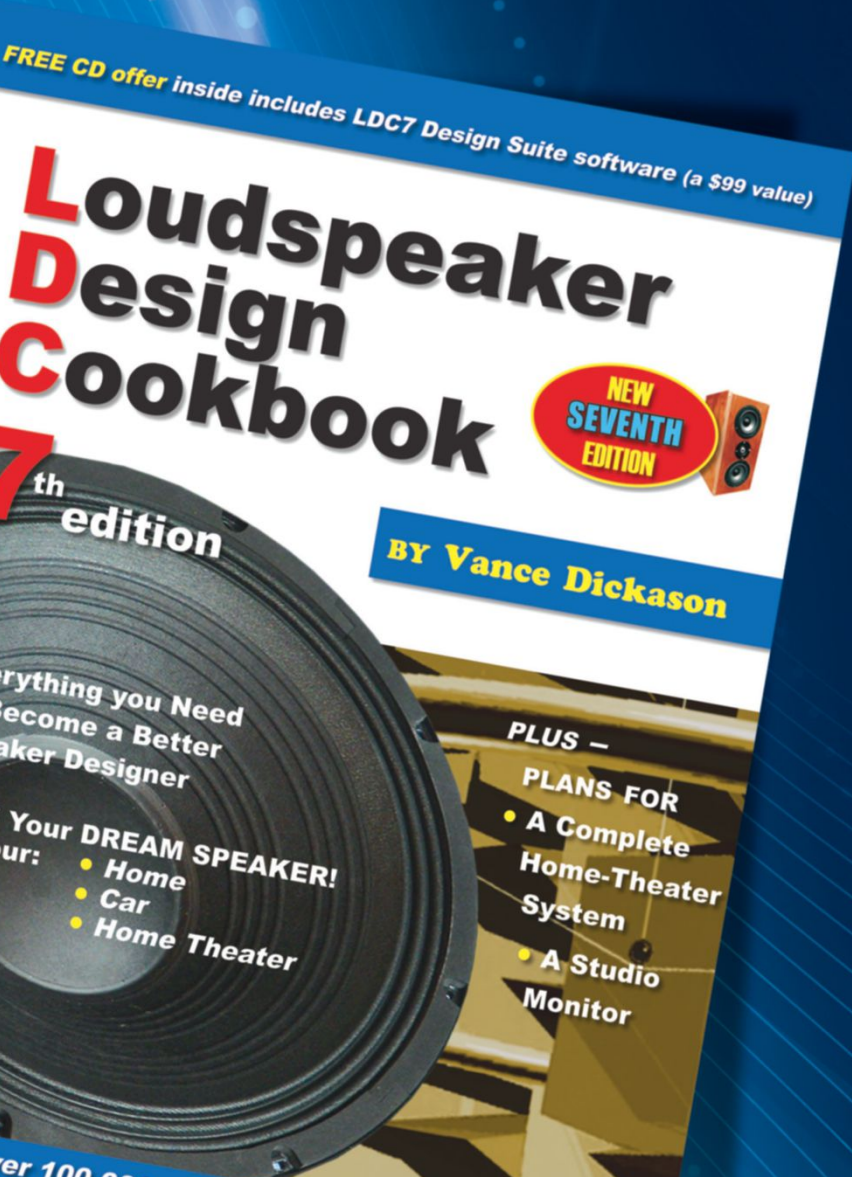
Back and better than ever, this 7th edition provides everything you need to become a better speaker designer. If you still have a 3rd, 4th, 5th or even the 6th edition of the *Loudspeaker Design Cookbook*, you are missing out on a tremendous amount of new and important information!

Now including: Klippel analysis of drivers, a chapter on loudspeaker voicing, advice on testing and crossover changes, and so much more! Ships complete with bonus CD containing over 100 additional figures and a full set of loudspeaker design tools.

A \$99 value!

Yours today for just \$39.95.

Shop for this book, and many other audio products, at
www.cc-webshop.com.



Love of Music Leads to New Business Venture

An Interview with Nancy Moon of MOON Amplification

By
Shannon Becker
(United States)



MOON Amplification is a family-owned company, bearing Nancy's maiden name of Moon.

SHANNON BECKER: Tell us a little about your background. How did you become interested in the audio field?

NANCY MOON: My interest in the audio field was driven by an immediate need to solve a long-

standing problem. I enjoyed listening to my husband play the keyboard. When my husband started using stereo headphones to try to get a better acoustical effect from the rotary speaker simulator in the organ, I found that I missed hearing the live music in the house.

It was at this point that I looked into purchasing a rotary speaker for my husband. But because of the cabinet size, low volume, and mechanical parts, I believed with some research and thought that this design could be updated using the latest technology available.

SHANNON: Why start your company, MOON Amplification (www.moonamp.com)?

NANCY: While looking for information on rotating speakers, I found that many forum members expressed the same interest in finding an existing portable product for live performance. I believe there is a market for an updated quality rotating speaker in a young, sexy, cool design with portability and reliability.

SHANNON: Is it a family run company? Who is involved and what are their roles?

NANCY: MOON Amplification is a family-owned company, bearing my maiden name Moon. As



skamp contains drivers from the Brazilian company PRV Audio.

inventor and president, I worked on the cabinet design as well as defining basics of what was needed to accomplish moving the sound around the room. Roy Davis, my husband and vice president, is the engineer who implemented many of the different engineering disciplines. My son, Charles Stone, works with social media and marketing. Other family members work in production.

SHANNON: What was your first product?

NANCY: Our first product was the ORBITED speaker, which we call "skamp." We use the word ORBIT to emphasize that the sound created moves around in a circle, not just turns on an axis. The model name skamp was chosen to emphasize high energy and portability, while taking advantage of the amplifier within it. We owned a wonderful SK keyboard and were building amps, it seemed logical. And with the dictionary meaning of the noun "scamp," it was perfect. The skamp weighs only 45.5 lbs, one-third of the weight of mechanical rotating speakers.

A unique advantage is that a clean piano and ORBITED organ can be played at the same time, using full power for both because the ORBITED spatial effect and the stationary left/right channels share the same amplifiers and speakers.

A highly efficient 36 V, 9.7 A switch-mode power supply, four 100 W RMS Class-D amplifiers, eight efficient neodymium speakers, and four compression horns make the skamp as loud as a guitar amplifier. There is a separate high-impedance guitar tube preamp for the ORBITED channel.

We use drivers from PRV Audio, of Brazil, because of our cabinet configuration and performance needs. The drivers had to be efficient and able handle high power, while fitting in a very constrained mechanical design. And because we needed the cabinet to be light and portable, neodymium magnets were a must.

Since there are eight mid-bass 6.5" drivers plus four horns in a cabinet that is 25" x 25" x 7.5" tall, the selection of reasonably priced drivers in this size with high-power handling is limited. The compression horn driver had to cover down to 800 Hz and drivers that would fit in this space and handle high power are difficult to find. I spent a lot of time looking for drivers with the right combination of performance, weight, and price.

SHANNON: Tell us more about the technology behind the skamp.



MOON's first product was the ORBITED speaker, which is also called "skamp." Here it is shown atop a tripod.



skamp is set on a deployed bassLite brown kartStand.

NANCY: With the electronically ORBITED skamp speaker, our patented technology has accomplished what was impossible. The skamp updates the 70-year-old mechanical rotating speaker, using a unique and critical acoustic design of the speaker cabinet as well as digital signal processing to literally move sound around the room.

skamp produces an acoustic spatial effect of the horn (treble) and bass rotors that literally interacts with the room, which is impossible to do with a simulator. Our engineering background came into play to solve mechanical design as well as electrical, software, DSP coding, and acoustic wave propagation problems.

Because the skamp cabinet produces the sound of both rotors, it can be played with or without a subwoofer cabinet—depending on how much bass you want. We also offer the skamp mixer/kontroller that puts all the knobs and inputs jacks close to the player with just a single cable.

SHANNON: How was design process? Do you have an engineering background?

NANCY: I spent a lot of time analyzing and researching how a mechanical rotating speaker really worked. I discovered the Doppler effect is only a part of it. The challenge was how to eliminate the mechanical parts and still have the nuances of the acoustic effect.

My professional background is technical writing. Roy is an electrical engineer with an audio engineering background. For many years, we both worked at Fortune 500 high-tech companies. In fact, we first met and worked together on the same project at one of those companies.

SHANNON: Please tell us a little about your other products and your customer base?

NANCY: I found that musicians who are interested in my product play live gigs and really want portability with ease of setup. And they want it loud! We offer bass cabinets in three sizes, depending on how much bass and portability is wanted.

We have the bassLite at 51 lbs, 350 WRMS through four 8" drivers. We use ferrite drivers because we have not found neodymium drivers that give us the quality bass we want.

The next step up is our bigBass, which weighs 55.5 lbs, 500 W RMS with a 15" neodymium driver and 3" voice coil. A down-firing front-loaded horn can be added with the kartStand when the side flaps are unfolded. The kartStand is used for easy transport and height utility for the skamp, bassLite, and bigBass.

Our subBase is styled to be a piece of furniture. We even offer one with a carved organ pipe motif. An opening at the bottom makes the cabinet look like it is on legs and a pyramid at the bottom increases the subwoofer's bass efficiency. It has 500 W RMS, a 15" ferrite driver, and a 4" voice coil.

SHANNON: How do you determine which speakers are best to use with the different instruments?

NANCY: The skamp is for small to medium venues such as a party or bar venue. Don't be fooled by



This is a Baroque subBass in wood grain with gold pipes. The wood grain skamp is on top.

the small size. Our artist Christopher Keyes plays at a 600-seat church each Sunday, using a skamp and bigBass alongside drum and guitar players. The pastor asked Chris to "please turn it down." We were all tickled at that request.

SHANNON: What's next for MOON Amplification?

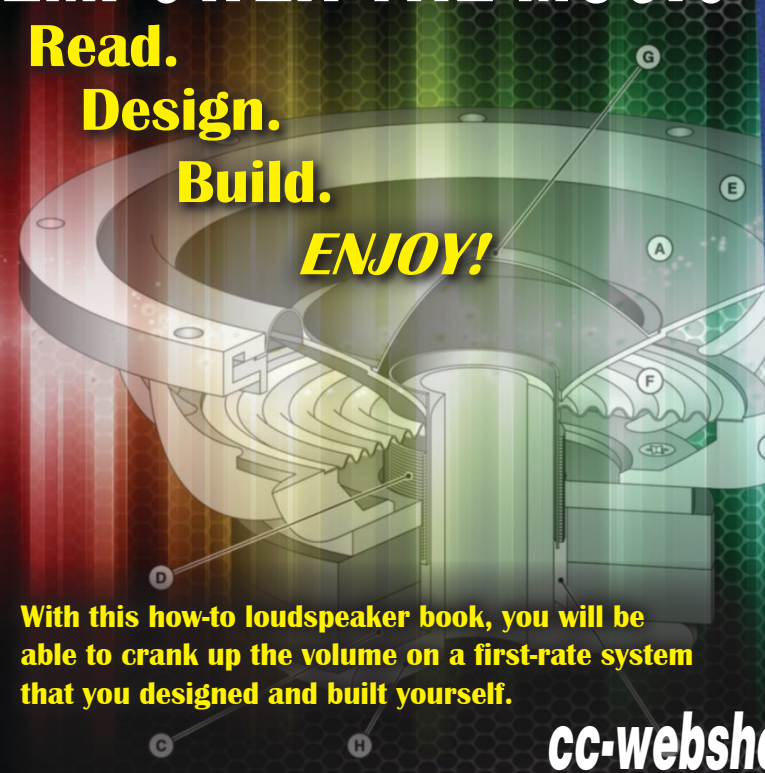
NANCY: We are exhibiting the skamp and our bass cabinets at the January NAMM 2017 (Booth# 1077, downstairs in Hall E) with all the other cool new technology exhibits. Next, we will finish the builds and beta testing of our Super skamp line based on larger horns and speakers. Further down the line, we are planning the Provado line with digital interfaces and the ability to stack any number of cabinets all spinning the sound around the room in synchrony. 



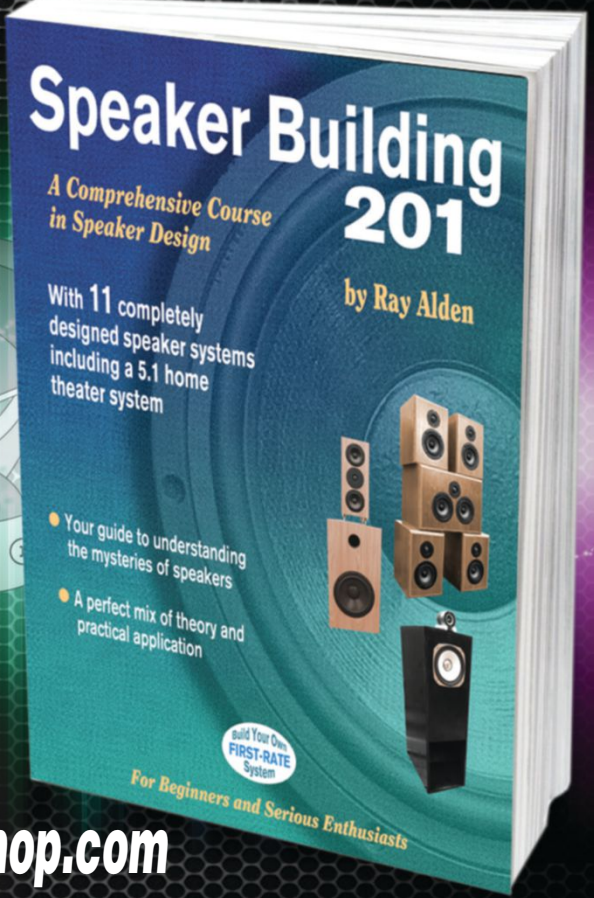
The bigBass weighs 55.5 lbs, 500 WRMS with a 15" neodymium driver and 3" voice coil.

EMPOWER THE MUSIC

**Read.
Design.
Build.
ENJOY!**



With this how-to loudspeaker book, you will be able to crank up the volume on a first-rate system that you designed and built yourself.



Speaker Building 201
A Comprehensive Course in Speaker Design
by Ray Alden

With 11 completely designed speaker systems including a 5.1 home theater system

- Your guide to understanding the mysteries of speakers
- A perfect mix of theory and practical application

build Your Own FIRST-RATE System

For Beginners and Serious Enthusiasts

cc-webshop.com

Colorful Amplification for Ultra-High Impedance Sources

If someone wants to record or amplify a musical instrument, the first step is to use some type of transducer to convert vibrations into moving voltages. In this article, Addie and Whisker of Toymaker Television (<http://tymkrs.com/>) describe their efforts to help a friend electrify her violin.

By
Addie and Whisker

(United States)

Transducers generally only output very low levels of electrical potential, so the second step is usually to amplify their output to a higher level. To help remove some of the guesswork from this step we have over time settled on some rough standards about what electrical properties this preamplifier stage should present at its output. We call this "Line Level." Fortunately, there are a myriad of commercially available preamplifiers designed to convert transducer sources into line level signals. Unfortunately ... a preamplifier can only do a good job if its input impedance matches the output impedance of the source transducer.

Recently, I ran into this exact problem while helping a friend design and build an electric violin. She is very experienced with acoustic

instruments in the context of an orchestra, but she is also interested in the world of live effects made possible via an electrified instrument.

A normal person would simply go out and buy an electric violin ... but my friend is an Aries. These type of people, who when they find out they are completely unequipped to deal with the mess they have gotten themselves into, simply enjoy it that much more. So off we went, working our way through each step as we came to it.

Throughout the project, we focused on using material we found on hand—scrap wood, leftover fiddle parts from other projects, no luthier-specific tools, you get the idea. At this point, we are working on the electric part. A normal person would go out and buy an instrument preamp, but we are (of course) building our own preamplifier from scratch and from salvaged parts (see **Photo 1**).

Electrifying the Violin

There are a few ways to electrify a violin, and for this we settled on a piezoelectric transducer embedded directly into the wooden bridge. Piezo pickups are great for this job, but come with a caveat: They have an ultra-high impedance. As in do-not-even-think-about-it unless your preamplifier supports an input impedance higher than 1 M Ω ! Even the "high-z" instrument inputs found on quality equipment will generally sound terrible with this

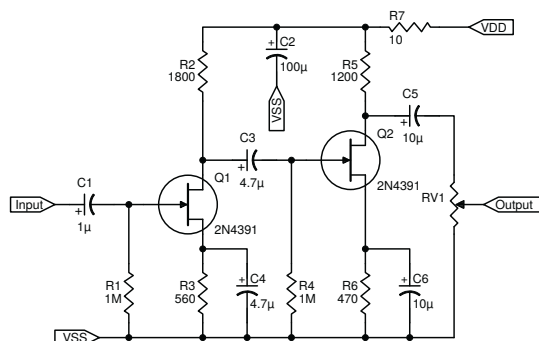


Figure 1: The electrical schematic is designed to fit into a very small package.

Photo 1: We opted to build our own preamplifier from scratch, using salvaged parts.

type of pickup due to the impedance mismatch.

Rather than let this be a problem for her each time she finds another interesting peddle to plug her instrument into, I thought it would be best to deal with the impedance matching right inside the preamplifier. If you look online for a piezoelectric preamplifier that buffers the ultra-high impedance to low impedance, you will find many similar designs based around a pair of MPF102 JFETs. I boiled a few of these designs down to come up with one I felt could be built into a very, very small package. (It does have to fit inside of a violin, after all.) **Figure 1** shows the electrical schematic in case you want to build one for yourself. I, however, did not use MPF102 JFETs for the final build. Something very odd led me down a different trajectory.

This past summer, we took a trip to Arizona to visit our friend John, a retired manufacturing engineer with a bit of a surplus auction hobby. John loves to collect broken-down old electronics, but he also lives with Lisa. Lisa is less thrilled about broken-down old electronics. Because I'm such a good friend (or because I like broken-down old electronics as much as John), I always offer to take John's surplus surplus back home to Minnesota. Lisa seems to approve of this offer, and sometimes John does too ... generally when he already has "five more of those over in that pile, I think."

Digging through the piles of geek gold (see **Photo 2**), I noticed something that piqued my interest. One of the mystery rack-mount enclosures was full of giant circuit board daughter cards, and each of those boards had a hundred or more metal-can type transistors. I asked John about it and he said, "that came from White Sands Missile Range. It was part of the telemetry system" (see **Photo 3**).

The Rackmount Enclosure

Wow! I pulled out a board and started looking for date-codes. 78 78 78 78 79 all discrete, no modern processors. Put together nearly 40 years ago, these circuit boards were assembled to "milspec" standards, using only the best components and probably spent their service life tracking missiles in flight or satellites in orbit. How cool! I thought about it for a minute, and started adding things up in my head. I mentioned to John that given how many multiples of each part there were on all these circuits, if any of the parts were presently in high demand for repairing vintage equipment ... he might be sitting on a big pile of cash.

High-quality vintage transistors are getting harder and harder to source! We did a quick search on a part number of a JFET we noticed was used over and over on the boards, and sure enough ...

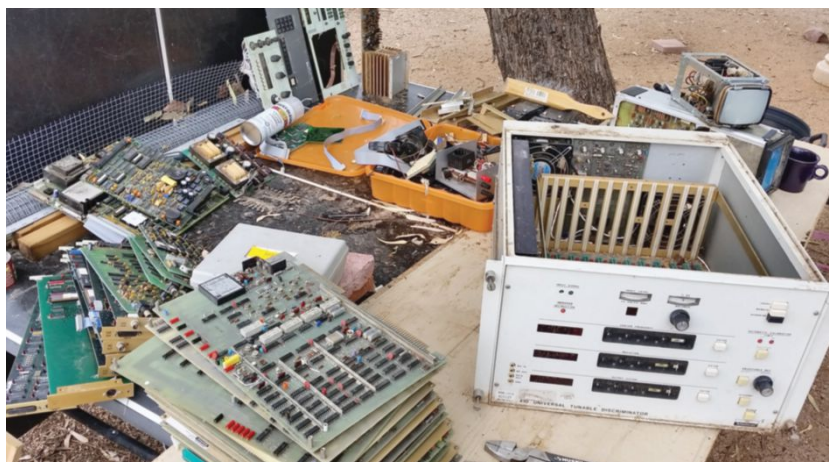


Photo 2: After digging through our friend's pile of geek gold, I found a pile of giant circuit board daughter cards.



Photo 3: The cards were part of a telemetry system from White Sands Missile Range.



Photo 4: I opened up the enclosures to see what we could salvage.

About the Authors

Whisker is a self-taught audio engineer with a focus on designing and building his own custom gear as well as production of all of the audiovisual content for Tymkrs. Whisker also designs and sells DIY audio kits, tools, and synthesizers at <http://tindie.com/stores/tymkrs>

Addie is a research study coordinator and cardiovascular surgery nurse who got into electronics while studying for her amateur radio operator license test. She sings, plays guitar, violin, and viola. Addie also designs and sells educational electronic kits on Tindie.

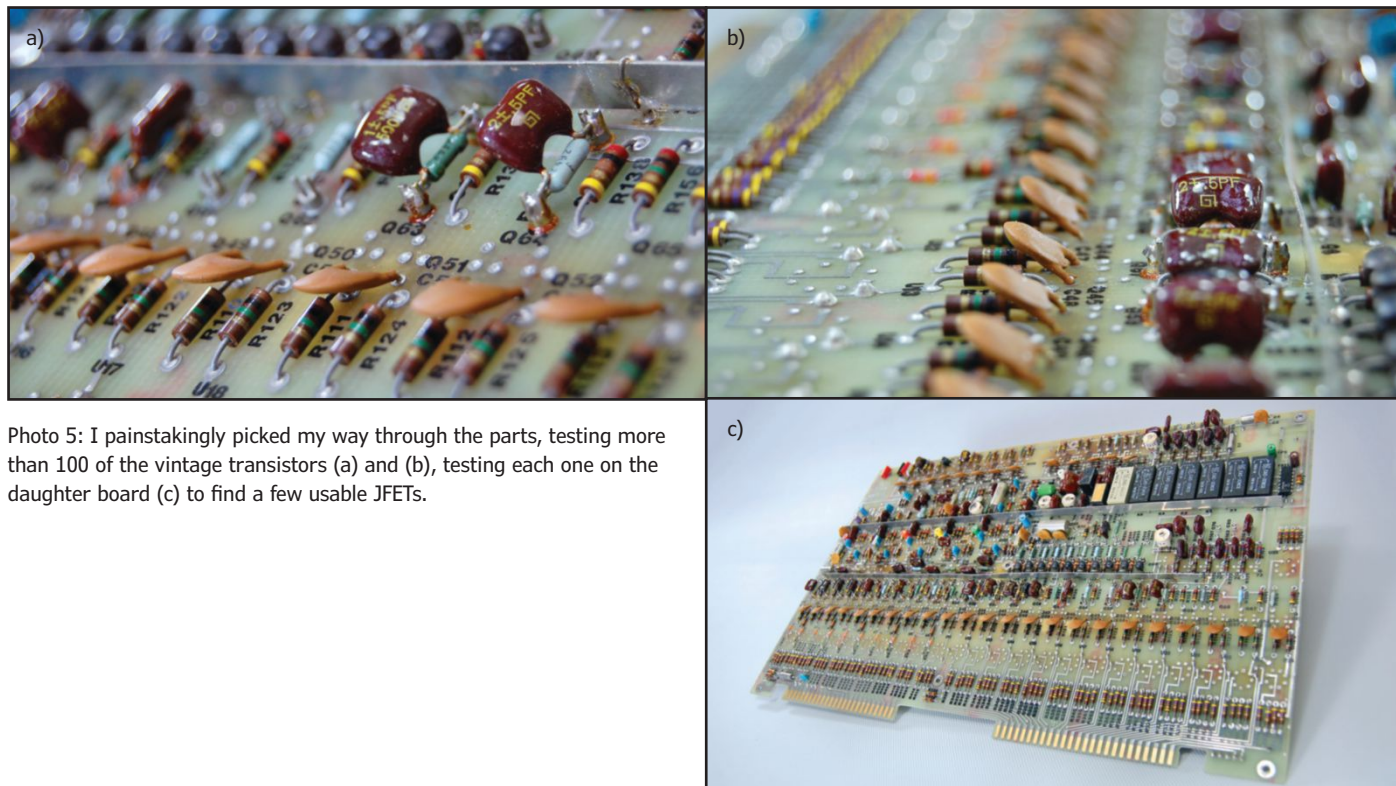


Photo 5: I painstakingly picked my way through the parts, testing more than 100 of the vintage transistors (a) and (b), testing each one on the daughter board (c) to find a few usable JFETs.

if the parts were still functioning then they would be well worth salvaging (see **Photo 4**). However, the rack-mounted enclosures had been baking in the hot desert sun for years, and a few of them had even been taken over by pack rat nests, eww ... the parts could be long ruined.

I wanted to find out if they were still working, so John sent me home with two of the daughter circuit boards. Using a hot-air pencil, I carefully removed about 100 JFETs. I went through the entire batch of parts, painstakingly testing each one (see **Photo 5**). Sadly, only a handful of them had survived. John's retirement fund (or pile of junk, if you ask Lisa) may not be worth millions. But on the bright side, a few JFETs are enough to build a couple of ridiculously mojo-filled musical instrument preamplifiers!

Using the 2N4391 JFETs

I shared these developments with my violin-building friend, and she agreed that we should use missile-tracking 2N4391 JFETs rather than the normal people's MPF102 JFETs. If a transistor is good enough to track Titan missiles, then it is obviously appropriate to be in her fiddle (Aries logic).

So, I designed a tiny little circuit board (see **Photo 6**). For the next step, we installed all of the electrical components and started testing the JFETitan (see **Photo 7**).

A bad impedance match to a piezoelectric pickup has a very obvious sound to it. The high end is exaggerated and brittle sounding. It also emits a bass sound I've heard described as "quacky." This was quite obviously not the case with our JFETitan preamplifier. This thing sounds great!

I tried it on various instruments here in studio, all with piezo pickups. It had a very natural sound to it, but different from other ultra-high impedance preamplifiers I've heard in the past. Piezos are typically used as a contact mic, and because they are physically attached to the instrument, they can be very boomy. This one was not boomy at all, so I decided to investigate further.

Testing the JFETitan Preamplifier

To get to the bottom of things, I replaced the piezo mic with an input jack and used my computer to do a test. I sent white noise into a line input on

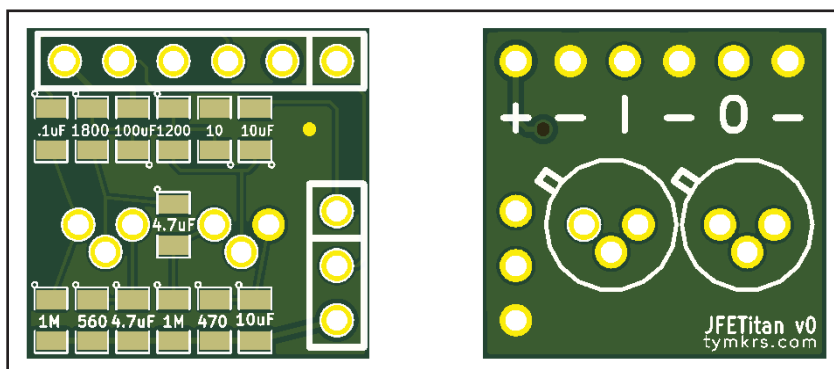
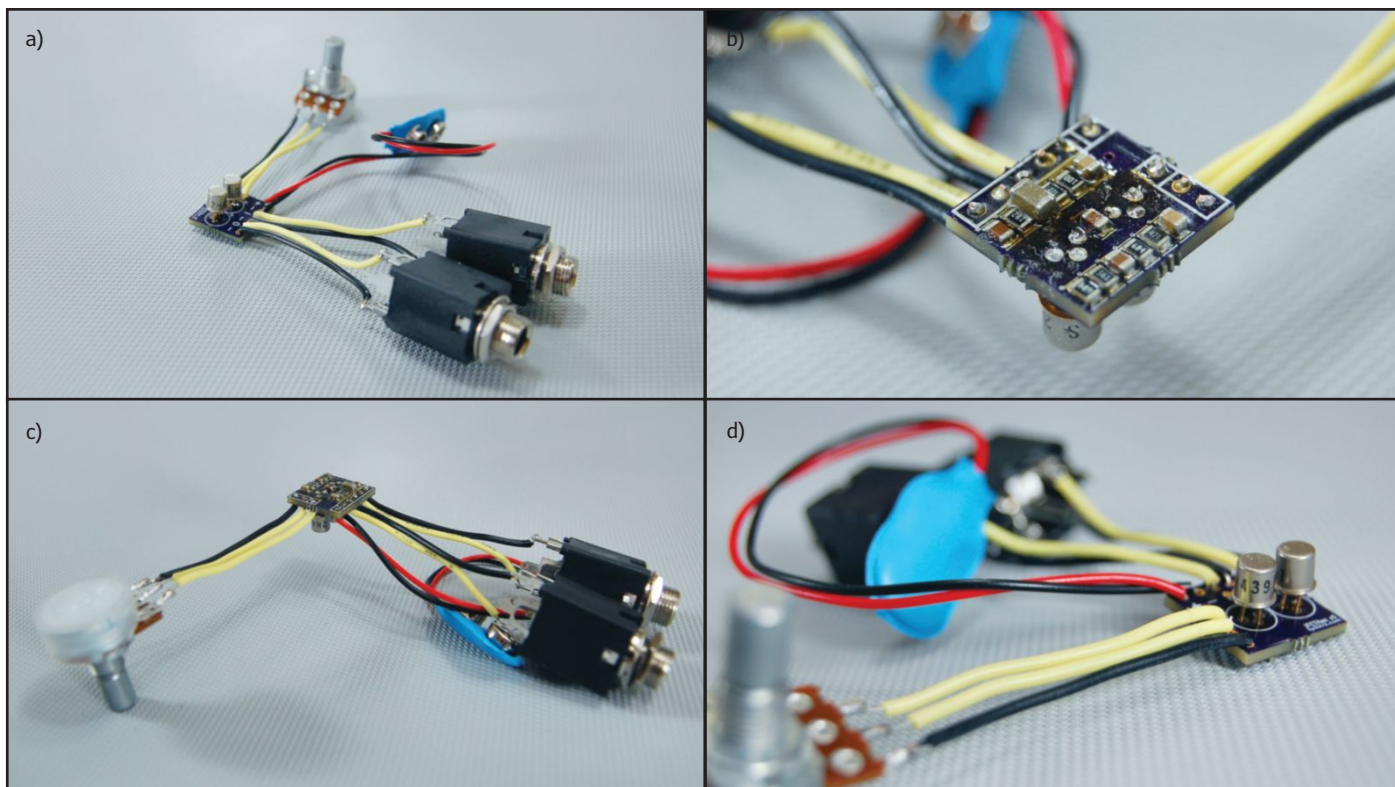


Photo 6: I designed a PCB that would be small enough to work with the violin.



my computer's audio interface, and looked at the frequency response curve. In theory, white noise should have all of the frequencies present, giving a pretty flat response curve, and it was, other than some steep attenuation in the high frequencies (probably something to do with the sampling frequency of the audio interface or something along those lines).

Then, I sent white noise to the preamplifier, and connected its output to the same line input on my computer's audio interface. I noticed a similar roll-off of the high frequencies but an additional gentle roll-off in the low and low-mid frequencies. **Photo 8** shows some images of the curves. The measurement results explain why there was no boominess. Those battered old JFETs were acting a bit like a very gentle high-pass filter.

I tested it with MPF102 JFETs as well, and it stays flat deeper into the low frequencies before rolling off (for anyone wanting to build one). It just worked out that the roll-off starts below the frequencies I want to capture from a guitar or violin. The filtering effect is only removing things I would normally have to remove later with an EQ, perfect!

The Result

Often, in most any art form, and especially in making music ... it is as much about what is imperfect in a medium as the perfection that draws us in closer, making us want to stay. Sometimes it is the texture of canvas peeking through the colors of a painting. Sometimes, it is a once terribly important little transistor—after losing his job and living in a hobo camp full of smelly pack rats—making a

comeback and joining a band.

Sometimes what a musical instrument needs to make it special is a little colorful character. 

Photo 7: From the salvaged parts, we put together the JFETitan.

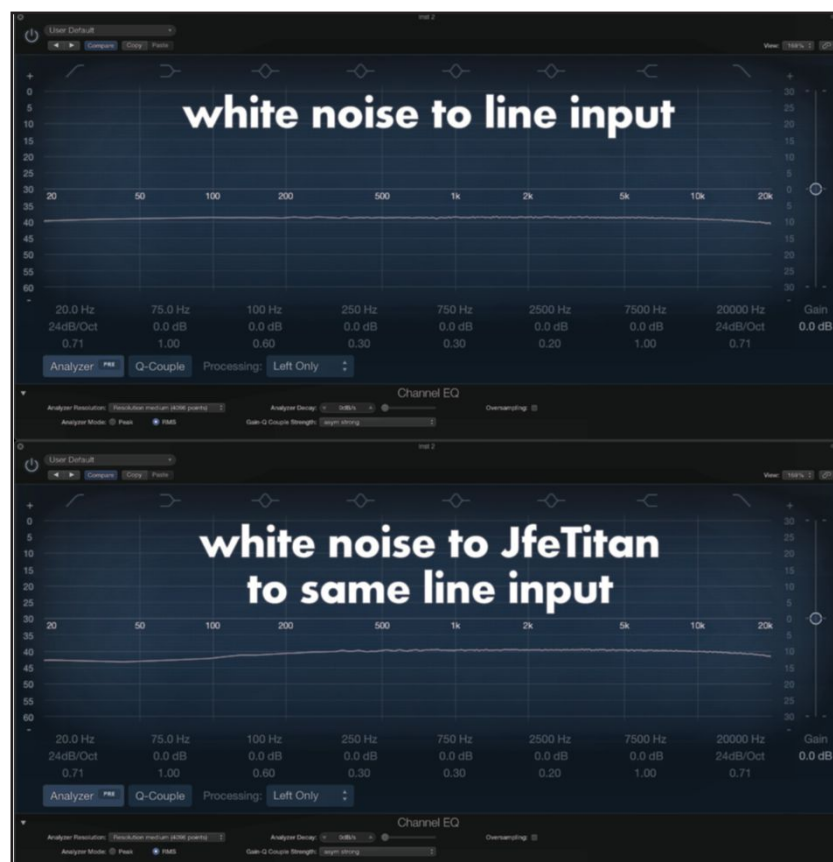


Photo 8: To test the JFETitan, I sent white noise into a line input on my computer's audio interface and examined the frequency response curve.

Build a Low-Power, Class-A Power Amplifier (Part 2)

Construction and Performance Tests

In the first part of this article, I presented the groundwork for my Class-A amplifier design. Now, I will describe the construction process and share my measurement test results.

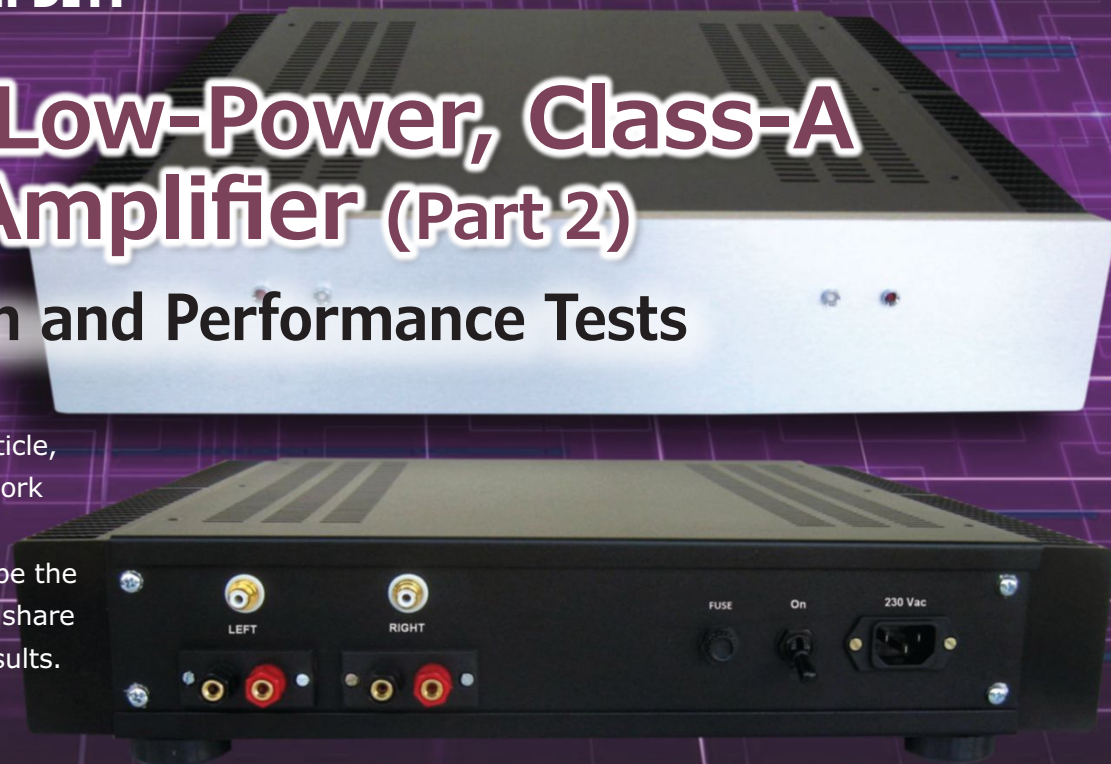


Photo 1: The amplifier's front and back panels are shown.

By
George Ntanavaras
(Greece)

In the first part of this article, I described my objectives for this Class-A amplifier design. I also analyzed the electronic diagrams of the power supply and the main amplifier circuits. In this second part, I will explain the construction of the amplifier and provide the measurements that I took to verify its performance.

But first, I would like to offer an important safety note. A DIY construction of an amplifier, as with any other equipment that contains a mains power supply, is potentially lethal. If you are not confident with your skills or don't take all the required precautions, you should not attempt this amplifier construction.

The Assembly

For the amplifier's construction, I used high-quality parts including high-precision metal film resistors, polypropylene capacitors, and non-polar electrolytic wherever possible. I have included the amplifier's parts list. For the transistors, I kept most of the original types as recommended by J. Linsley-Hood. Before I placed the transistors on the PCB, I checked every one of them in a separate circuit.

I used double-sided PCBs with ground planes on the top and bottom layers. The board material is from FR4 with 1.6 mm thickness and 35 μ m copper. The guide assembly for the PCB1 with the

raw power supply and the protection circuit is shown in **Figure 1**. I made two 4-conductors cables for connectors CON1 and CON2. **Table 1** describes the assignment of every pin of the CON1 and CON2 connectors. The CON1 is used for the connection of PCB1 to the thermal switch and the temperature alarm LED.

The CON2 is used to connect PCB1 to PCB2, except pin 4, which should be connected to the 18 VAC of the power transformer. I used one of the AC pins of the bridge rectifier to connect this pin. When the PCB1 was assembled, I double-checked the polarity of all the electrolytic capacitors and I used a separate setup to test the PCB1 to avoid any possible damages. The external power supply unit (PSU) that I have in my lab contains regulated outputs that are protected from short circuits or over currents and was connected to the +V1, -V2, and the two GNDs of the PCB1. I slowly increased the voltage output of the PSU up to about ± 24 VDC, observing that everything was operating well. Then I connected the PCB1 to the bridge rectifier. I used a variac to slowly increase the mains voltage. I measured the output voltages at about ± 24 VDC. For safety reasons, I temporarily placed 3 k Ω /2 W resistors across the power supply capacitors during testing to slowly bleed off the charge after the AC power had been removed. The reservoir capacitors

store a significant amount of energy and if they are accidentally shorted out by anything conductive, then a dangerous burn is likely. So, caution is needed and remember not to put fingers or metallic tools into the amplifier.

I checked the protection circuit for correct operation when a DC fault appeared. I temporarily connected the coil of relay K1 at pins 2 and 3 of the CON2. Then, I directly applied a DC voltage of about +2.5 V first and then -2.5 V to Pin 1 of the connector CON2 of the protection circuit. Both voltages should activate the protection circuit by deactivating the relay. The LED should be red. Finally, I short circuited pins 1 and 2 of the thermal switch. This simulates a temperature alarm and the Temperature Alarm LED should be turned on, and the Relay K1 should be deactivated.

The assembly of the second PCB2 is shown in **Figure 2**. The output transistors TR1 and TR2, the pass transistors TR8 and TR9, the regulators IC1 and IC2, and the TR5 transistor were placed on an L-shaped angle bracket, which is in thermal contact with the main heatsink. All the above transistors, through this bracket, were directly soldered onto the PCB.

All the transistors and regulators should be electrically isolated from the heatsink. I used insulating bushes and mica insulators. I also put thermal compound between the L bracket and the main heatsink, between the L bracket and the mica insulators and between the mica insulators and the transistor cases to ensure good thermal contact.

After assembling all the transistors and the regulators, I carefully checked to ensure there was no electrical connection between the transistor cases and the IC regulator cases with the metal L bracket. Once I completed assembling the PCB2, I tested it in a separate setup to avoid any possible damages. I took the external PSU in my lab and set its output to ± 20 VDC. Then, I connected them to the +VC, -VC and the GND+ and GND- of the PCB. The two ground points of the regulated power supplies GND+ and GND- and the signal ground point IN- are not connected together on the PCB so I connected them externally.

I also made sure the output of the voltage regulators close to ± 14.5 VDC. Next, I set the bias current of the output power transistors at about 100 mA or 34 mV across the resistor R10A or R10B using the trimmer BIAS. The output voltage should also be set close to 0 V with the trimmer OFFSET.

The Enclosure

Then, I continued assembling the rest of the amplifier, the adjustment, and the testing.

I chose a ready-made, nice-looking, metal enclosure (450 mm wide $\times$ 400 mm deep $\times$ 80 mm high) to house all the amplifier components. The front panel is made of 10 mm aluminum. On the sides of the enclosure, I have four large heatsinks (each one is 200 mm long $\times$ 40 mm wide $\times$ 80 mm high) to provide adequate heat dissipation for the power amplifier. This is very important because it will keep the temperature rise low and will increase the life of the amplifier's power components.

The amplifier's front panel (see **Photo 1**) has four indication LEDs—two for each channel. One

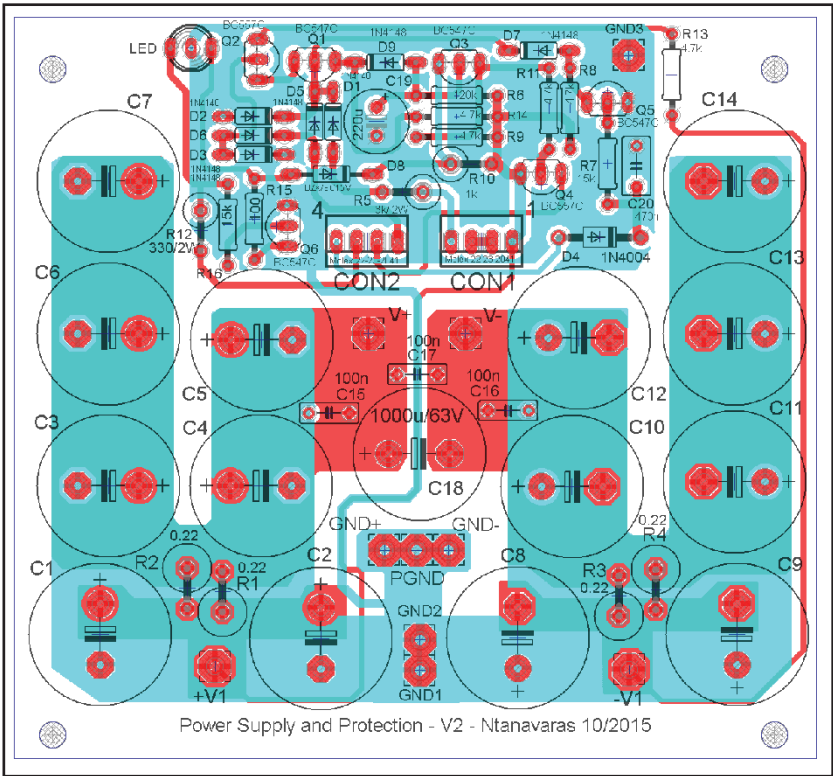
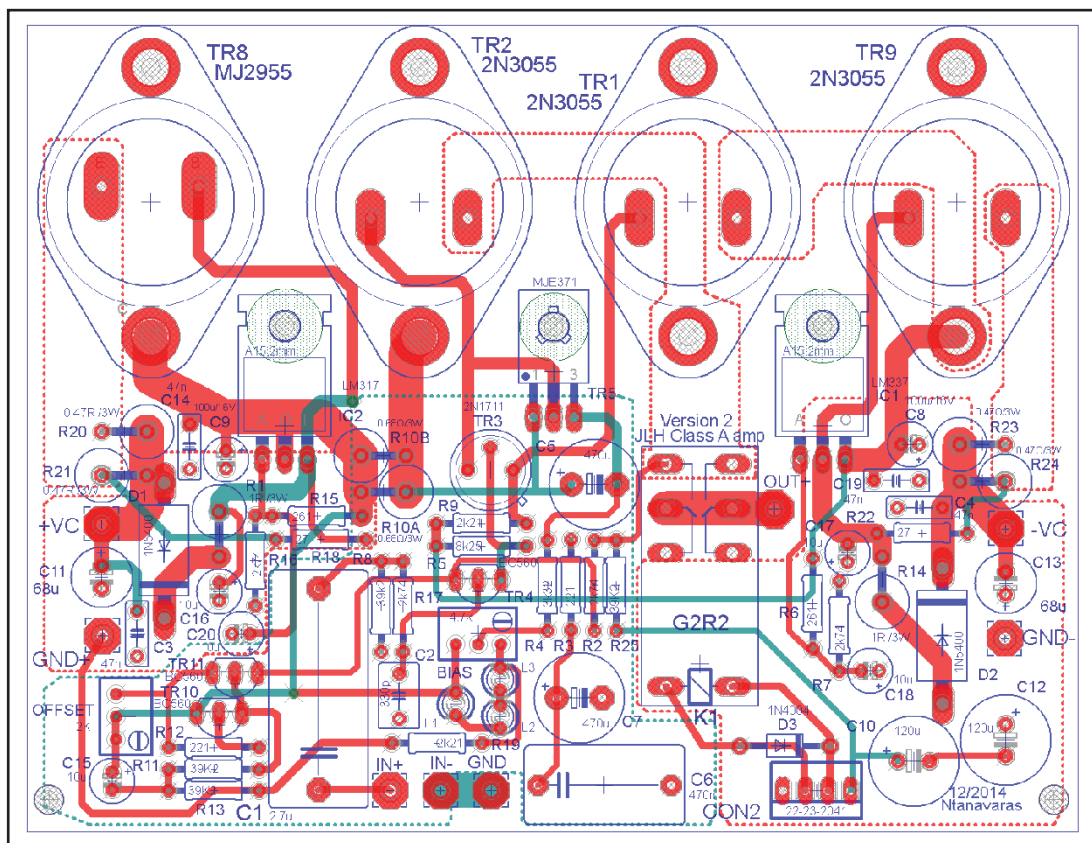


Figure 1: This is the guide assembly for the PCB1 with the raw power supply and the protection circuit.

CON1	Pin	Description
	1	Thermal switch pin 1
	2	Thermal switch pin 2
	3	Temperature alarm LED (+)
	4	Temperature alarm LED (-)
CON2	Pin	Description
	1	RC point
	2	Relay pin (+)
	3	Relay pin (-)
	4	Connected to 18 VAC

Table 1: Pin assignments for the connectors CON1 and CON2

Figure 2: Here is the assembly of the second PCB2.



LED indicates the amplifier's normal operation and the other indicates an over-temperature alarm when the thermal switch is activated.

On the amplifier's back panel (see **Photo 1**), the following are mounted from left to right: the IEC socket for the 230 V input, the Power switch, the fuse holder, two gold-plated RCA phono sockets for the signal inputs, and the connectors of the amplifier outputs. There is enough distance between the two groups to avoid any interference from the line voltage to the signal voltages. The phono sockets are insulated from the chassis and the twisted pair cables were used for the connection of the input sockets to the PCB to eliminate any possible electromagnetic interference from external sources.

Photo 2 shows an internal view of the amplifier. I properly insulated all the mains connections so they cannot be accidentally touched and maintained adequate spacing between the live connections and other conductive parts. I connected the protective Mains ground from the AC power input to the amplifier's chassis near the input ground. For safety reasons, I consider it absolutely necessary that the metal chassis of the equipment must be securely earthed to the mains earth to prevent it from becoming live if a mains connection falls off.

I mounted the power transformer in the front of the enclosure with additional metal shielding around the transformer to eliminate any possible interference to the amplifier's electronics parts. I placed two bridge rectifiers to the left and to the right side of the transformer. I also put the two PCBs for each channel close to each other to keep the length of the cables as short as possible. I attached the L-shaped bracket of the main



Photo 2: This is an internal view of the completed amplifier.

Part list

Description	Value	Pieces	Comments
Metal box for the amplifier (Dimensions: 450 mm × 400 mm × 80 mm)	1NPD02400B	1	MODUSHOP, HIPI 2000
Mains fuse	1 A, 250 V, Slow blow, Anti-surge	1	At the input of the transformer primary
Cartridge fuse holder, panel mounted		1	For the 5× 20 mm mains fuse
Inlet IEC, 250 VAC, 10 A		1	For panel mount
Power transformer	2× 18 VAC, 225 VA	1	One transformer for both channels
Fuses (transformer secondary)	2 A, 250 V, Time lag	4	At the output of the transformer secondary
Cartridge fuse holder, throughhole		4	for 5× 20 mm fuse
Bridge rectifier	35 A, 100 V	2	GBPC3501
Phono socket, chassis, white		1	For panel mount
Phono socket, chassis, red		1	For panel mount
Double banana jack for loudspeaker connection		2	For panel mount
Bi-color LED	Common cathode	2	For panel mount
Over-temperature Alarm LED	Red LED	2	For panel mount, 6 mm
Thermal switch	Max. actuating temperature 70°C	2	Mounted on the heatsink

Parts for One PCB1

C1, C2, C3, C4, C5, C6, C7, C8, C9, C10, C11, C12, C13, C14	3900 uF/35 V	14	Electrolytic capacitor, Panasonic FC series, RM =7.5 mm, Diam. = 18 mm
C15, C16, C17	100 nF/100 V	3	Polyester, RM = 5 mm
C18	1000 uF/63 V	1	Electrolytic capacitor, Panasonic FC series, RM = 7.5 mm, Diam. = 16 mm
C19	220 uF/35 V	1	Electrolytic capacitor, Panasonic FC series, RM = 3.5, Diam. = 8 mm
C20	470 nF/63 V	1	Polyester
CON1, CON2	Molex 22-23-2041	2	Connector Molex header four-way
Mating connector for CON1, CON2	Molex 22-01-2047	2	Molex housing, receptacle
Pins for connectors	Molex 08-50-0114	8	Molex terminal, solderless
D1, D2, D3, D5, D6, D7, D9	1N4148	7	Diode
D4	1N4004	1	Diode
D8	BZX55C15V	1	Zener diode 15 V, 0.5 W
LED	Bi-color LED 5 mm	1	Red, Green LED with Common cathode
Q1, Q3, Q5, Q6	BC547C	4	NPN transistor
Q2, Q4	BC557C	2	PNP transistor
R1, R2, R3, R4	0.22 Ω/3 W	4	Metal film, 5%, 3 W
R5	3 kΩ/2 W	1	Metal film, 5%, 2 W
R6	120 kΩ	1	Metal film, 1%, 0.25 W
R7, R16	15 kΩ	2	Metal film, 1%, 0.25 W
R8, R9, R11, R13, R14	4.7 kΩ	5	Metal film, 1%, 0.25 W
R10	1 kΩ	1	Metal film, 1%, 0.5 W
R12	330 Ω/3 W	1	Metal film, 5%, 3 W
R15	100 Ω	1	Metal film, 1%, 0.25 W

Parts for one PCB2

C1	2.7 µF/250 V	1	Metallized polypropylene
C2	330 pF/100 V	1	Polypropylene film, 5%
C3, C4, C14, C19	47 nF/63 V	4	Metallized polyester (PET)
C5, C7	470 uF / 16 V	2	Electrolytic capacitor, nonpolar
C6	470 nF / 250 V	1	Polypropylene capacitor
C8, C9	560 µF/25 V	2	Electrolytic capacitor, Diam. = 5 mm
C10, C12	120 µF/35 V	2	Electrolytic capacitor
C11, C13	68 µF/35 V	2	Electrolytic capacitor, Panasonic FC Series, Diam. = 6.3 mm
C15, C16, C17, C18, C20	10 µF/35 V	5	Electrolytic capacitor, nonpolar
CL	100 nF/250 V	1	Polyester (mounted on output connector)
CON2	22-23-2041	1	Connector Molex header four-way
Mating connector for CON2	22-01-2047	1	MOLEX housing, receptacle
Pins for connector CON	08-50-0114	4	MOLEX terminal, solderless
D1, D2	1N5401	2	Diode
D3	1N4004	1	Diode
IC1	LM337 BTG	1	Linear voltage regulator
IC2	LM317 BTG	1	Linear voltage regulator
K1	Output relay	1	Omron Relay G2RL-2-CF, 8A, 12VDC
L1, L2, L3	LED	3	3 mm, RED, 20 mcd, 625 nm
OFFSET	Trimmer 2 kΩ	1	25 turns, Type 3299W
BIAS	Trimmer 5 kΩ	1	25 turns, Type 3299W
R1, R14	1 Ω/3 W	2	Metal film, 5%, 3 W
R2, R7, R16, R17	2.74 kΩ	4	Metal film, 0.1%, 0.25 W
R3, R12	221	2	Metal film, 0.1%, 0.25 W
R4	3.32 kΩ	1	Metal film, 0.1%, 0.25 W
R5	8.25 kΩ	1	Metal film, 0.1%, 0.25 W
R6, R15	261 Ω	2	Metal film, 0.1%, 0.25 W
R8, R11, R13, R25	39.2 kΩ	4	Metal film, 0.1%, 0.25 W
R9, R19	2.21 kΩ	2	Metal film, 1%, 0.25 W
R10A, R10B	0.68 Ω/3 W	2	Metal film, 5%, 3 W
R18, R22	27 Ω	2	Metal film, 1%, 0.4 W
R20, R21, R23, R24	0.47 Ω/3 W	4	Metal film, 5%, 3 W
RL	5.6 Ω/3 W	1	Metal film, 5% (mounted on output connector)
TR1, TR2, TR9	2N3055	3	Power transistor, NPN, on heatsink
TR3	2N1711	1	Transistor, NPN, on heatsink
TR4, TR10, TR11	BC560	3	Transistor PNP
TR5	MJE371	1	Transistor PNP, on heatsink
TR8	MJ2955	1	Power transistor PNP, on heatsink
Mica insulating kit	For TO-3 case	4	
Mica insulating kit	For TO-220 case	2	
Fixing kit		1	M3 Screws, Nuts, Spring, Flat Washer, Solder Tag
TO-5 heatsink 44 °C/W		1	Heatsink for TR3

***Corrections: In Part 1 of the article (*audioXpress*, January 2017): In Figure 1: TR9 should be 2N3055

In Figure 3: R1, R2, R3, R4, and R12 should be 3 W and D8 should be BZX55C15V

amplifier to the middle of the heatsink for uniform heat dissipation. The ground signal and the two power supply grounds of the main amplifier are not connected on the PCB2 to avoid ground loops problems. I connected them together at a common point near the signal input ground. Then, I used a small PCB to place the components RL and CL and connect them across the output of the amplifier.

Adjusting the Amplifier

After I finished assembling the amplifier, I connected it to my variac and slowly increased the mains voltage, while measuring the power supply voltages. When everything was correct, I adjusted the bias current and the output offset of the amplifier. I connected the voltmeter to read the voltage across the R10B resistor and I adjusted the BIAS trimmer setting the voltage at 340 mV. This corresponds to a bias current of 1 A. I also adjusted the output offset with the trimmer OFFSET close to 0 mV. In my prototype, it was within ± 10 mV. I let the amplifier continuously operate for more than 24 hours and I used a digital thermometer to occasionally monitor the bias current, the output offset, and the temperature of the heatsinks.

A fast-acting fuse of approximately 1 to 1.2 A (depending on the load of the amplifier) can also be placed at the amplifier's output to offer additional protection in case of a fault.

The Amplifier's Performance Measurements

I performed several measurements to check the amplifier's performance and to confirm that its performance was at least similar to that of the amplifier presented in Linsley-Hood's original article.

I measured the amplifier's gain at 22.3 dB ($\times 13$). The amplifier's maximum voltage output for a 0.3% distortion was measured at 7.2 VRMS (or 10.2 V peak) with an 8 Ω load and 4.4 VRMS (about 6.2 V peak) with a 5 Ω load. Those measurements correspond to an output power of 6.5 WRMS for 8 Ω and 4 WRMS for 5 Ω .

I measured the amplifier's total harmonic distortion (THD) with the ARTA software. Its second-order harmonic was predominant as expected. All other harmonics are an order of magnitude lower than the second harmonic except the third harmonic, which rises slowly above 1 kHz.

Since the amplifier will drive a high-frequency unit, I also took another measurement with a Seas MU10 full-range driver connected to the amplifier's output. I set the output level at 1 VRMS—a typical value during a normal listening level. **Figure 3** shows the distortion that I measured. Again the second-order distortion is predominant at about 0.1% up to 500 Hz and then starts to slowly drop probably because the impedance of the driver unit is increased.

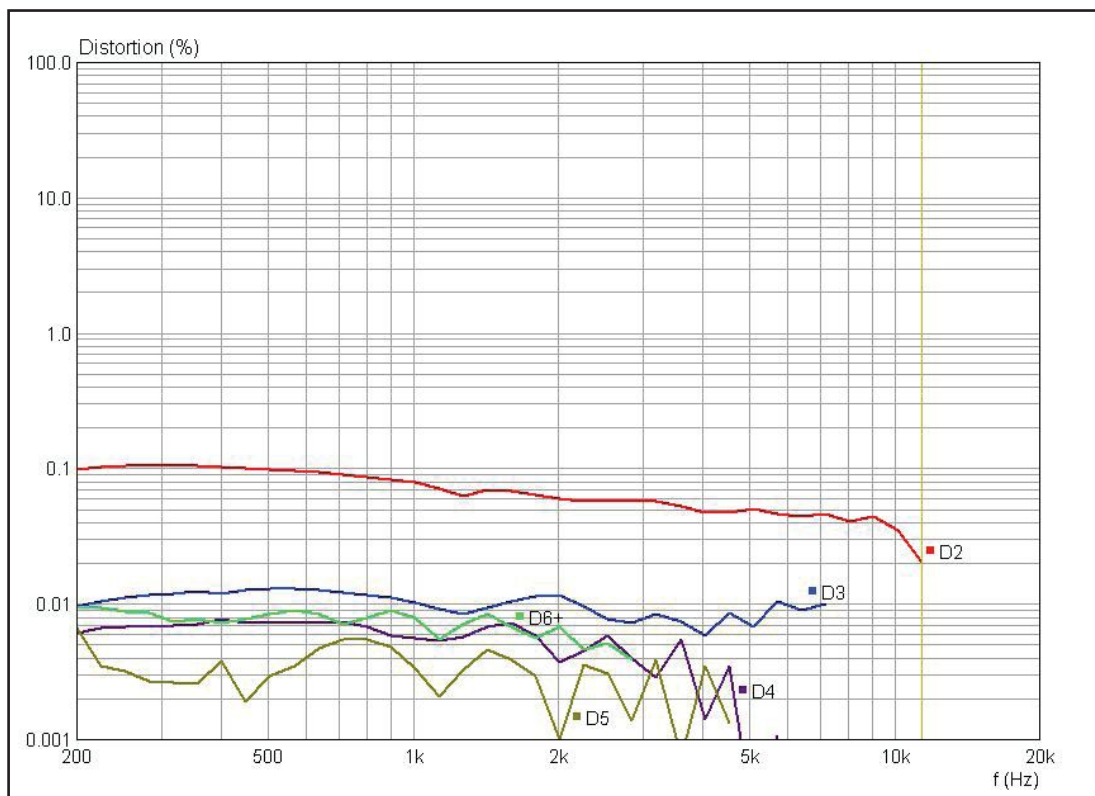


Figure 3: I tested the amplifier with the output level set at 1 VRMS, a typical value during a normal listening level.

When I finished the tests, I added the amplifier to my home stereo system in a place that provided good ventilation around the amplifier, since it runs hot (about 25°C above room temperature) with the heatsinks. It is important to keep the temperature of the heatsinks as low as possible, since this will increase the life of all the amplifier's components, especially the transistors and the regulators.

Sound Quality

I know I should not express my opinion about an amplifier that I have built, but I cannot resist the opportunity to quote the words of J. Linsley-Hood from his original article about the quality of the amplifier: "The Class A systems have a complete freedom from the slight 'edginess' found on some high string notes with all the Class B units, but they appeared also to give a fuller, 'rounder,' quality." I think his words exactly describe what someone should expect to hear from this amplifier.

Power

How much power do I really need for my tweeters? As this is a low-power amplifier, one question was always on my mind. Does it have enough power to drive the tweeters? So I decided to perform some measurements to find out.

I connected my voltage and current peak detector between the output of the power amplifier and the tweeters. This is a simple portable device that can simultaneously measure the peak voltage and the peak current at the input of a loudspeaker during reproduction of normal music. I set my system's volume to its normal position that gives—not very often, I have to admit—peak levels of about 90 dB SPL.

Since I was measuring a tweeter driver that was crossed over at 2 kHz with a fourth-order filter and I usually listen to classical music, I chose musical tracks that had loud passages of trumpets, oboes, and piccolo oboes.

What I measured was promising. The peak voltage was usually at 1.5 to 2 V. Even during loud passages—above my normal listening levels, where my sound level meter showed almost 100 dB SPL peaks—the peak voltage went up to 3.5 V while the peak current was 0.6 A. This is not surprising since the characteristic sensitivity of the tweeter is 93 dB SPL/2.83 VRMS/1 m and has a minimum impedance of 5.5 Ω at the region of 1 to 20 kHz.

After a lot of research, I found two music tracks that forced the amplifier to clip. I would recommend using these tracks to anyone who wants to test the limits of their own system's tweeters.

The first track is the first movement of Gustav

Mahler's "Symphony No. 5" with Eliahu Inbal as the conductor of the Frankfurt Radio Symphony Orchestra, on a CD published by Denon. I analyzed the first 1.5 minutes of this track and I found the peak value of the recorded signal was just -0.35 dB below the maximum level. Then, I applied a high-pass filter at 2 kHz with a 24 dB/octave slope and the peak value dropped to -2.3 dB, very close to the previous value.

The second track I used for testing was from a 1965 Miles Davis album titled *E.S.P.* The last track of this album has the song "Mood" and it is a real test for the tweeter. About 1 minute and 5 seconds from the start is a loud trumpet passage almost at the clipping level, which has a broad spectrum from 4 to 10 kHz. When I played these two tracks at high levels, I measured a peak voltage of 6.5 V, which is close to the maximum amplifier's output level when it drives a load close to 5.5 Ω, similar to that of the tweeter.

I think the amplifier is capable of driving my tweeters without clipping, but you can always find a track that will force the amplifier into clipping. Of course, all the above measurements are strongly dependent on the peak sound level, the kind of music, and of course, the sensitivity of the drivers.

Conclusion

This is a low-power, high-quality power amplifier. I use it to drive the tweeters of my loudspeaker system, but it could also drive other high-efficiency passive speakers. It has excellent transparency and I am fully satisfied with its performance. I enjoy its reproduction quality so much that my next project will be a version with more output voltage and current capabilities. 

Resources

EAGLE PCB Design

Cadsoft | www.cadsoftusa.com

LTspice simulator program

Linear Technology | www.linear.com/designtools/software

About the Author

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

The Oscilloscope

Learn more about voltages and waveforms produced by specific failures and how an oscilloscope can help troubleshoot hollow-state circuit problems.



Photo 1: A 10:1 probe increases both the maximum voltage and the input impedance of an oscilloscope by a factor of 10.

By
Richard Honeycutt
(United States)

Whether your involvement with hollow-state circuits is as a DIY builder, a designer, or a repair technician working with hollow-state circuits, knowing how to troubleshoot circuit problems will be important at some point. Back in August 2015 we published an article on troubleshooting hollow-state equipment. That article was symptom-oriented. A couple of months later, we published an advanced troubleshooting article discussing signal injection and signal tracing. This month's article shows the voltages and the waveforms produced by specific failures.

DC and AC voltages can, of course, be measured by a voltmeter, but many technicians prefer to use an oscilloscope since it can also display waveforms. Because many hollow-state circuits use supply voltages exceeding 500 V, it is important not only to exercise caution to protect yourself from injury, but also to make sure your test equipment can handle the voltages that may be encountered.

Some handheld and benchtop oscilloscopes have maximum AC and DC input voltages of 400 V, making them subject to damage if used to troubleshoot

hollow-state amplifiers. This issue can be overcome by the use of a 10:1 probe (see **Photo 1**).

Potential Preamplifier Malfunctions

Figure 1 shows the schematic of a two-stage preamplifier for a magnetic phono cartridge. Naturally, if such a preamp malfunctions, we begin with the easiest step: replacing the tube. If that doesn't work, we look under the chassis. In this article, we discuss using an oscilloscope to isolate the problem. The likely causes of malfunction in this simple circuit include an open or shorted filter capacitor; an open, leaky, or shorted coupling or cathode-bypass capacitor; or a resistor that has changed value.

Figure 2 shows the input and the output waves for the first triode section of the 7025 tube, using a 0.21-mV_p sine-wave input. It is an essentially perfect sine wave. The amplitudes show a voltage gain of about 2.8 for this stage including the loss in the equalization network. The traces are not in phase, because of the equalization network made up of R6, R10, C6, C7, and C8.

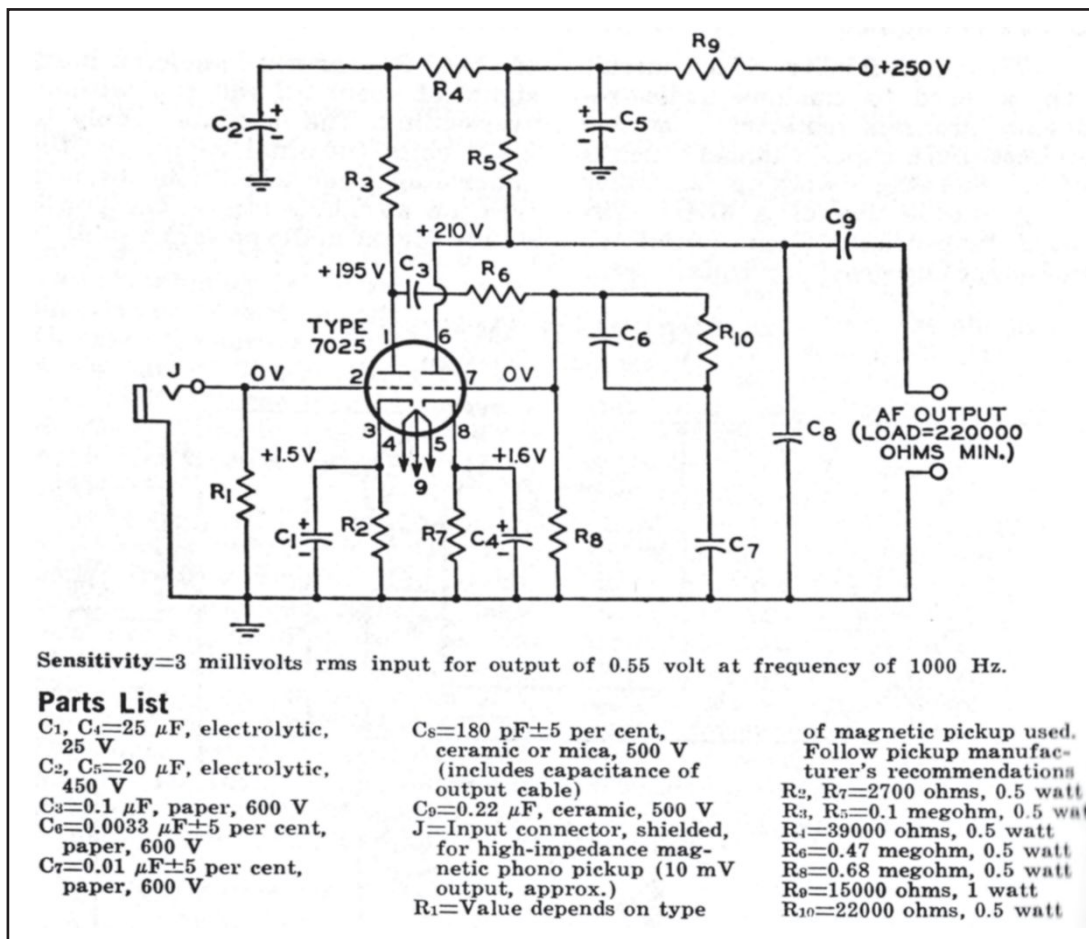


Figure 1: This phono preamp circuit appeared in the 1975 RCA Receiving Tube Manual.

Figure 3 shows the signal at the cathode of the first triode section. The cathode voltage is about 1.75 VDC (vertical center of the of the sine wave), indicating a quiescent plate current of about 0.65 mA in that stage. We usually pretend the cathode bypass capacitor C1 prevents the appearance of any AC at the cathode, but since the reactance of a 25- μ F capacitor at 1 kHz is about 6.4 Ω rather than exactly zero, we see about a 1.8- μ V sine wave at the cathode.

As shown in **Figure 4**, the signal at the preamp output will look the same as the one from the plate of the first triode section, except that the output has an amplitude of about 28 mV_p. The plate voltage of the second triode section is 185 VDC.

Figures 2–4 show the normal waveforms for the preamp, when driven by a signal of the proper amplitude. If coupling capacitor C3 were to open, the oscilloscope would show no signal at the preamp output, although there would be a normal signal at the plate of the first triode stage.

If the coupling capacitor shorts or becomes leaky, we might simplistically expect the output to become very distorted because of the second triode's grid being driven positive. However, driving the grid very positive only increases the plate current because the negative DC feedback provided by the cathode resistor prevents the grid-to-cathode voltage from actually going positive and causing clipping.

Instead, what happens is that grid-current

limiting reduces the stage gain while adding little or no significant distortion. This effect is shown in **Figure 5**. Notice that although the waveform is not distorted, the amplitude is now only about 120 nV_p. Under these conditions, the cathode voltage will rise to about 6.7 VDC.

If the equalization capacitors C6–C8 should open, the frequency response would no longer conform to the RIAA curve. If any of these capacitors were to become shorted, the output amplitude would decrease—or in the case of C8, the output would disappear entirely.

A shorted cathode capacitor will change the triode

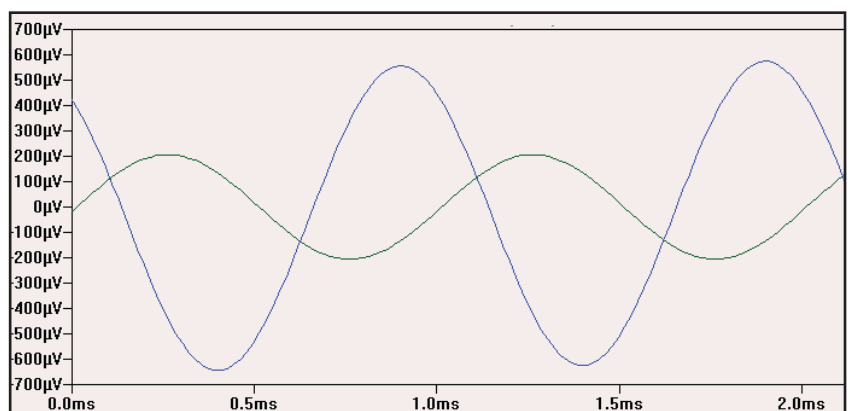


Figure 2: The green trace is the input to the first triode. The blue trace is the output from the plate of the first triode, after coupling capacitor C3.

stage from cathode-biased to grid-leak bias mode. Since in both stages the resistor from grid to ground is relatively small, the grid will be biased near 0 V. With the small-amplitude signals encountered in this circuit, we will not see noticeable output distortion on the oscilloscope, but the gain of each stage will increase by about 10% due to the absence of negative feedback at the cathode. It is conceivable that the

distortion would increase enough to be noticeable for a careful listener. The oscilloscopes will show no signals and 0 VDC at the cathode. An open cathode capacitor will just decrease the gain; it will not affect the output waveform.

Potential Problems with Resistors

As mentioned earlier, the other passive components that can cause problems are resistors. In most cases where this occurs, the resistors are carbon-composition types. **Photo 2** and **Photo 3** will help you to visually distinguish among the common types of resistors.

When carbon composition resistors are subjected to heat for a long time, they can change value, usually increasing. If plate resistor R3 in our preamp circuit were to increase from 100 k Ω to 500 k Ω , the signal at the plate of the first triode would be as shown in **Figure 6**. The amplitude has increased somewhat compared to the normal condition, and the DC plate voltage has dropped from the normal 185 V to about 80 V. A similar effect would result if the second-stage plate resistor R5 increased significantly. In both cases, if the signal amplitudes were large enough, increases in plate resistor values could result in distortion, mainly because of the shift in DC plate voltage. The increase in gain caused by a high-value plate resistor

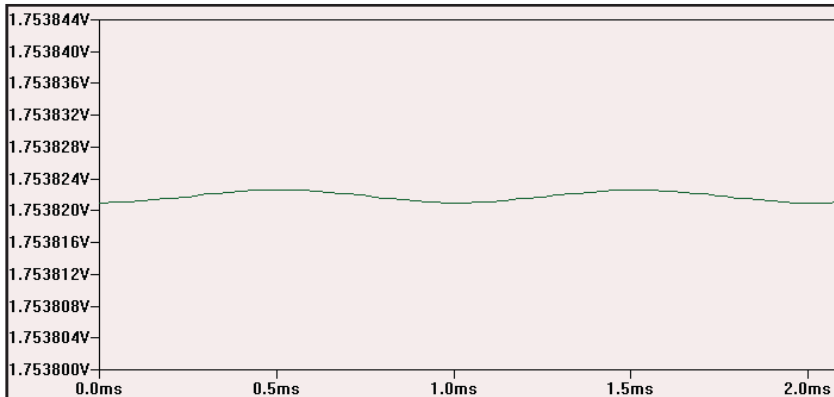


Figure 3: The waveform at the cathode shows about 1.75 VDC and very little AC.

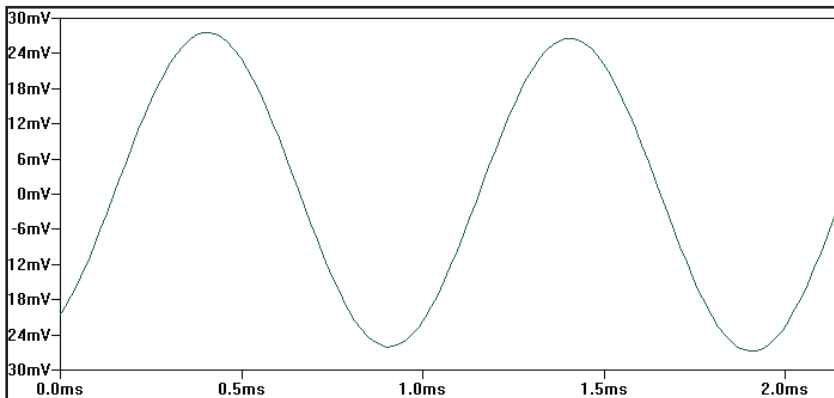


Figure 4: The normal output from the preamp is a sine wave with an amplitude of about 28 mV_p.

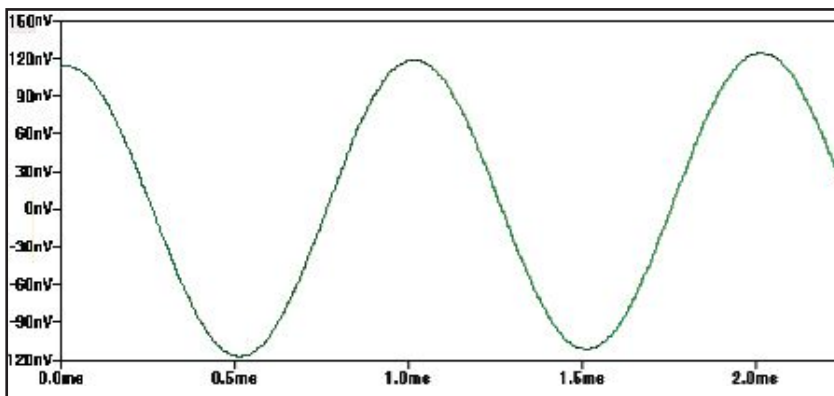


Figure 5: A shorted or leaky coupling capacitor may only cause decreased output amplitude.



Photo 2: A carbon composition resistor can be recognized by its near-perfect cylindrical shape.

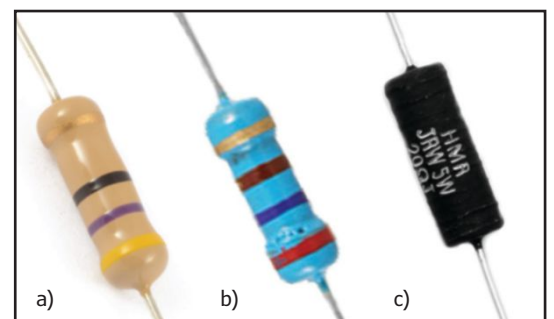


Photo 3: Carbon film (a), metal film (b) and wire-wound (c) resistors are not as perfectly cylindrical as carbon-composition resistors are.

is limited by the fact that the gain depends upon the parallel resistance of the plate resistor and the resistance of the load being driven, making gain vary by a smaller percentage than that by which the plate resistor's value changes.

Hum in the output signal is one of the most common complaints related to phono preamplifiers, due to the small input signal amplitudes involved. **Figure 7** shows a 60-Hz hum wave with a 1-kHz signal riding on it. Note that the hum wave is sinusoidal, which indicates that it probably comes from electrical or magnetic induction or improper grounding. If the hum were caused by poor power-supply filtering, the hum wave would be a series of positive half-cycles at a frequency of 120 Hz (a full-wave rectifier) or 60 Hz (a half-wave rectifier).

The top waveform of **Figure 8** shows the buzz wave produced by a power supply with an open filter capacitor. The bottom waveform shows what would happen if the filter capacitor had decreased in value and/or increased in equivalent series resistance. (In areas with 50-Hz power, the frequencies would be 100 Hz and 50 Hz, respectively, rather than 120 Hz and 60 Hz.)

Wet Electrolytics

In old hollow-state circuits (approximately 1930s-vintage or earlier), wet electrolytics were used as filter capacitors. These capacitors were sealed by a natural rubber grommet around the terminals. With age and exposure to heat, this grommet could dry out, enabling the electrolyte to leak out. The capacitor would then drop in value from perhaps 20 μF to a few hundred picofarads, leaving the DC practically unfiltered.

In later "dry" electrolytics, the electrolyte is a paste, and better seals were used, resulting in longer life and less tendency for power-supply-related buzz in audio circuits. Even dry electrolytics are subject to aging effects, though, and can decrease in value while increasing in equivalent series resistance (ESR), causing increased power-supply hum, though not to the extent caused by failed wet electrolytics.

As we have seen, an oscilloscope can allow both waveform monitoring and DC voltage measurement, and coupled with an understanding of hollow-state audio circuit principles, can be valuable when you have to chase down a circuit malfunction. 🛠️

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.

Resistor Power

Most electronic equipment built before 1960 used carbon-composition resistors. These resistors were made of a mixture of carbon powder and clay, and over time, they can change value, particularly with exposure to high temperatures. Usually, the value increases rather than decreasing when this happens. Most newer equipment uses carbon-film or metal-film resistors. Carbon-film units are much less likely to change value, and metal-film ones almost never do. But carbon-composition resistors did have the ability to withstand power surges of about five times the rated power without catastrophic damage. Only wire-wound resistors are better in this regard, being able to handle surges of about 10 times the rated power. Carbon-film and metal-film resistors can handle surges of approximately two to three times the rated power before being destroyed.

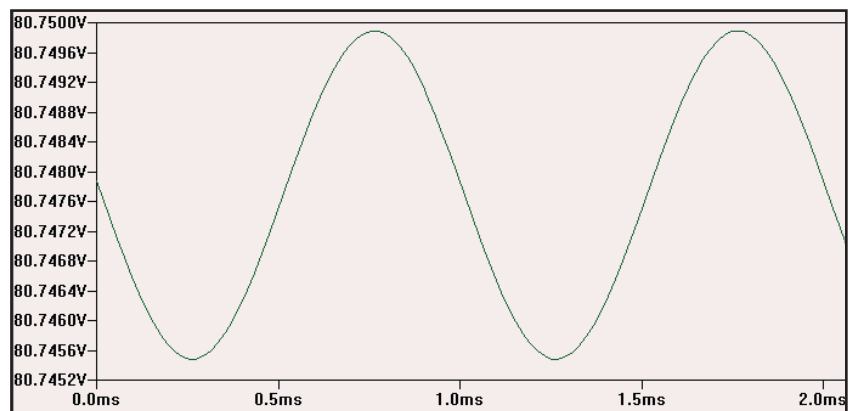


Figure 6: A plate resistor that has increased in value will result in increased gain and lower DC plate voltage.

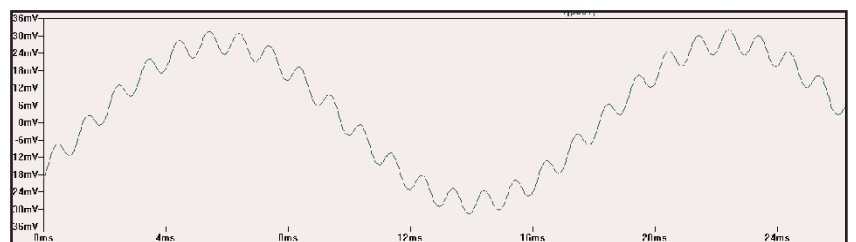


Figure 7: When hum is a problem, the oscilloscope trace will show the high-frequency signal riding on the low-frequency hum wave.

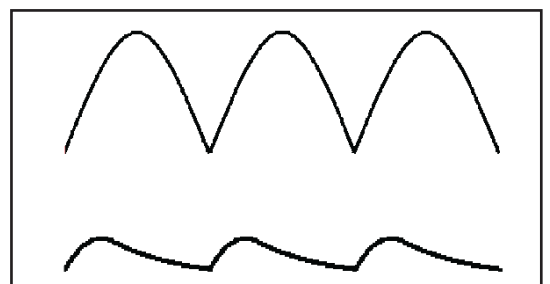


Figure 8: Buzz caused by poor power-supply filtering will cause the high-frequency signal wave to ride on low-frequency waves such as these.

Resources

R. Honeycutt, "Troubleshooting the Tough Ones," *audioXpress*, October 2015.

—, "Troubleshooting Hollow-State Equipment," *audioXpress*, August 2015.

RCA Tube Manual, Radio Corporation of America, 1975.



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

February 7–10, 2017

Integrated Systems Europe (ISE)

Amsterdam RAI, The Netherlands
www.iseurope.org

Integrated Systems Europe is a joint venture of the Custom Electronic Design and Installation Association (CEDIA) and InfoComm International. The 2016 edition was officially larger than the US InfoComm and CEDIA shows combined and ISE's co-owners both reported their huge satisfaction with the high levels of interest in their onsite educational programs and sessions from attendees, attracting record attendance numbers. The four ISE Show Floor Theaters, dedicated to Residential, Commercial, Unified Communications, and Audio Solutions, has attracted more than 3,000 visitors to over 150 technology and case study presentations given by manufacturers, consultants, and end-users.

The 2017 edition of the world's largest AV and systems integration show to take place at the RAI Amsterdam Exhibition Centre follows a series of structural changes to the market leading exhibition and conference venue. For 2017, the RAI Amsterdam will introduce an additional hall and pavilion along with a new hall numbering system and additional car parking facilities. The show continues to demonstrate growth on all fronts.

By the close of the ISE 2016, the net sold exhibitor floor space for the 2017 event had already exceeded 100% of that sold for the 2016 edition. Registered attendance for the first four-day show was 65,686, a rise of almost 11% on the previous year. This trend is expected to continue for ISE 2017.

More than 80 new exhibitors from 14 different countries and four continents will take their place on ISE's largest ever show floor, joining more than 1,000 returning companies from across the AV and systems integration industry. That equates to over 1,600 m² of new exhibitor floor space.

**Integrated
Systems
Europe**

7-10 February 2017
Amsterdam, RAI, NL

showcase for key international audio brands (which has expanded from one hall in 2013 to four halls in 2016 and 2017) now is complemented with the Enping Microphone Hall, which will contain product zones for microphones, conference systems, and public address systems.

Prolight+Sound Guangzhou is organized by Messe Frankfurt and the Guangdong International Science and Technology Exhibition Company (STE).

February 27–March 2, 2017

207 GSMA Mobile World Congress

Fira Gran Via
Barcelona, Spain
www.mobileworldcongress.com



The Mobile World Congress will be held February 27 to March 2, 2017, at Fira Gran Via in Barcelona. There will also be some events taking place at Fira Montjuïc. Even though this is mainly the world stage for mobile operators and smartphone manufacturers, the show is increasingly a launch pad for the latest wireless audio technologies, IoT, and wearables.

The GSMA represents the interests of mobile operators worldwide, uniting nearly 800 operators with almost 300 companies in the broader mobile ecosystem, including handset and device makers, software companies, equipment providers and Internet companies, as well as organizations in adjacent industry sectors. Now located in Hall 4, the GSMA Innovation City will feature more innovative mobile products and services than ever before, allowing visitors to explore and connect with the latest innovation in areas such as 5G, the Internet of Things, security and identity. Hall 8.0 NEXTech showcases forward-thinking companies and disruptive technologies, from virtual reality/augmented reality and robotics, to cognitive computing, artificial intelligence and more. Building on the success of the Graphene Pavilion in 2016, the Graphene Flagship, a consortium of over 150 academic and industrial research groups in 23 countries, will return to Mobile World Congress with the Graphene Experience Zone.

February 22–25, 2017

Prolight+Sound Guangzhou
China Import & Export Fair Complex, Guangzhou, China
www.messefrankfurt.com
www.prolightsound-guangzhou.com

prolight+sound
GUANGZHOU

Prolight+Sound Guangzhou, the largest trade fair for the pro audio and lighting industry in China, will stage a full array of top-of-the-line products plus industry-focused events within 13 halls (130,000 m²) plus the entire outdoor space in Area A of the China Import & Export Fair Complex in Guangzhou.

Celebrating its 15th anniversary this year, Prolight+Sound Guangzhou 2017 will welcome approximately 1,250 local and international exhibitors, including some of the most notable names from the pro audio, lighting, and entertainment technology sectors. Starting in 2003, Prolight+Sound Guangzhou has successfully transformed from a regional to an international business platform for the pro audio and lighting industry through the strategic partnership formed between Messe Frankfurt and the Guangdong International Science and Technology Exhibition Company (STE) four years ago.

The event will continue to provide an all-encompassing marketplace with specific product zones for conference systems, PA systems, headphones, microphones, and accessories, as well as the signature Audio Brand Name Halls. The expanded outdoor line array across three demonstration areas will feature the largest number of local and overseas audio brands to provide a wider variety of sourcing options for attendees. The signature Audio Brand Name Halls—a special

March 14–17, 2017

electronica China 16th International Trade Fair for Electronic Components, Systems, and Applications

China Import & Export Fair Complex, Guangzhou, China
Shanghai New International Expo Centre, China
<http://electronica-china.com>



electronica China 2016 continues to be the premier showcase of electronics innovation in Asia, offering a multi-faceted experience to the visitors. It gets bigger and better every year. Founded in 2002, electronica China is now one of the leading platforms for electronic components, systems and applications.

electronica China takes place in conjunction with productronica China (dispensing, EMS, materials, SMT, test and measurement, automation, and component manufacturing), Photonics China and SEMICON at the Shanghai New International Expo Centre. The combination of these shows and market segments creates attractive synergies and added value for visitors, covering the electronics industry's entire value chain.

electronica China, is part of the electronica network—including the parent trade fair electronica Munich, held at Messe München every two years in November, and electronica India, which has been held in India every year in September since 2000. In 2016, electronica China combined 62,000 m² of exhibition space (gross), attracting 61,455 trade visitors and 1,088 exhibitors from 23 countries.

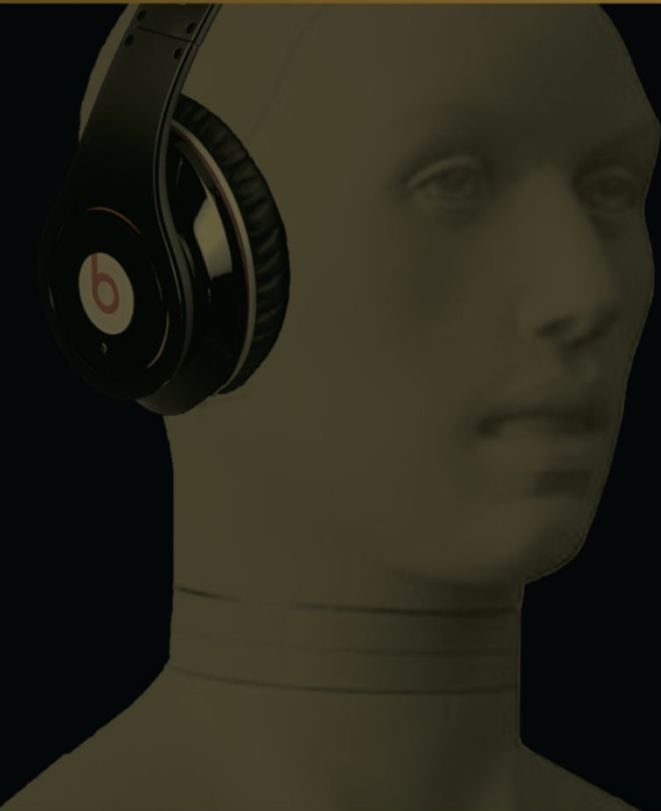
Engineering Consulting for the Audio Community

Celebrating
34 YEARS
1983-2017

MENLO SCIENTIFIC



- Liaison Between Materials, Suppliers, & Audio Manufacturers
- Technical Support for Headset, Mic, Speaker & Manufacturers
- Product Development
- Test & Measurement Instrumentation
- Sourcing Guidance



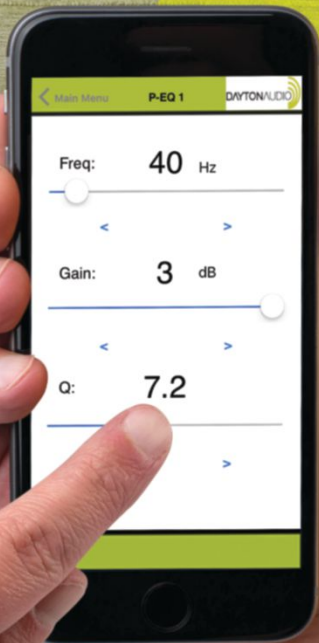
MENLO
SCIENTIFIC

USA • tel 510.758.9014 • Michael Klasco • info@menloscientific.com • www.menloscientific.com



TAKE CONTROL

With the Dayton Audio SPA2400DSP 2400W Subwoofer Amplifier with DSP, you have the power of 2400 Watts RMS through two 1200W RMS amplifier modules. You have the flexibility to use this plate amp to control two subwoofers or a portable PA system using the high pass outputs in conjunction with a powered sub. Shape the sound with the built in DSP by adjusting PEQs, HPF, LPF, Phase, Sub Tuning and Mains Delay according to your needs and tastes. Use the included microphone with Auto Room EQ function to get the best sound no matter where the system is placed. You can adjust various parameters using the display selection knob, laptop or phone app. You're in complete control with the SPA2400DSP amp!



Room EQ



PEQ/Limiter



Distributed By:

PARTS
EXPRESS
YOUR ELECTRONICS CONNECTION

Tel: 800-338-0531
725 Pleasant Valley Dr.
Springboro, OH 45066

Visit parts-express.com to learn more about the Dayton Audio SPA2400DSP amplifier and **save \$5 off \$75 order!** Use promo code: AXJ17P
Offer expires 2/28/2017