INNOVATIONS IN AUDIO • AUDIO ELECTRONICS • THE BEST IN DIY AUDIO

www.audioxpress.com

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Fresh From The Bench

QuantAsylum QA401/QA451 Modular Audio Analyzer

By David Logvin

Practical Test & Measurement

Mastering Wireless Multi-Tone Testing

By Andrew Taylor (Klippel)

Speaker Builder

A Tribute to Jeff Bagby

By Ben Shaffer

Fresh From the Bench

Puffin Phono DSP by Parks Audio

By Mike Harkins

You Can DIY!

Build a Passive Moving-Coil Step-Up Device

By Gary Galo

R&D Stories

Getting Started with A²B —Designing in A²B

By Brewster LaMacchia

Hollow-State Electronics

Circuit Designs for Tube Mixers

By Richard Honeycutt

Audio Scope

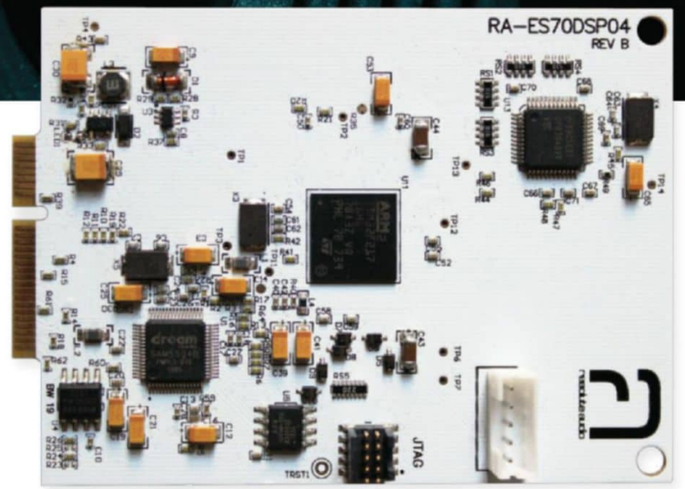
Simple Improvements to Legacy Differential Transimpedance for Single-Ended Audio DAC Output Designs

By Michael Steffes

DECEMBER 2020

AES70

network-controlled DSP module



AES70 lays the foundation for a multitude of new control apps for monitoring, automation and workflow improvements, and expands the reach of current apps, thereby adding value to AES70 compliant hardware products.

Resolute's **RA-ES70DSP04 DSP module** is easily programmable, so you can add AES70 networked control and monitoring to your next generation product, be that a powered speaker, microphone pre-amp, power amp or small mixer.

The algorithms available for the RA-ES70DSP04 include multiband EQ, Crossover Filters, Vocal Reverb, Compressor, Noise Gate, Spring Reverb and many more. An onboard web server provides rich controls for each algorithm.

resolute audio

You'll find all Resolute products and other audio parts at profusionplc.com

DSP Design Tools

The RA-ES70DSP04 comes with all the software tools to help you program the DSP and implement AES70. In addition, there is an evaluation board that includes 4 channels of analog audio in and out, and 2 channels of AES67 audio in and out, ideal for developing networked audio products.

Profusion
Audio semiconductors

We do audio. Talk to us.

profusionplc.com/aes70

HAMMOND MANUFACTURING®

1140 Series

Studio-Grade Impedance Matching Transformers

Rooted In Solid ENGINEERING

- **Microphone Input**
- **Microphone Splitter**
- **Line Input**
- **Line Output**
- **Direct Box**

- Designed with Nickel alloy cores for high performance and Wide Bandwidth
- Frequency Response: 20 Hz – 20 kHz ±0.35 dB, (ref. @ 1 kHz)
- Low distortion at levels up to +27 dBu @ 20 Hz
- Excellent Shielding – Mu Metal enclosures
- Line Output Transformers are open style (not Mu Metal)

www.hammondmfg.com

December 2020 **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., P.O. Box 417, Chase City, VA 23924, US Periodical Postage paid at Chase City, VA, and additional offices.

Head Office:
KCK Media Corp.
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
Fax: 888-980-1303

Subscription Management:
audioXpress
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
E-mail: custserv@audioxpress.com
Internet: www.audioxpress.com

Postmaster: Send address changes to:
audioXpress
P.O. Box 417
Chase City, VA 23924, 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
P.O. Box 417
Chase City, VA 23924, 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. 2020
Printed in the US

For Future Memory

With this December 2020 edition we are proud to conclude another year of progress in helping to construct a better publication, more relevant to its readership. And in the constant search for the most relevant topics, we have been looking at industry announcements and developments in the ever more critical area of wireless audio. But either we did too good a job last year or there isn't much more to report for now, especially considering that we didn't have industry events and there were no major product innovations in the category this year. There were some announcements, particularly early in the year at CES 2020. But nothing has resulted as yet.

For me, one of the major disappointments (maybe I'm too impatient with technology) is the fact that there weren't significant developments regarding the introduction of Bluetooth Low Energy (LE) Audio. Recently, the Bluetooth Special Interest Group (SIG) announced it will start qualifying products using the LC3 Low Complexity Communication Codec, part of the new LE Audio Specification announced at CES, back in January (and which we covered in our April 2020 edition). But the organization also confirmed that it doesn't expect the rest of the specifications that define all the use cases for LE Audio to be released before mid-year 2021. For now, it is just encouraging manufacturers to start implementing the new Fraunhofer/Ericsson codec, and we are still waiting to hear about the many possible uses of Multi-Stream Synchronized Audio, Broadcast Audio, or Bluetooth Audio Sharing.

And while licensing efforts already indicate that there is wide industry support for LC3 and LE Audio, the problem now seems to be that we just don't know when products supporting the next generation of Bluetooth Audio will start showing up. As the new global pandemic conditions dictate, products will simply "drop" when we least expect it.

With its major adoption in personal audio and mobile categories, Bluetooth is already dominating consumer electronics and the audio industry. Bluetooth is even being offered in classes of products that absolutely don't need it, such as turntables. And most using the "old" Bluetooth 4 specification and SBC codecs...

The pandemic and the increasing time spent at home has also paved the way for growth and opportunities in home audio. And naturally, with more people from the same household sharing the space for more time, renewed pressure has been placed on existing technologies, including wireless networks, as audio is also being streamed over Wi-Fi (IEEE 802.11) within the home. The problem is, wireless audio streaming over Wi-Fi remains the realm of just a few companies, some of which continue to fight each other in the courts over the multi-room protocols—which naturally deters further competition. And while that happens, even with Wi-Fi 6 seeing one of the fastest adoption rates ever for the technology (according to the Wi-Fi Alliance), the audio industry is not doing much to improve its side of things.

That's where I feel we will have more opportunities to publish an interesting industry update... next year. For now, there's just too much uncertainty. As an example, in March 2020, the Wireless Speaker and Audio Association (WiSA - www.wisaassociation.org), made the crucial announcement that it would be expanding support for Dolby Atmos. More than 60 leading consumer electronics brands are currently shipping WiSA products supporting 5.1 and 7.1 surround configurations—which are great to have with the convenience of wireless, but are not exactly game-changing.

Supporting immersive audio, and specifically Dolby Atmos height channels, will enable WiSA speaker partners to develop products with increased mass appeal. Making it easy for consumers to configure wireless systems for Dolby Atmos content and enjoy immersive 3D audio experiences with WiSA Certified speakers can take the technology to a whole new level. Unfortunately, at the time of writing, this firmware update has not yet materialized.

It seems like leaving 2020 behind is the only good thing we can look for. Not only did things not progress the way they could have, all over the world we mourn the losses of precious lives, including many industry friends who have left unfinished work that will never be replaced.

In this edition, we are honored to pay tribute to the life of someone who positively inspired so many of us. Jeffery Allen Bagby, 60, of Kokomo, IN, sadly passed away March 24, 2020. The entire audio community realized a deep sense of loss and sadness for the disappearance of one of its friendliest voices, one of the most enthusiastic speaker builders and a prolific contributor to the global DIY audio community. To highlight the significance of Jeff Bagby's contributions, we asked one of his lifelong friends, Ben Shaffer, to look back on his work and help us preserve his memory.

J. 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, Chuck Hansen, Richard Honeycutt, Charlie Hughes, Mike Klasco/Nora Wong, David Logvin, Oliver Masciarotte, Nelson Pass, Christopher Paul, Bill Reeve, Fernando Rodrigues, Michael Steffes, Stuart Yaniger

OUR NETWORK

audioxpress · circuit cellar · LIS Loudspeaker Industry Sourcebook · Voice Coil

SUPPORTING COMPANIES

ACO Pacific, Inc.	47	KAB Electro-Acoustics	57
Acoustic Sciences Corp.	57	Menlo Scientific, Ltd.	25
All Electronics Corp.	41	Oxford Digital, Ltd.	33
Clockworks Signal Processing, LLC	53	Parts Express International, Inc.	68
COMSOL, Inc.	9	Profusion plc	2
GRAS Sound & Vibration	11	QuantAsylum USA LLC	15
Hammond Manufacturing, Ltd.	3	Redco Audio	47
Hypex Electronics B.V.	13	Triad Magnetics	67

NOT A SUPPORTING COMPANY YET?
Contact Peter Wostrel (audioxpress@smmarketing.us, Phone: 978-281-7708, Fax: 978-281-7706)
to reserve your own ad space in the next issue of the magazine.

COLUMNISTS

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

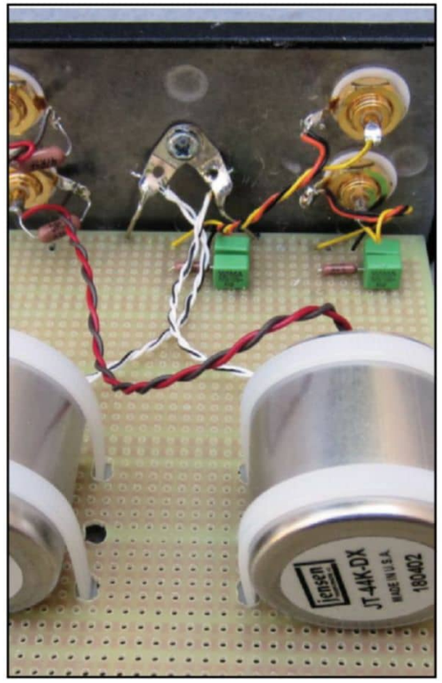
Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is a graduate from New York University, with post graduate work in signal processing, and he holds multiple patents licensed or assigned. For the past 35 years, Mike has worked on countless R&D projects for large and small companies. Mike specializes in materials and fabrication techniques to enhance audio performance.

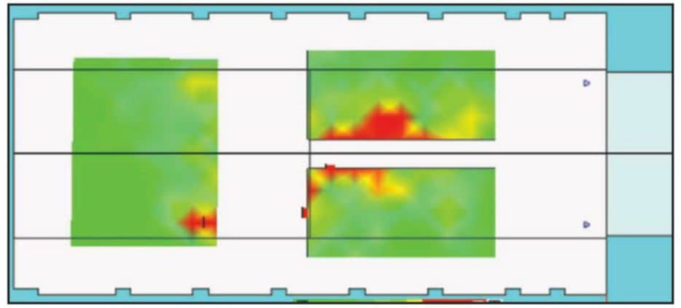
Michael Steffes has worked in high-speed signal path IC development, spanning IC design roles for the BurrBrown high-speed amplifier products, and product launch/application manager roles through Texas Instruments/Intersil. He continues to focus on high-performance ADC interface solutions, DSL/PLC line driver design and optimization, and signal path design optimization tools. In a career spanning five different major IC vendors, he has participated in the introduction of 150 industry leading amplifier products (setting many of the now standard practices in high speed amplifier characterization curves), publishing more than 100 contributed articles and application notes from 1989 onward.

STATEMENT REQUIRED BY THE ACT OF AUGUST 12, 1970, TITLE 39, UNITED STATES CODE SHOWING THE OWNERSHIP, MANAGEMENT AND COPY CIRCULATION OF audioXpress. Published monthly at 650 Dairy Farm Rd, Red Oak, VA 23964. Annual subscription price is $50.00. Publisher: KC Prescott. The owner is KCK Media Corp., 650 Dairy Farm Rd, Red Oak, VA 23964. The names and addresses of stockholders holding one percent or more of the total amount of stock are: KC Prescott, 650 Dairy Farm Rd., Red Oak VA 23964. EXTENT AND NATURE OF CIRCULATION: Average number of printed copies of each issue published during the preceding twelve months *For each single issue nearest to filing date; total number of copies printed, 1000 *1254; paid/requested mail print subscriptions, 982 *972; sales through dealers and carriers, street vendors and counter sales, 100 *100; paid/requested copies distributed by other mail classes, 23 *24; total paid/requested print circulation, 982 *972; Nonrequested copies distributed outside the mail, 234 *0; total nonrequested distribution (sum of D.1 & D.4), 234 *0; total distribution (sum of C & E), 1776 *2862; copies not distributed (office use, leftover, unaccounted, spoiled after printing, returns from news agents), 24 *28; total (sum of F & G), 1254 *1000. Percent Paid Requested: 80.75% *100%; total number of paid Electronic Copies, 1043 *1374; Total Paid Print & Paid Electronic Copies 2025 *2346; Total Print Distributions and Paid Electronic Copies 2259 *2346; Percent Paid: 89.64% *100% I certify that the statements made by me above are correct and complete. KC Prescott Publisher.

Volume 51 – No. 12 December 2020

audioxpress





Features

8 Remembering Jeff Bagby
By Ben Shaffer
Jeffery Allen Bagby passed away in March of this year. On that day, the audio community lost one of its most prolific contributors to the global DIY audio community. One of Jeff Bagby's lifelong friends, Ben Shaffer, reflects on his work.

26 QuantAsylum QA401/QA451
Redefining the Compact and Modular Audio Measurement System
By David Logvin
David Logvin explores a valuable solution for audio measurements. The QuantAsylum QA401 audio analyzer is a complete standalone system of hardware and software that keeps evolving and growing in popularity, now complemented with the completely new QuantAsylum QA451 programmable load—so new that Logvin tested and even helped debug a pre-production version.

34 Mastering Wireless Multi-Tone Testing
By Andrew Taylor
This article explains the issues resulting from measuring a wireless audio connection and how wireless multi-tone testing can help us to overcome those obstacles.

38 The Puffin Phono DSP by Parks Audio
By Mike Harkins
In this product review article, Mike Harkins puts the Puffin, a next-generation phono preamp, under the microscope. The design adds a programmable DSP and truly takes the concept of a phono stage to a different level.

48 Build a Passive Moving-Coil Step-Up Device
By Gary Galo
After reviewing other products using Jensen transformers, Gary Galo decided to build his own step-up device and now shares his experience in this project article.

54 Getting Started with Automotive Audio Bus (Part 3)
Designing in A²B
By Brewster LaMacchia
The first two parts of this article examined the overall A²B system operation and how its features can enable new types of digital audio applications. This part takes a deeper look at A²B data transport and Analog Device Inc.'s AD242x series of parts.

Columns

SOUND CONTROL
16 Acoustical Design of a Multi-Purpose Room
By Richard Honeycutt

AUDIO SCOPE
22 Simple Improvements to Legacy Differential Transimpedance to Single-Ended Audio DAC Output Designs
By Michael Steffes

HOLLOW-STATE ELECTRONICS
59 Circuit Designs for a Hollow-State Mixer
The Development of Recording Mixers
By Richard Honeycutt

ARTICLE UPDATE
64 Comparing IMD (and THD) Measurements
Following Michel Nieuwenhuizen's two-part article that was published in *audioXpress*' October 2020 and November 2020 issues, we received a letter from Burkhard Vogel with a question that motivated a reply from the author.

Departments

4 From the Editor's Desk

5 Client Index

Websites

www.audioxpress.com
www.voicecoilmagazine.com
www.cc-webshop.com
www.loudspeakerindustrysourcebook.com
www.circuitcellar.com
www.linuxgizmos.com

@audioxp_editor audioxpresscommunity
linkedin.com/company/audioxpress

Remembering Jeff Bagby

By
Ben Shaffer
(United States)
Screenshots snapped directly from the software by Jeff Bagby

Jeffery Allen Bagby, 60, of Kokomo, IN, sadly passed away on March 24, 2020. On that day, the audio community lost one of its friendliest voices, one of its most enthusiastic speaker builders, and a prolific contributor to the global DIY audio community. To highlight the significance of Jeff Bagby's contributions, we asked one of his lifelong friends, Ben Shaffer, to reflect on his work.

I knew Jeff for almost 20 years and he will be sorely missed. His audio legacy will, however, live on indefinitely. His white papers on how to measure or optimize the frequency response and interaction of drivers is second to none and can be found all over the web. There is no singular textbook that equates the black art of crossover design as a learnable and complete solution. However, I am confident that Jeff's complete generated library of information could teach a student all the required details.

For Jeff, it started back in the 1970s with the spatial capabilities of a Quatrovox system, and how the system could move the image around the room. He was fascinated that the dial on the front panel was able to do this. After blowing a set of tweeters in the 1980s, and a subsequent trip to RadioShack for replacements, he was surprised at how they just didn't sound the same anymore. Being who Jeff was, he then dove deep into learning all he could to solve the problems and questions that he had from this occurrence. Jeff was a "problem solver" (his words, not mine), both in career and the hobby that came from his love of music and its reproduction.

Jeff was first and foremost a believer of God, and everything he did was based on those beliefs. He also believed that science and religion did not have to exist in separate spaces. Research was the scientific portion while having faith and diligence he'd find the answer.

Jeff is survived by his wife Chris and son David. He worked for Chrysler in Kokomo, for the majority of his adult life. When my wife was driving a Jeep, he had told me he designed a lot of the transmission assembly for that particular model. He told me when Chrysler had a problem at their plant to solve, he was the guy they would trust. He had a college degree, but not in Engineering although he worked and designed heavily in both career and loudspeaker design.

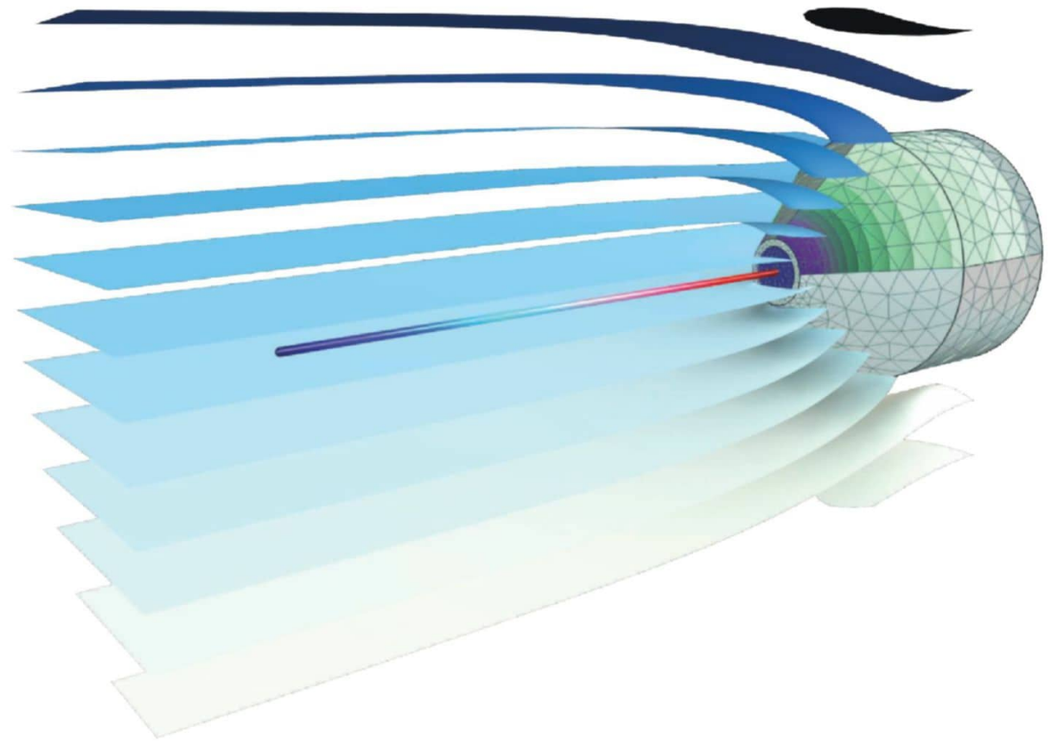
Jeff became friends with Joe D'Appolito, who answered a lot of the questions he had by phone, as this was long before the Internet came around. He read the books that were available, and taught himself as he experimented. When the Internet really got going in the early to mid 1990s, he also became friends with Bill Waslo, Paul Verdone, John Kreskovsky, and eventually, Siegfried Linkwitz via the old Madisound BBS Internet forum board. I also met Jeff electronically via Madisound BBS around 1998, and we met in-person for the first time about 2000, which really kicked off my own audio journey. Jeff was always supportive, and a great mentor.

Without the Madisound BBS board, we would not have what we know today as .frd or .zma files. Jeff, Paul, and Bill really helped culminate these results, so that they could form the FRD Consortium website with the available free tools of the time. Paul taught Jeff about programming, so that Jeff could make the suite of tools he eventually developed within Microsoft Excel.

Audio Contributions

Now, at this time, the programs available to

Hearing aids can't solve the cocktail party problem...yet.



Visualization of the total acoustic pressure field around and inside an elastic probe tube extension attached to a microphone case.

Many people are able to naturally solve the cocktail party problem without thinking much about it. Hearing aids are not there yet. Understanding how the human brain processes sound in loud environments can lead to advancements in hearing aid designs.

The COMSOL Multiphysics® software is used for simulating designs, devices, and processes in all fields of engineering, manufacturing, and scientific research. See how you can apply it to hearing aid design.

comsol.blog/cocktail-party-problem

COMSOL

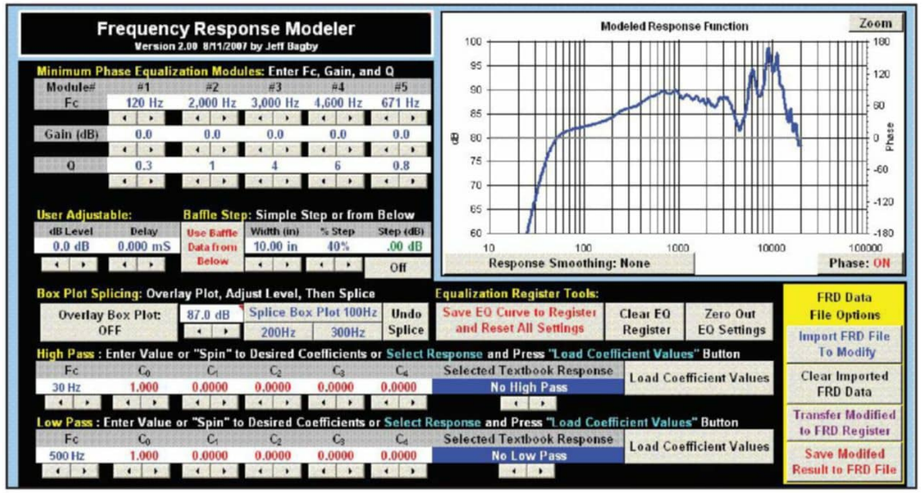


Figure 1: This a screenshot of Jeff's Frequency Response Modeler 3.0.

actually do this process of designing crossovers were terribly expensive—CALSOD, LAUD and PRAXIS (Bill Waslo), Soundeasy, and CLIO, for example. Fast forward to about 2005, and Jeff developed the programs, the Series Crossover Designer and the Passive Crossover Designer. These two programs were later combined into the second edition of the Passive Crossover Designer (PCD), and all were available for FREE! Jeff didn't ask for anything in return—he just paid it forward as others did him, and turned the world of crossover development on its ear in one fell swoop. Learning how to use the general user interface (GUI) was fairly simple, and that was also another thing that made this program what it became. It is still used by commercial entities to this day and still available for free.

Of course, measuring the drivers' properties, impedance, and frequency response was still imperative to use the PCD. You could use a free tracing program from another source to generate your files and work from spec sheets, but this was still not optimal without measuring your projects in a proper cabinet to get the best results. When Jeff dropped the Frequency Response Modeler (RM) program to fix this problem in about 2007, it gave DIYers everything they needed to do a full simulation by making the creation, adaptation, and phase extraction of the initial files easy as pie (see **Figure 1**).

You could open the program and generate both frequency and impedance plots to match the spec sheet from a manufacturer by using Q and gain adjustments. The box was then modeled and spliced to the generated plots for the woofers, all within the RM program. Then the same files could be adapted to the boxes and driver positions they would reside in by way of baffle step. Of course, saving each of these would allow the regression and revamp in another cabinet, another tuning, etc. And last, you could use the Hilbert-Bode-Transform to extract minimum phase of the final files you created.

Furthermore, the completed RM-generated files can be adapted to room scenarios within the separate Diffraction and Boundary Simulator (**Figure 2**). This comprehensive tool is also used for complex speaker to room interactions such as dipolar responses, and can provide valuable information unattainable within RM since it only

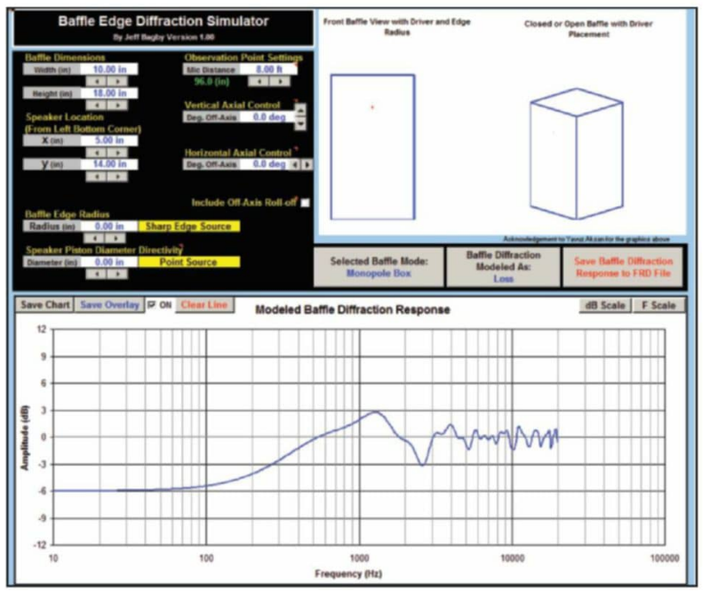


Figure 2: Here is a screenshot from Jeff's Baffle Diffraction and Boundary Simulator 1.30.

Whatever your testing challenge, GRAS has the right acoustic sensor for your audio application.

GRAS

Only GRAS offers a complete line of high-performance, test microphones and related products ideal for use in consumer audio and electronics applications.

As the leading global provider of microphones, GRAS has a long tradition of working with audio engineers to ensure accurate data is captured, each and every time.

Microphones from GRAS are designed for the high quality, durability and reliability that our customers demand.

Contact GRAS today for a free evaluation of the perfect GRAS microphone for your application.

grasacoustics.com

- Measurement microphone sets
- Microphone cartridges
- Preamplifiers
- Low-noise sensors
- High frequency ear simulators
- Head & torso simulators
- Test fixtures
- Custom designed microphones
- Speech intelligibility
- THD
- Frequency response
- Calibration systems and services

Ask about our new Hi-Res Ear Simulators! Uses a 1/4" microphone to measure up to 50kHz

Need an analyzer?
Audio precision
www.ap.com

GRAS Sound & Vibration

focuses alone on the cabinet to driver interactions.

The addition of PCD and RM is what jumpstarted my modeling of crossovers in about 2007. Before all of this existed, textbooks of ideal and improbable scenarios, and the expensive programs were all that the average audio enthusiast had at their fingertips. It was not easy to acquire the necessary items, without considerable investment and time. Since I had already done the initial reading of those books, this made the application of the process much faster and considerably easier. My real-world learning took off from there.

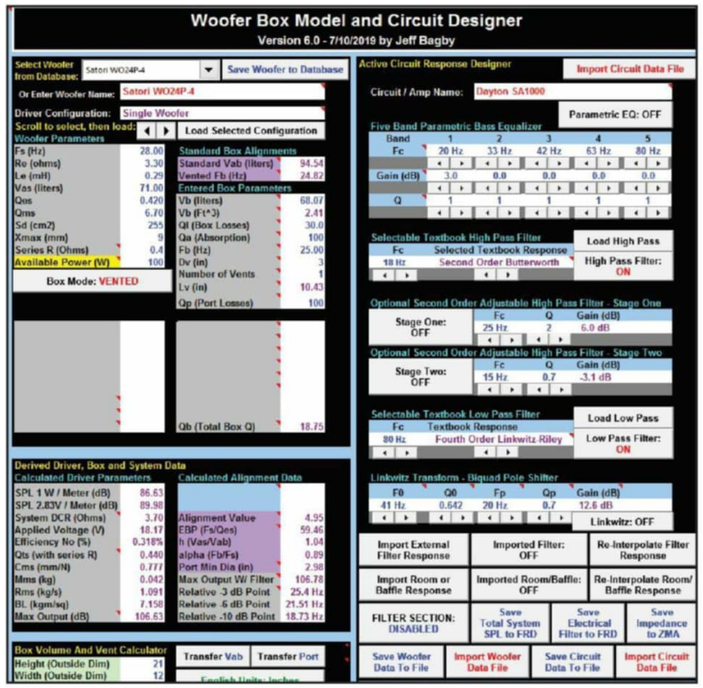


Figure 3: This is a screenshot from Jeff's Woofer Box Model and Circuit Designer 6.0, a box modeling program.

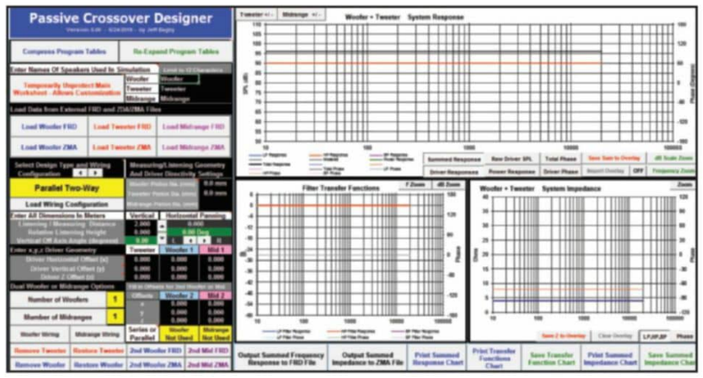


Figure 4: This screenshot shows the Passive Crossover Designer 8, which is the last iteration of PCD and was introduced in July 2019.

Next, Jeff developed the Woofer Box Model and Circuit Designer. This was Jeff's full-featured box modeling program (see **Figure 3**). It has active circuitry applications as well as the initial box model to aid design in both actively assisted subwoofers and passive towered three-ways. At the time it was introduced, it really had the full spectrum of options to get the box and results right whether using a plate amp or not. Most other programs did not include the active portion without importing the information from elsewhere. Sealed, vented, Passive Radiator (PR), and Aperiodic are all possible here.

The last program, simply called Frequency Response Blender, was a supposed better rendition of some of the attributes RM offered a few years prior. This was authored in cooperation with Charlie Laub, and the main difference is that it exhibits Charlie's user interface style. If you don't mind having this program open by itself to extract phase, then it won't be a problem for you. Since I prefer to have RM and PCD open together, I use the initial suite. The last iteration of PCD was introduced in July of 2019, version 8.0, and works with Windows 10 and Office 2016 (see **Figure 4**). It should be viable for some time to come.

When OmniMic came out, Jeff was onboard and developed methods to make the best use of this frequency response rig, and then promptly shared his findings with others so we also could make the most of it.

Jeff's Awards

What hasn't likely been said publicly is that Jeff tested his programs against some of the other programs, such as CALSOD, Unibox, for viability. Most things should be done with such acid tests to make sure they perform as accurately as necessary. When it came to accuracy, Jeff was a stickler. Jeff went through many versions of his programs before they were introduced to the public, and Jeff was always one to lead with correct information. If there was a bug in the program—he fixed it. Jeff had a lot of respect for those innovators who generated the programs before him, and knew how powerful they were. I also am certain there are other programs that Jeff never released, such as his transmission line (TL) designer, which he used to make the Solstice TL design.

A wise man doesn't reveal all of his secrets, but to quote Jeff from his acceptance speech for the Lifetime DIY Award (see **Photo 1**) in 2018:

"I didn't do what I did in a vacuum though. I learned from people…I wasn't always the little genius you see up here. I always wanted to give it back….That's why I helped everybody I could help….It's really been worth it. I have been very, very happy and fulfilled to be a part of all of this."

Photo 1: Jeff Bagby accepts the Lifetime DIY Award at InDIYana 2018.

Right before his acceptance speech, this exchange occurred:
Craig Salin: "I think you're gonna win an award."
Jeff: "Aw man, I hope not."

He was always humble and did not expect to get any rewards for his work (see **Photo 2**). As to helping people, he managed to post over 10,000 posts on one forum, the fourth member in a group of five. He also had those postings on the old Madisound board. Last, with the mere recent creation of the Facebook group "DIY Loudspeaker Project Pad" with fellow member Javad Shadzi,

Photo 2: Jeff is shown with one of his many beautiful speakers in 2018.

hypex electronics
www.hypex.nl
NEW!
NCAS500MP
400 + 100W in 8 ohm
Compatible with our MP-DSP
to create the perfect setup
for your active speaker
New family Ncore modules
NCAS500MP is the first to be released
Future members:
NCAS250MP: 200 + 50W
NCAS1000MP: 800 + 200W
For further information:
website: www.hypex.nl
mail: sales@hypex.nl
Ncore
audio power conversion
© Hypex Electronics B.V. All Rights Reserved

Photo 3: Jeff is pictured in 2015 with his award-winning Continuum.

they managed to attract and discuss this subject with more than 25,000 people worldwide.

Notable Speaker Projects

As for notable speaker projects, the Continuum (2007) really is the most well known. It has been available as a kit for a long time, and loved by many. This was his personal reference vocal monitor, patterned after the LS3/5A. He took a single entry to Midwest Audiofest (MWAF) in 2012, his Continuums and the Continuum subwoofer, and won the Unlimited category. In **Photo 3**, Jeff is pictured in 2015 with the Corian Cabinet Continuum.

Other notable Jeff Bagby designs include:

- The Kairos and Quarks projects are quasi-linear-phase designs (not pictured), and are not something cut from the average DIY builder's cloth.
- He brought the Tributes (2014, not pictured) to InDIYana with the basis of being designed with the goals of Bud Fried, and won the challenge that year. Furthermore, a company commissioned his help in designing a speaker with those goals in mind after they heard of the Tributes' success.
- The Testarossas (2014) was a floor stander that used a PR alignment to invert the room's bass curve, and used a really nice ribbon (see **Photo 4**).
- The Spirit Winds (2015) were the project he took on when he thought he might not get a chance to make another, as he was battling kidney disease (see **Photo 5**).
- The Helios (2019) also did the inverted PR alignment in a state-of-the-art stand mount speaker (see **Photo 6**). He stated in the write-up for these, that these were his best work thus far.

Photo 4: The Testarossas was a floor-standing speaker that Jeff showcased in 2014.

Photo 5: Jeff displayed The Spirit Winds in 2015 at InDIYana.

Photo 6: The Helios, built in 2019, featured an inverted PR alignment in a state-of-the-art stand-mount speaker.

Jeff lived just shy of two hours away from me. I'll never forget hearing his "Davids" (named after his son who was 4 at the time) for the first time, and being totally impressed. Jeff modeled the crossover for my first "serious" build, and I still have the notes-package he gave me for that. He took me under his wing when I was first starting out in the hobby, and I am far from the only person he helped along the way.

Remarkable, selfless, generous, humble, honest, teaching, and sincere, but most of all genuine describe this man who I was lucky to call a friend. Accolades abound, but they are truly well deserved. He didn't want fame in the hobby, but it tended to follow him around a lot.

Resources

FRD Consortium link via archive.org:
https://web.archive.org/web/20060208022032
http://www.pvconsultants.com/audio/frdgroup.htm

Charlie Laub's Audio Pages link to Jeff's Software:
http://audio.claub.net/software/jbabgy/jbagby.html

DIY Loudspeaker Project Pad:
https://www.facebook.com/groups/DIYLoudspeakerProjecPad

Video of Jeff accepting the Lifetime DIY Award in 2018, and some mentions of whom he learned from:
https://www.facebook.com/midwestaudioclub/videos/2128507043962184

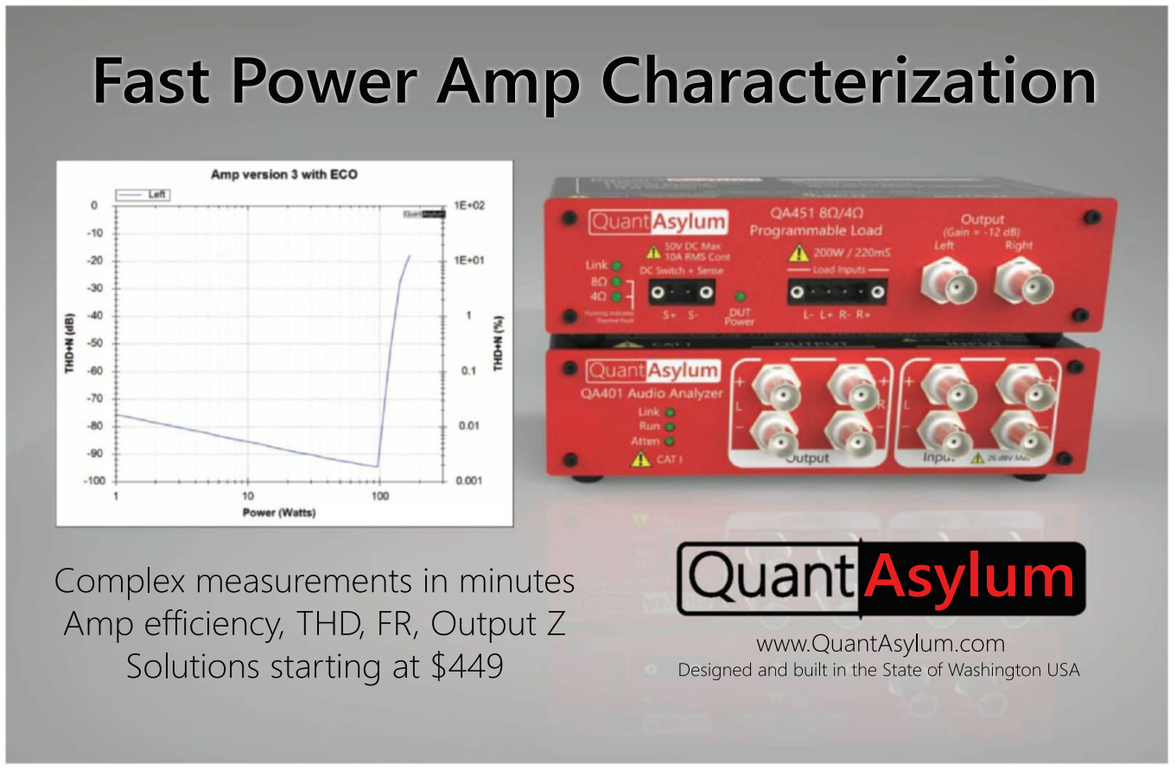
Video of Jeff talking about his Continuums:
https://www.youtube.com/watch?v=gDRAt1fcfF8

Fox article of his passing:
https://fox59.com/news/coronavirus/chrysler-engineer-who-died-from-covid-19-remembered-as-devoted-family-man-and-friend

Video of Jeff's wife Chris' interview after Jeff's passing:
https://celebritiesdeaths.com/jeff-bagby-death-dead-jeff-bagby-obituary-cause-of-death-coronavirus

About the Author

Ben (Wolf) Shaffer has been into the speaker-building hobby since circa 1997, and has amassed an immense number of speaker projects since his entry. He has a very understanding wife who puts up with him, and daughter, age 8. Vowing never to fully grow up, he uses his spare time in the pursuit of all things audio and music enjoyment.



Acoustical Design of a Multi-Purpose Room

Discover how through the use of careful acoustical design, even large spaces can provide excellent acoustics for contemporary music concerts, speeches, and social events.

Photo 1: This is a view of a multi-purpose room from house-right rear.

By
Richard Honeycutt
(United States)

Many buildings—schools, civic centers, businesses, churches—have rooms that are used for multiple purposes. These purposes may include meetings, social events with or without dining, concerts, worship services, or others. The ideal acoustics for any event include noise control, echo control, and reverberation time (RT). These parameters are perceived by attendees in terms of musical character (bass vs. treble and clarity) and speech intelligibility.

Noise, whether from outside the building, another room inside the building, many people talking at once, or HVAC systems, is a concern common to all events. So is freedom from annoying echoes. Ideal RT varies from one type of event to another.

The Multi-Purpose Room

I was recently asked to consult on the acoustical design for renovation of a church multi-purpose room, and the process presented some interesting challenges. **Photo 1** shows the multi-purpose (MP) room after the original finish materials have been removed but before reconstruction has begun. Materials in the room include: gypsum wall board (GWB) on the ceiling, soffits, and walls, wood decking on the floor, 1" fiberglass panels on the rear wall, and the house-left wall made mainly of glass.

Acoustical measurements showed that the only noise concerns would be people talking in social conditions. Of course, depending upon the acoustical treatment needed for RT and echo reasons, that noise might not be an issue.

As you can see from the plan view shown in **Figure 1**, the MP room is about 48′ × 92′, not counting the stage. This means that the round-trip time for a sound at the downstage lip to travel to the rear wall and return to the front is about 162 ms. An echo with this much delay would certainly be annoying. The cloth-covered 1" fiberglass panels on the rear wall help to reduce slap echoes. Side-to-side echoes would fall into the class known as "flutter echoes," which are not as annoying as long "slap echoes," but still should be prevented.

Preventing echoes can be achieved by adding acoustical absorption or diffusion. Absorption is less expensive, and will be needed to control reverberation, so it is best in this case to begin by choosing the appropriate treatment to obtain the proper RT.

Room Measurements

This particular MP room is to be used for speech, contemporary music, and social events. For the events planned for this room, and for the cubic volume of the room (~134,894 ft^3), a good target RT would be one second at 500 Hz. Music clarity (C80) should be above +1 dB. If there were

The "Must Have" reference for loudspeaker engineering professionals.

Home, Car, or Home Theater!

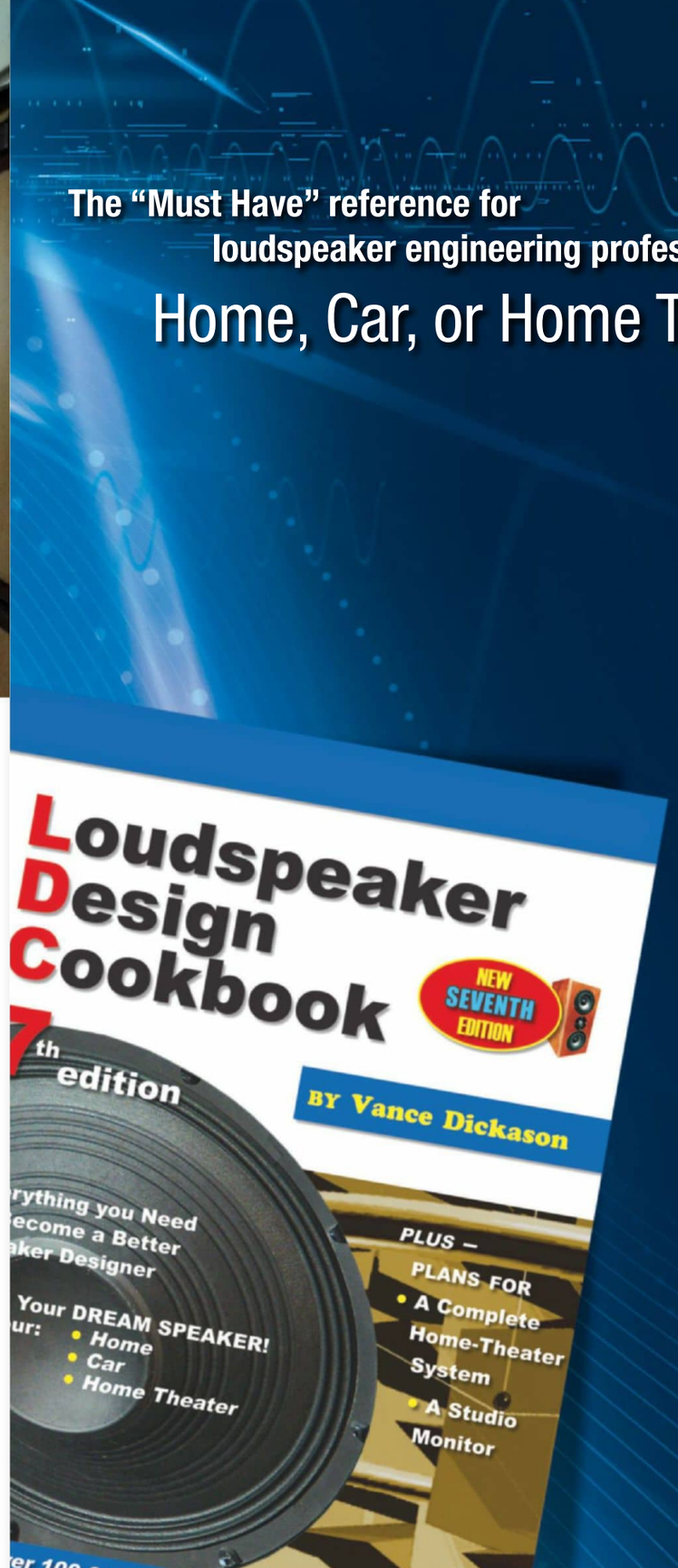
Back and better than ever, this 7th edition provides everything you need to become a better speaker designer. If you still have a 3rd, 4th, 5th or even the 6th edition of the *Loudspeaker Design Cookbook*, you are missing out on a tremendous amount of new and important information!

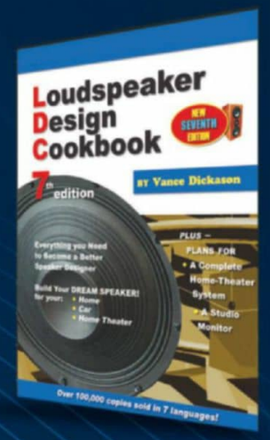
Now including: Klippel analysis of drivers, a chapter on loudspeaker voicing, advice on testing and crossover changes, and so much more! Ships complete with bonus CD containing over 100 additional figures and a full set of loudspeaker design tools.

A $99 value!

Yours today for just $39.95.

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





to be choral concerts or classical music concerts, a longer RT and lower C80 would be desirable. The RT of any room varies with frequency, because sound-absorbing materials, including human bodies, are more absorbent at higher frequencies. We have become accustomed to this frequency dependence of RT, so that a room sounds more normal to us if the RT at low frequencies is longer by 10% to 45% than at mid-frequencies (500 Hz to 1,000 Hz), and the RT at the highest frequencies is shorter by about 25% than at mid-frequencies (see **Figure 2**).

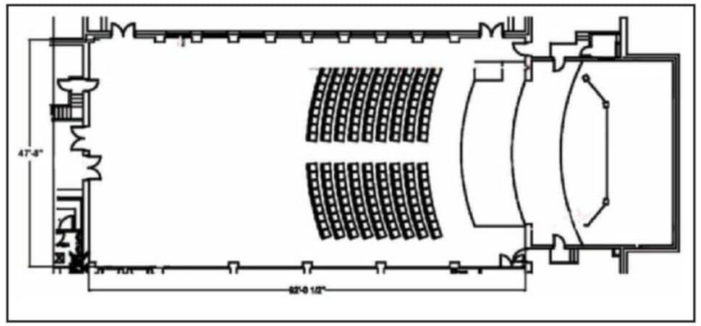


Figure 1: The plan view of the MP room indicates its size.

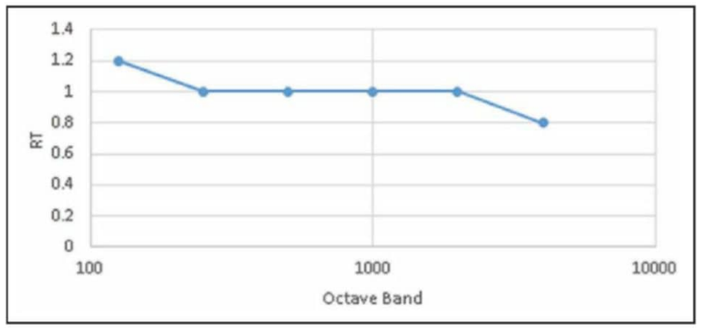


Figure 2: The target RT increases slightly at low frequencies and decreases slightly at high frequencies.

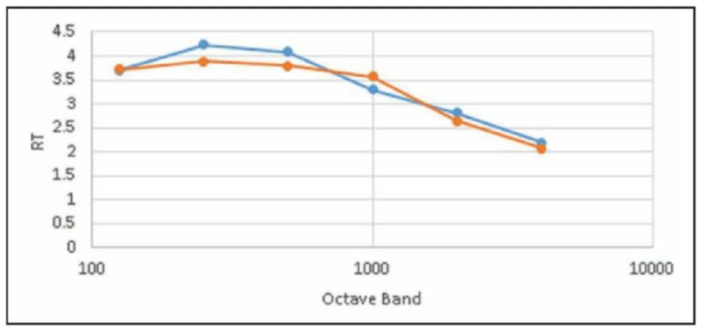


Figure 3: The measured and modeled RT—while not identical, are nearly the same.

The measured RT was too high, as expected, and significantly decreased at low frequencies, making the room sound "thin." RT also decreases too much at high frequencies, making the room sound "dull." I built an EASE computer model and tweaked the materials to make the modeled RT as similar to the measured values as possible. The results are shown in **Figure 3**. **Photo 2** shows a map of STI values of the untreated MP room. These values, averaging about 0.45, indicate poor-to-fair speech intelligibility. The music clarity in the untreated room was 0.63 dB.

The thickness of fiberglass wall panels affects the frequency dependence of the acoustical absorption. **Figure 4** shows the variation of absorption coefficient with frequency for three types of fiberglass panels: 1", 2 1/8", and a special panel (RS1) made by Golterman and Sabo for improved low-frequency absorption relative to the mid-frequency value.

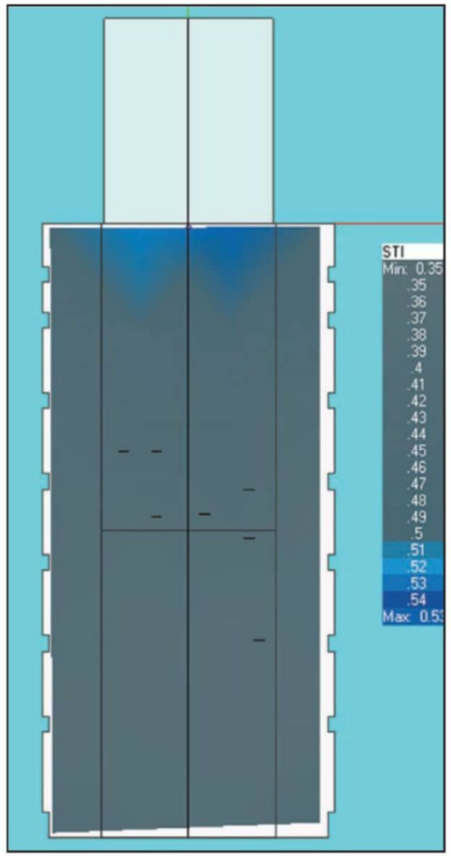


Photo 2: This STI map shows the speech intelligibility in the seating areas of the untreated room.

Figure 5 shows the average RT vs. frequency when the rear and house-right walls were treated with 2 1/8" fiberglass, and the ceiling was treated with suspended mineral fiber ceiling tile. As you can see, low-frequency RT is excessive, which would make the room sound boomy.

Photo 3 shows the echo performance in the front and rear seating areas. This is based on analysis developed by L. Dietsch and W. Kraak. In this analysis, an "echo speech" rating of 1 or greater signifies poor intelligibility due to excessive echo. The colors used in the map shown in Photo 3 are:

- Green for values below 0.7 (echo not an issue)

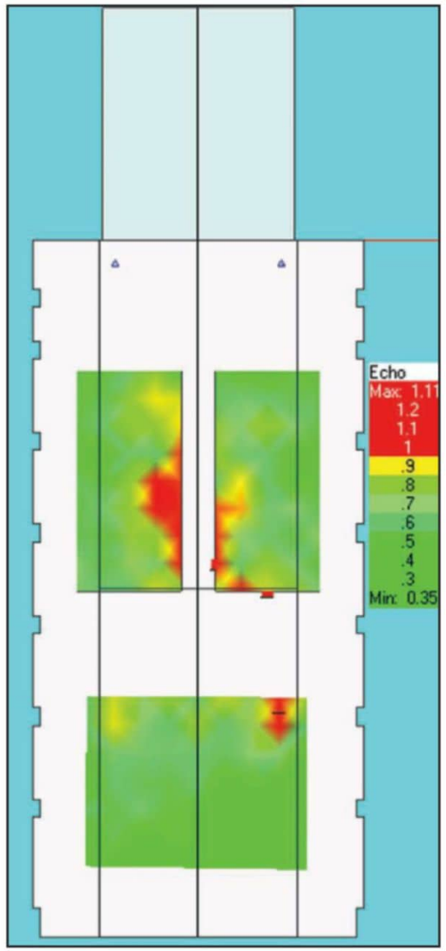


Photo 3: The "Echo Speech" criteria show serious echo issues in several locations.

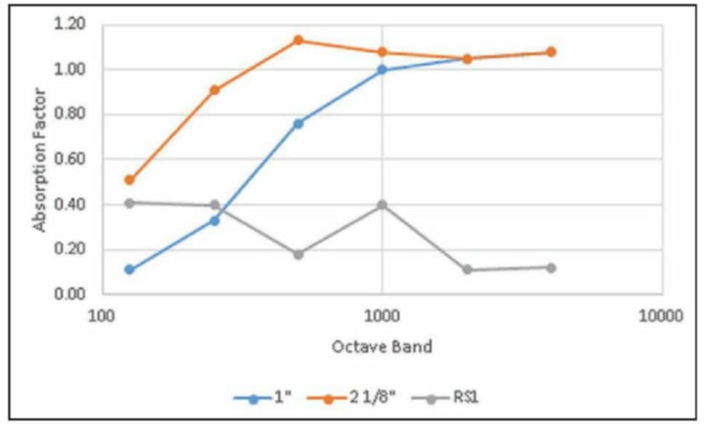


Figure 4: Different fiberglass panels have different variations in absorption with frequency.

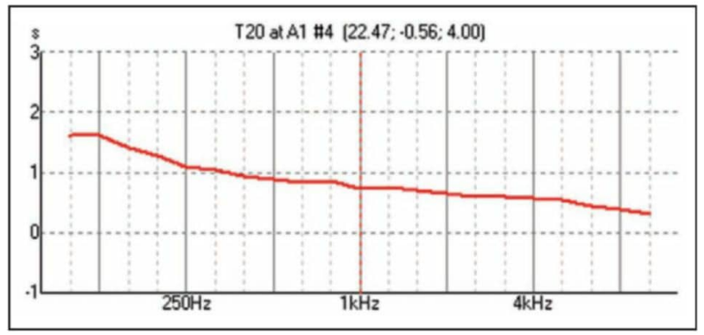


Figure 5: The RT was too high when only 2 1/8" fiberglass wall treatment and mineral fiber ceiling tiles were used.

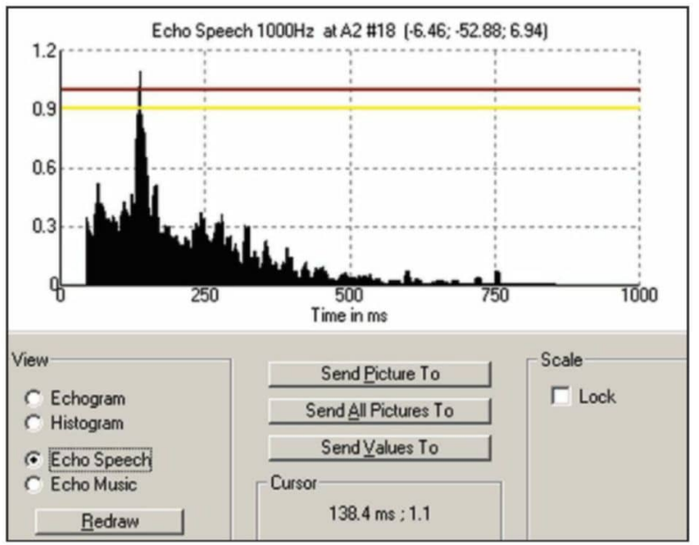


Photo 4: The Echo Speech graph enables users to find the level and arrival time of offending reflections.

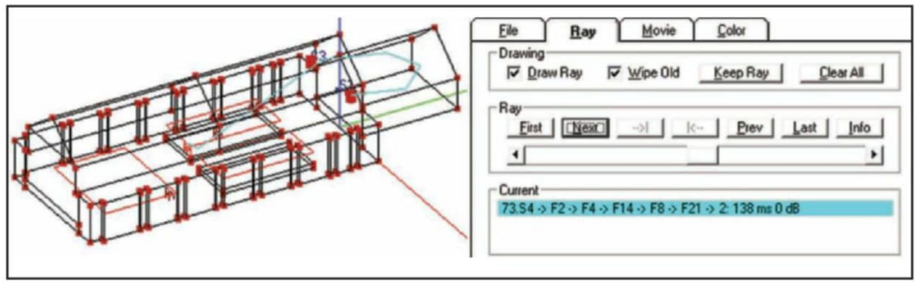


Figure 6: EASE Ray Tracing Impacts helps find which surfaces are involved in problematic echoes.

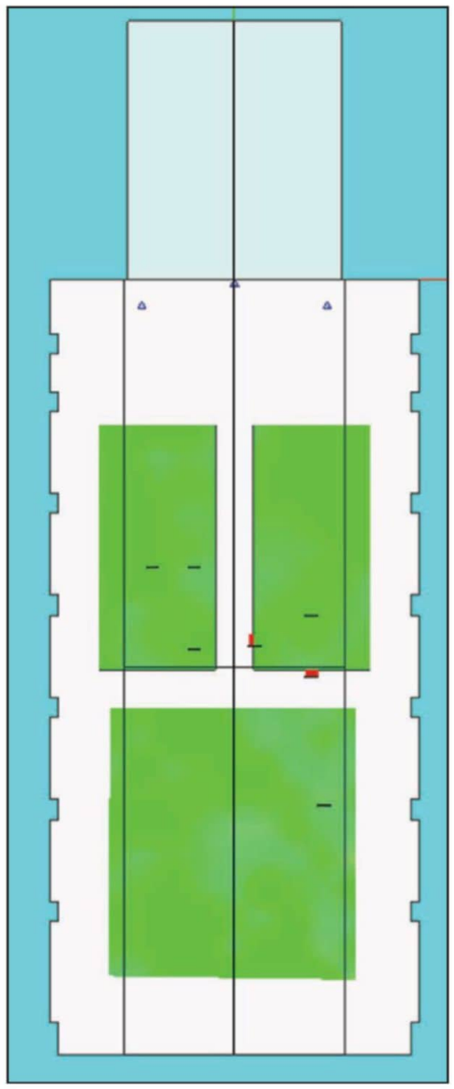


Photo 5: The final treatment design eliminated echo problems.

- Yellow-green to yellow for values 0.7-0.9 (echo potentially an issue)
- Red for values 1.0 or higher (echo definitely an issue)

Photo 4 shows an echo speech echogram. Note that the highest offending peak occurs at 138.4 ms (see the lower center of the illustration). The Ray Tracing Impacts feature of EASE enables us to see which surfaces ("faces," in EASE terminology) are involved in the offending echo. Ray Tracing Impacts can display one ray at a time. Each ray is accompanied by a history listing source, surfaces (by face number) from which the ray is reflected, time of flight before it hits the target (listener seat), and level at the target. We can trace the rays from the speaker to a listener seat located at the position of highest echo speech, then plot the ray whose time matches the highest offending peak from the graph. **Figure 6** shows that in this case, the ray begins at speaker S8, reflects off faces 2, 4, 14, 8, and 21, and then arrives at listener seat 2. We will need to treat one or more of those five faces in order to prevent the echo.

In order to correct the excessive low-frequency RT, I began by applying Golterman and Sabo RS1 panels to all walls except, of course, the glass ones. I then ran a new echo analysis and noted which walls were involved in offending echoes. I changed the treatment of these walls to 2 1/8" fiberglass, which has greater absorption than RS1. Eventually, I eliminated the echo problems, as shown in **Photo 5**.

Unfortunately after the echo was eliminated and the low-frequency RT was brought under control, overall RT was a bit lower than the target (see **Figure 7**). Using diffusion panels instead of absorption panels on some walls would have allowed us to more closely approach the target RT. However, the small acoustical benefit was not judged to be worth the added cost.

Photo 6 shows the speech intelligibility after treatment to eliminate echoes and improve RT. The average value is 0.705: solidly in the "good" range. Music clarity is predicted to be >7 dB, permitting excellent musical articulation.

Room Uses

When a room is used for social events such as dinners, receptions, rummage sales, and so forth, most of the occupants are likely to be talking at any given time. The combined sound from all these talkers can build up to quite high levels of noise. When the background noise is high, each talker automatically raises his/her voice in an effort to "talk over" the noise. Naturally, this increases overall noise, causing occupants to talk louder, resulting in a vicious cycle! The desirable conditions for social conversation can be stated in terms of signal-to-noise ratio (SNR), usually specified in decibels (dB). The human ear/brain system is able to understand speech in the presence of noise that is louder than the speech (an SNR in the negative decibel range).

For brief exposures (e.g., people passing through a theater lobby), the lowest SNR that can be considered comfortable is -2 dB. Lower (more negative) ratios cause vocal and emotional stress, and interfere with communication. For longer exposures (e.g., dinners or receptions), the SNR can be as high as - 1 dB. Of course, less negative (i.e., more positive) SNRs are better. The SNR can be predicted based upon the number of occupants and the total acoustical absorption (measured in sabins) in the room.

For the MP room we have been examining, the total absorption after acoustical treatment has been applied will be about 8820 sabins. The estimated maximum occupancy is 200 people, and with that occupancy, the predicted SNR is 2.03. The maximum number of occupants the room could handle for extended periods, based upon conversation noise conditions, is about 435.

With careful acoustical design, even large spaces can be made to provide excellent acoustics for contemporary music concerts, speeches, and social events.

- RT is the time required for sound at the level of ordinary conversation to decay to inaudibility.
- C80 is the decibel ratio of sound energy occurring in the first 80ms of an acoustical event to the overall acoustical energy in the event. It is one quantification of the "early-to-late" energy ratio.
- Speech Transmission Index (STI) is a measure of how much a sound wave is modulated by noise, echo, and reverberation. It has a value between 0 and 1, it higher values indicating better intelligibility.

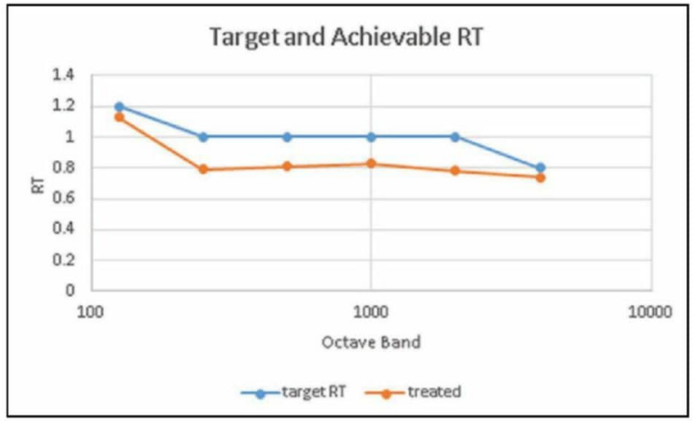


Figure 7: Eliminating echo brought the RT a bit too low.

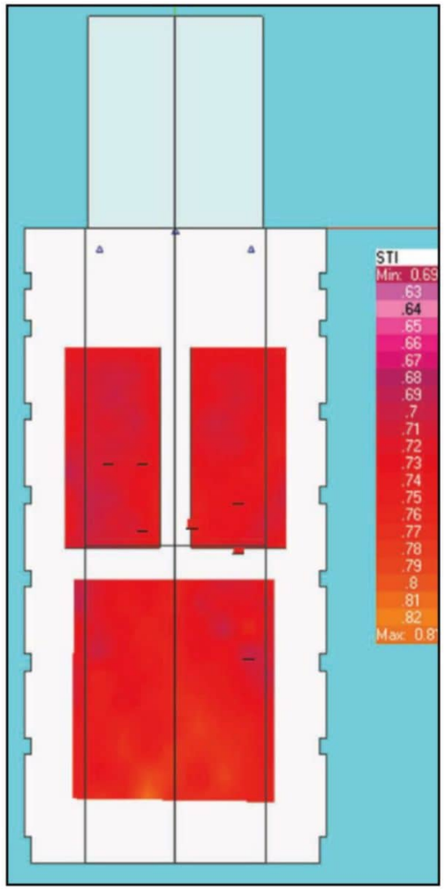


Photo 6: The final STI map shows good speech intelligibility.

Simple Improvements to Legacy Differential Transimpedance to Single-Ended Audio DAC Output Designs

By
Michael Steffes
(United States)

In this article, Michael Steffes explores some simple linearity improvements for a two-stage differential transimpedance to a differential to single-ended design incorporating an active filter.

There appears to have been a long legacy of audio DAC output designs implementing a differential transimpedance first stage into a differential to single-ended stage providing an active filter in that second stage. **Figure 1** shows a typical example design (see Figure 35, Reference 1).

Each of the two 820 Ω OPA1611 transimpedance stages provide a virtual ground to this current output DAC. The DAC's common mode output current (2 mA assumed here) will level-shift the amplifier's output common mode voltage negatively. But that DC level will be rejected by the second stage. This second stage looks very balanced with the symmetric RC values. It is actually quite unbalanced—but can be easily improved.

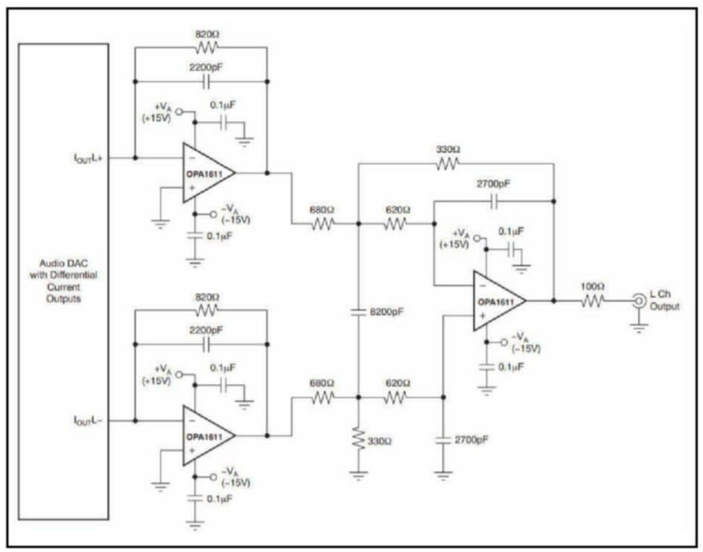


Figure 1: Differential transimpedance to differential to single-ended active filter stage.

With the emergence of low power, very low distortion, fully differential amplifiers (FDAs), a simpler first stage current to voltage conversion can be implemented with any one of the example devices listed in **Table 1**. The THS4561 (see Reference 2) will be used here—reducing the quiescent current requirement in that stage from the 7.2 mA for two OPA1611s, to a nominal 0.83 mA for the supply current trimmed THS4561. Only two slight adjustments to the design of Figure 1 are required to apply the THS4561 FDA in this application.

All FDAs control their "output" common mode voltage. To control the input common mode voltage (the DAC output voltage), set the common mode control into the FDA to target an output common mode that will level shift up to the desired DAC output voltage via the DAC's common mode current times the feedback resistor value.

Some FDAs (e.g., the THS4561) are nominally compensated for a slightly peaked frequency response at a noise gain of 2 (all equal resistors, signal gain of 1). Using the signal path bandlimiting feedback capacitor of Figure 1, will reduce that noise gain to 1 V/V at higher frequencies reducing the phase margin possibly moving toward instability. A simple fix is to add a differential input capacitor that shapes the noise gain up over frequency providing a phase margin tune. This will be peaking the output noise from the FDA. But, with that second stage active filter, this higher frequency noise peaking will be cut-off by the second stage filter.

Figure 2 shows an example THS4561 design using ±6 V supplies implementing this complementary current output DAC transimpedance stage. The simulated results are assuming a 2 mA midscale current from the DAC with ±2 mA full-scale output on each side. Continuing with the 820 Ω transimpedance gain of Figure 1, this requires a V_{OCM} control voltage input of -1.64 V to level shift up from the FDA output common mode to a 0 V virtual ground at the DAC outputs. Adding the 6.8 nF across the input summing junctions improves the nominal phase margin from 21° to 36°, using the original 2018 THS4561 TINA model.

The DC gain is simply 20log(1640 Ω) = 64.28 dBΩ, where the 2.2 nF feedback capacitors set a nominal F -3dB of 89 kHz in this stage. The slight response peaking at 8.4 MHz shows the loop gain crossover point with approximately 36° phase margin. The differential input capacitor is effectively increasing the high-frequency noise gain (well above the audio band) to 1+(13.6 nF/2.2 nF) = 7.2 V/V. This does start to peak the output spot noise at approximately 20 kHz—but that will be attenuated by the second stage filter. Some of the alternate FDAs listed in Table 1 (e.g., the OPA1637) are designed for higher nominal phase margin in a unity gain feedback and would not require this differential input noise gain shaping capacitor.

Device #	Typical Iq	GBP	Slew Rate	Input Noise Voltage	Comments
OPA1611	7.2 mA	79 MHz	54 V/µsec	1.1 nV/√Hz	Using 2, original Ckt. (2×, 3.6 mA/channel) diff. slew rate
THS4561	0.83 mA	60 MHz	215 V/µsec	4 nV/√Hz	±6.3 V maximum supplies
OPA1637	0.95 mA	8.5 MHz	15 V/µsec	3.7 nV/√Hz	
LTC6363	1.7 mA	500 MHz	75 V/µsec	2.9 nV/√Hz	Shaped Aol, 35 MHz gain of 1 bandwidth
AD8137	3.2 mA	70 MHz	450 V/µsec	8.25 nV/√Hz	±6 V maximum supplies
OPA1632	14 mA	180 MHz	50 V/µsec	1.3 nV/√Hz	Same as THS4131

Table 1: Original two OPA1611 solutions with ascending quiescent current FDA options.

Simple Improvements to the Differential to Single-Ended Active Filter Stage

Ignoring the capacitors in the second stage of Figure 1 (considering only the less than 20 kHz audio passband for now), this basic circuit also suffers from an imbalanced input impedance (see Reference 3) since the inverting input path shows an active impedance characteristic. The divider ratio into the non-inverting side provides an α = 0.33 causing the input load on the inverting side to appear as 680 Ω/(1 + α) = 511 Ω. These loads are not strictly speaking a differential load to the FDA (see Figure 2) driving this stage. But it should reduce even-order distortion to match the single-ended loading in the audio band as closely as possible. This simply requires adjusting the two input resistors on the non-inverting side to match both the 511 Ω inverting side load and the original divider ratio (342 Ω to ground with a 169 Ω input resistor) to get the differential to single-ended operation.

The filtering provided by this original design is considerably more involved but essentially trying to provide an output response cut-off in the 80 kHz region—but how? Intuitively, this circuit is a combination of a second-order multiple feedback (MFB) low-pass filter on the inverting side with a single low-pass pole into the non-inverting input pin. Some operational observations include:

The differential input capacitor will get an increase in effective value through the partial inverted signal appearing at the non-inverting input voltage divider. While this capacitor appears as a 2× value for FDA-based MFB filters (see Reference 4), here it will appear as (1 + α) × 8.2 nF = 10.9 nF for the filter shape applied to the inverting signal side. The MFB response will use the input resistor value on the inverting side, not the active load value.

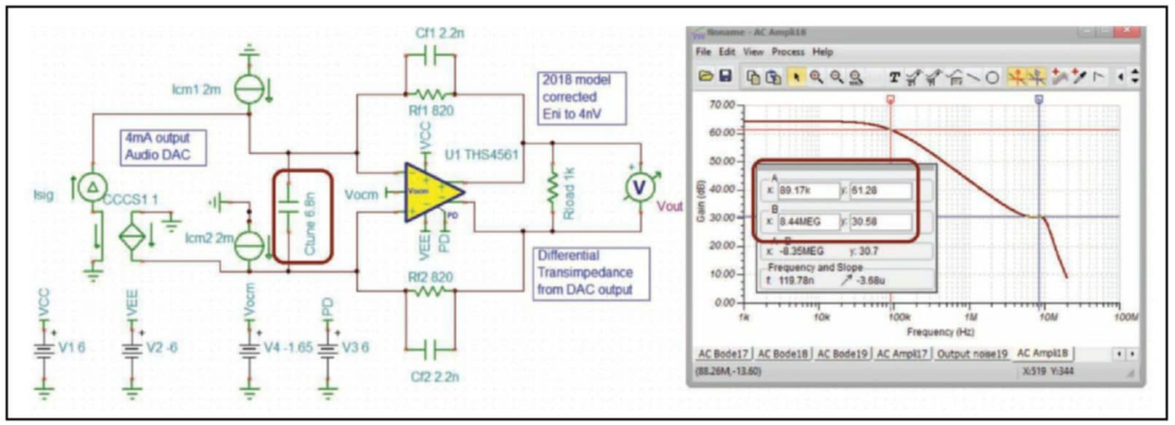


Figure 2: FDA-based differential transimpedance Audio DAC implementation

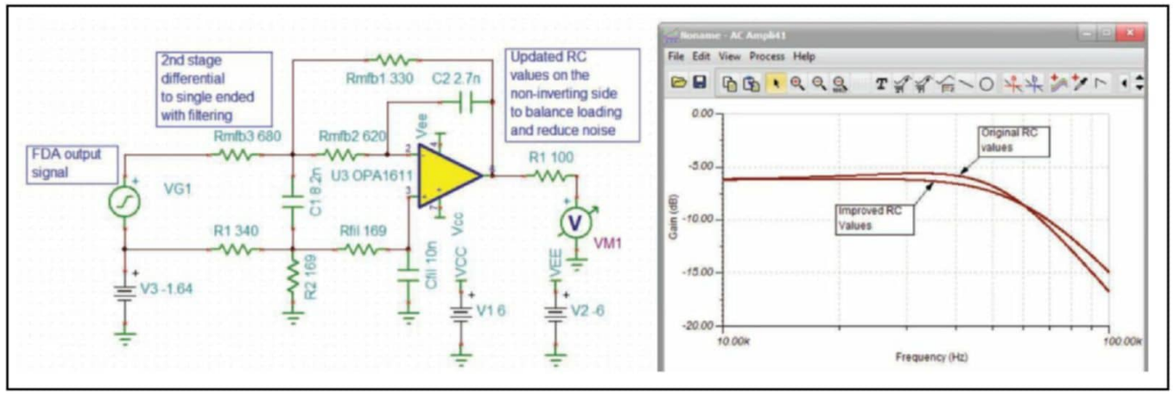


Figure 3: Improved differential to single-ended design response comparison to original RC values

Since the signal present at the V+ input pin gets the "noise gain" response of the MFB filter (and that always peaks around the filter F_0), the simple RC into the V+ input is intended to roll off that peaking. It is not necessary to use those symmetric RC values (see Figure 1) and reducing the series R into the V+ input will provide a meaningful reduction in output noise. The original 620 Ω value adds a 3.15 nV/√Hz noise term—dominating the very low 1.1 nV/√Hz for the OPA1611 audio op-amp. Considering just the last RC into the V+ input of Figure 1, that pole at 95 kHz could be approximately matched by a series 169 Ω to a 10 nF capacitor at the V+ input pin.

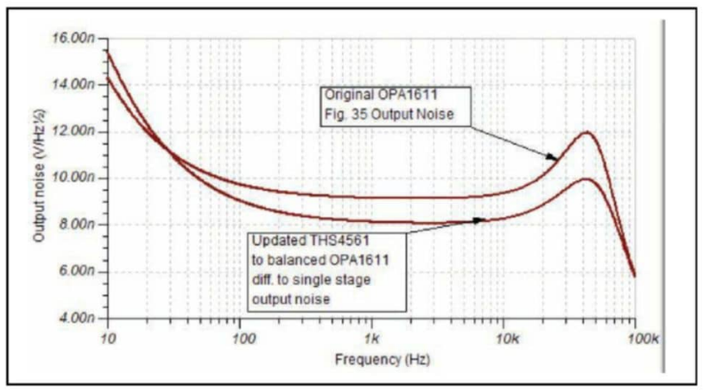


Figure 4: Output spot noise voltage comparison

These simple changes are shown in **Figure 3**, where the response shape closely matches the original for this improved second-stage design.

Putting the THS4561 differential transimpedance stage (see Figure 2) together with this improved second stage gives a similar frequency response as the original design, with the slightly improved output spot noise voltage shown in **Figure 4** (using the 2018 THS4561 TINA model with corrected input noise terms).

The lower mid-band output noise in the improved design is largely due to the reduced resistor values into the V+ input of the OPA1611. Each output of the FDA now drives a single-ended 511 Ω load for an equivalent 1.02 kΩ differential load. This should yield extremely low odd-order harmonics (less than -120 dBc) in the audio band with slightly improved even-order terms (see Figure 7-7, Reference 2) implementing the matched loading on each side.

If you need to implement a differential transimpedance design from a complementary DAC output current, consider the low power FDA options shown in Table 1. That differential FDA output can often be used directly into the next power amplifier, but, if a single-ended output is desired, consider the simple improvements described here to the legacy differential to single-ended design exemplified by Figure 1.

References

[1] "SoundPlus High Performance, Bipolar Input Audio Op Amp," Texas Instruments OPA1611 Datasheet, revised August 2014, www.ti.com/lit/ds/symlink/opa1611.pdf

[2] "Low-Power, High Supply Range, 60MHz, Fully Differential Amplifier," Texas Instruments THS4561 Datasheet, revised August 2020, www.ti.com/lit/ds/symlink/ths4561.pdf

[3] M. Steffes, "Simple Improvements to Legacy Audio DAC Output Interfaces Using Op-Amps," *audioXpress*, November 2020.

[4] T. Kuehl, "Using the infinite-gain, MFB filter topology in fully differential active filters," *Texas Instruments Analog Applications Journal*, 3Q2009, www.ti.com/lit/an/slyt343/slyt343.pdf

Engineering Consulting for the Audio Community

Celebrating 37 YEARS 1983-2020

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

QuantAsylum QA401/QA451

Redefining the Compact and Modular Audio Measurement System

By
David Logvin
(United States)

In the world of audio measurement, it has long been accepted that: the setup time for electroacoustic testing often takes longer than the actual measurement and the cost of audio test hardware prohibits deploying an analyzer on every engineer's desk. QuantAsylum—a rising test and measurement company based in Snoqualmie, WA—is looking to change both of those ideas with affordable hardware offering up to a 192 kHz sample rate, differential inputs and outputs, full-isolation from the USB-connected host, and programmable electronic loads permitting rapid power amp testing up to 200 W. In this article, David Logvin reviews the core product, the QA401 Audio Analyzer and its new companion accessory, the QA451 Programmable Load.

QA401 Audio Analyzer
QuantAsylum USA LLC
7829 Center Blvd SE #225
Snoqualmie, WA 98065
https://quantasylum.com
List Price: $449

QA451 Programmable 8 Ω/4 Ω Load
List Price: $289

By its very nature, the world of electroacoustics is an interdisciplinary one. Audio practitioners are often involved in varied aspects of the product's life cycle, from research and development, ideation, design, production test, and service. This constant need for data acquisition involves complex and time-consuming setups and fixtures.

In today's flexible workspaces, access to the sophisticated test and measurement systems required for modern products often presents a resource bottleneck.

So when I first learned about the QA401 audio analyzer, I was intrigued by the value offered. It provides dual-channel FFTs, balanced inputs and outputs, up to 192 kHz sample rates, and a -118 dBV self-noise. While there are other software solutions in the sub-$500 price point, competitive offerings typically force you to bring your own sound card to the party. The QA401 is a complete standalone system of hardware and software that does not utilize a sound card (see **Photo 1**).

During my discussions with QuantAsylum, I learned that they are in the process of upgrading one of their key accessory products for this analyzer, so I also requested a pre-production version of the QA451—a programmable load that easily allows testing of audio amplifiers of up to 300 W, thanks to the use of burst mode test signals. The QA451 is the successor to the QA450. The primary upgrades of this hardware companion are improved tolerances, as well as steeper roll-off on the low-pass filter at 34 kHz (sixth-order on the QA451 vs. first-order on the QA450).

Standalone Hardware, No Sound Card Needed

Audio test systems fall into two general categories: dedicated hardware or PC sound card-based. When cost is no object, or in large commercial settings, the dedicated hardware system is typically chosen for increased precision. It also better meets calibration and traceability customer requirements. Sound card solutions offer lower cost, increased portability and are often the first solution when a new production test system needs to be deployed and replicated with minimal investment. It is well known that dealing with Windows audio issues can be problematic, due to the multiple volume mixers or Microsoft updates that can reset gain settings for inputs and outputs. To address these concerns, audio test systems often favor using ASIO-based sound card solutions to prevent audio gain changes that occur when standard Windows audio drivers are engaged. But switching to ASIO sound cards will add further cost to a system that is often a budget-sensitive expense. With standalone hardware, this "pain point" of calibration and setup is eliminated. This is a strong positive usability advantage for the QA401.

Ready Already

In most cases, audio test systems often have a lengthy calibration setup, as well as a manual process where at least once per day, a series of reference "loop back" measurements need to be performed prior to measuring your DUT. It is quite refreshing that the QA401 comes ready to go and calibrated out of the box with both stereo differential inputs and outputs. The calibration of the system can be further fine-tuned by adjusting the calibration text files and validating them with a known lab-grade DMM or oscilloscope. The system's actual tolerance is ±0.06 dB. With fine tuning, it is possible to bring that variation down to ±0.02 dB for a single channel. Typically, the factory provided calibration is more than sufficient for most applications.

Mechanicals

The QA401 is a compact audio analyzer. The front panel has stereo differential inputs and outputs, as well as indicator LEDs for PC link, attenuation, and run status (see **Figure 1**). The back panel has a single USB Type B port and requires up to 600 mA of current, so the use of a powered USB hub is recommended, in case a USB port in your PC is current-limited. The attractive casing is sturdy aluminum with a full color UV printed front panel. There are no buttons, knobs, or switches, as the device is fully controlled by the system software (see **Photo 2**).

Photo 1: QuantAsylum's QA401 audio analyzer is a flexible hardware solution for electroacoustic measurements.

The pre-production QA451 sample is a programmable load that streamlines testing of audio power amplifiers. QA451 I/O is similarly streamlined, with indicators, load inputs, single-ended outputs, and current sense on the front panel. The back panel has a single USB port with a current draw of under 500 mA.

Basic Measurements and GUI

The QA401 offers four primary measurements: RMS/Power, THD/THD+N, SNR, and Frequency Response. These are all accessible under the "Measurements" panel in the QAAnalyzer software package.

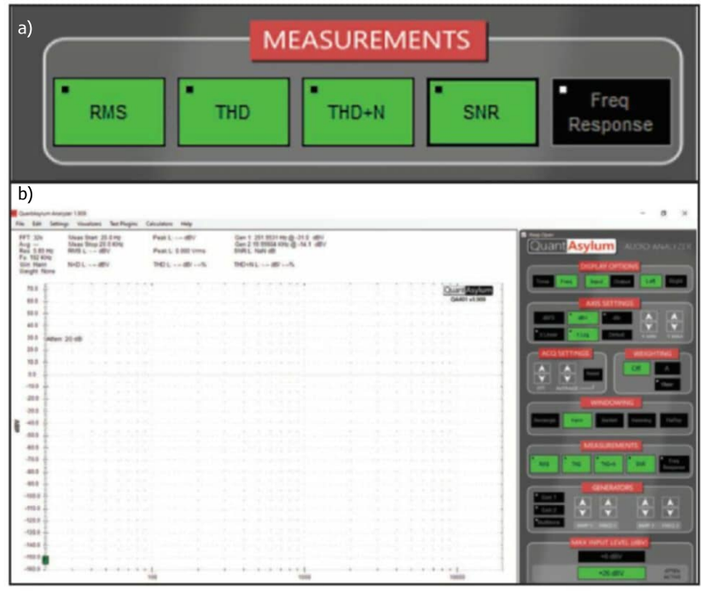


Figure 1: This panel offers quick access to the four primary measurements that are accessible in the QAAnalyzer application (a). This image shows the main panel for QAAnalyzer software package (b).

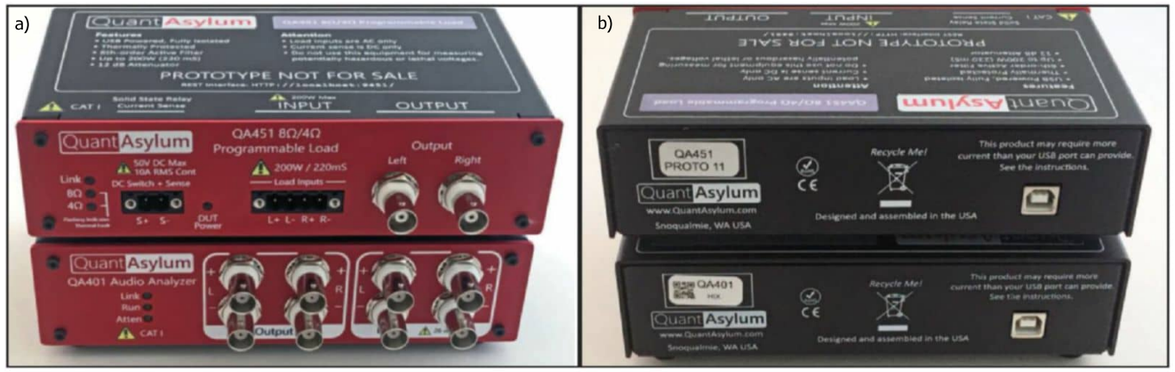


Photo 2: This close-up image shows the front panels for both the QA401 and the QA451 (a). The rear panels of the QA401 and QA451 each have a single USB port (b).

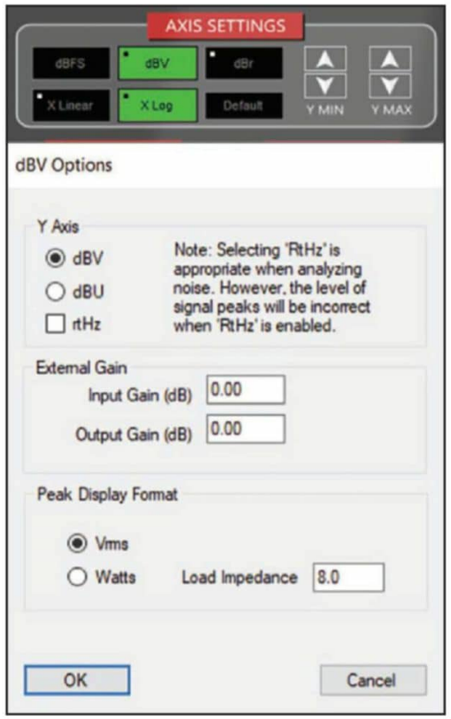


Figure 2: These are the dBV options from the submenu panel.

From inspection of the application GUI, it is readily apparent that the software architecture is focused on delivering results in an uncluttered, well-organized and deliberate fashion. Distinctively, the menu bar is not the key method to drive and control the application. Instead, controls are contained within a graphical panel on the right side of the screen. From this interface, the end user can select display and axis settings, as well as choose basic tests, filter windows and weighting, signal generators, and the attenuator level. When there are more user options, this is denoted with a dot on the relevant button (see **Figure 2**). To access these context menus, simply long press or control-click the button to make the necessary changes to the submenus. Updates are reflected in the display area on the top of screen.

Interestingly, the program eschews keyboard shortcuts, in favor of this embedded contextual menu approach. Checking the documentation, I noted that there are only six keyboard shortcuts available within the application, for things such as adding markers to the display or pressing the Run/Stop button. While different than many Windows apps, after a brief acclimation period, it becomes quite simple to navigate and operate like a pro.

Exploring the menu bar, you will find the intriguing Test Plugins section. This provides access to advanced setups, streamlines processes, and offers production-ready automation tools. Rapid access to tests such as THD vs. SPL, amplifier gain vs. distortion, microphone comparison, or even speaker rub and buzz offers a myriad of options for electroacoustic testing.

Additionally, QuantAsylum has released an open-source application called Tractor that allows typical test lists to be executed and logged, without the need to write any code or scripts. It is as simple as just adding tests from pull-down menus and arranging the necessary order of operations (see sample in **Figure 3**). If you do need to add a new test, source code is publicly available, and you can build the needed .DLL files using Microsoft Visual Studio.

Another added benefit of Tractor is that data can be logged as a simple .html file. Data can be stored locally, or QuantAsylum also provides free cloud access and storage of your data—permitting you to post-process and analyze your test results as you further refine your production test methods.

Solutions to Go

Since 2015, when QuantAsylum released its first audio analyzer, its product offerings have been quite distinctive and full-featured. Traditionally, audio measurement software is often distributed in three tiers of increasing cost: basic, advanced, and quality control. The engineers at QuantAsylum have chosen a more impactful model and instead provide all three software tiers free with the hardware, freeing use, which has the added benefit of not limiting user contributions due to arbitrary constraints.

In addition to expediting production test software and methods, QuantAsylum offers additional hardware tools to enhance and simplify hardware setups. The QA451 is a programmable load (4 Ω or 8 Ω) and can support up to 300 W in short bursts of 300 ms. Key features include full isolation from the PC, thermal-protection, a built-in 12 dB attenuator, and a sixth-order active low-pass filter at 34 kHz to keep any power supply switching noise out of your measurements. Manual control of the device is easy via the QA451 mini-app, or there is full support using the REST interface from your preferred development language, if you choose a highly customized web-based solution.

In addition to the QA451, other specialized hardware accessories available from QuantAsylum include a microvolt DC voltmeter, a transducer driver for measuring Thiele-Small parameters, a low-noise mic preamp, and a high-side current sense amplifier.

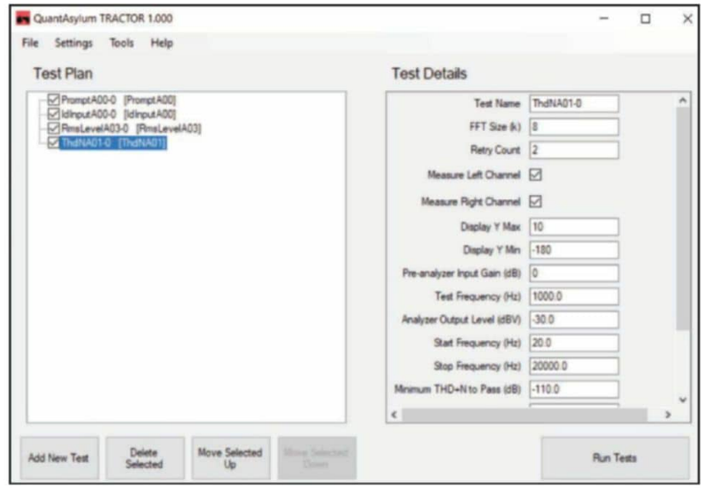


Figure 3: This is a sample Tractor interactive test plan, including operator prompts and inputs.

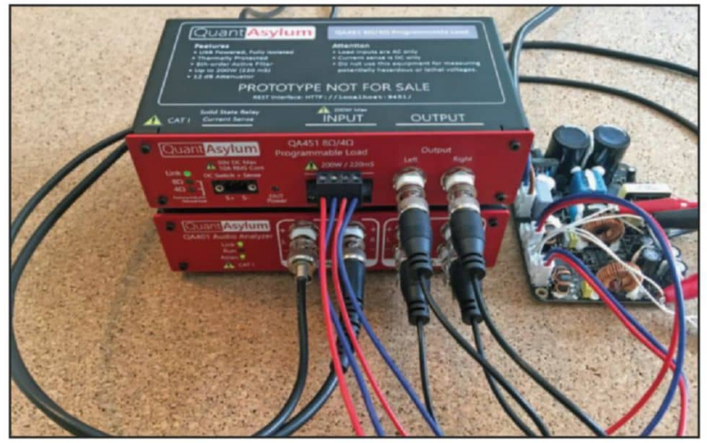


Photo 3: This is the ICEpower amplifier DUT setup.

FFT: 32k	Meas Start: 20.0 Hz	Peak L: -39.39 dBV	Gen 1: 1.001953 KHz @ -60.0 dBV	Phase L: 175.25 deg
Avg: ---	Meas Stop:20.0 KHz	Peak R: -39.43 dBV	Gen 2:1.000488 KHz @ -60.0 dBV	Phase R: 173.89 deg
Res: 1.46 Hz	RMS L: -39.4 dBV	Peak L: 10.72 mVrms	SNR L: 42.0 dB	Delay L: 523 uSec
Fs: 48.0 KHz	RMS R: -39.4 dBV	Peak R: 10.67 mVrms	SNR R: 43.2 dB	Delay R: 527 uSec
Win: Hann	N+D L: -81.0 dBV			Gain L: 20.62 dB
Weight: None	N+D R: -82.3 dBV			Gain R: 20.58 dB

Figure 4: RMS gain measurement of the ICEpower 50ASX2SE amplifier module shows a value of 20.62 dB/20.58 dB, which is in close agreement with the datasheet specification of 20.5 dB.

About the Author

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

First Look

For initial testing, I chose to measure a spare Class D amplifier, the venerable ICEpower 50ASX2SE. This is the single-ended stereo version, which delivers 2× 50 W into 4 Ω. As it has been in storage for a number of years, I decided that three suitable tests to verify performance would be measurements of gain, frequency response, and THD+N. As this runs off mains AC and does not use an outboard DC power supply, the current sense feature of the QA451 cannot be used. In **Photo 3**, the DUT setup for the ICEpower module and connections to the QA401 and QA451 are shown.

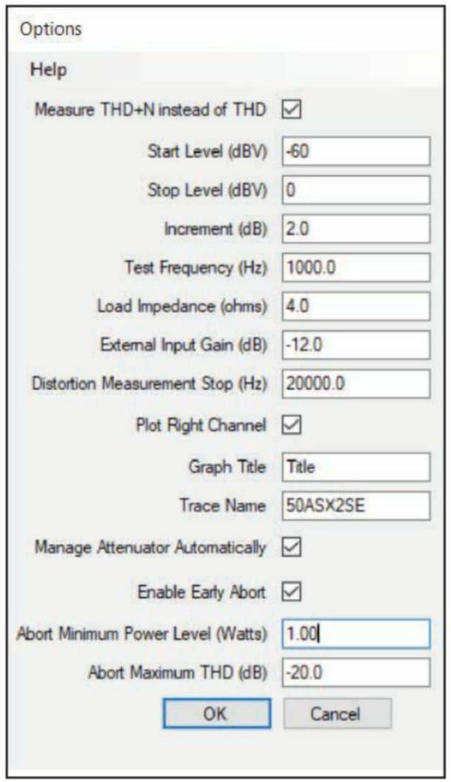


Figure 5: Option settings are shown for the Test Plugin PWR THD vs. Output Power.

I measured the gain using the RMS/Power measurement feature. Results are shown in **Figure 4**. Next, I measured the full-range frequency response via the AMP FrequencyResponseChirp. Finally, I measured THD+N using the Test Plugin PWR THD vs. Output Power (see **Figure 5** and **Figure 6**).

Next, I tested another Class D amplifier, the Fosi Audio TDA7498E Class D Amp (2× 165 W). This unit uses an external DC power brick (24 V × 4.5 A), so the current sense features of the QA451 are relevant here and will be highlighted. The setup is shown in **Photo 4**. The QA451 software panel (see **Figure 7**) conveniently provides the current draw of the DUT as well as thermal monitoring on the load resistors, ensuring that test cycle time can be properly optimized. For this device, I measured the amp self-noise, THD vs power, and frequency response into both a 4 Ω and an 8 Ω load (see **Figure 8**).

Test Plans in the Cloud

As previously mentioned, QuantAsylum has provided a free open-source software tool called Tractor, dedicated to easily executing automation test plans. In **Figure 9**, a sample test plan that I constructed is shown. No code or scripting is necessary (although source-code is publicly available on GitHub if you are interested in deep customization). Creating the plan only requires the user to press the Add New Test button, select the appropriate test and setting, and then adjust the desired execution order.

Data logging options are surprisingly rich. Test data can be saved in .csv format for ease of post-processing, exported to a corporate SQL server, or even stored on a free Cloud database

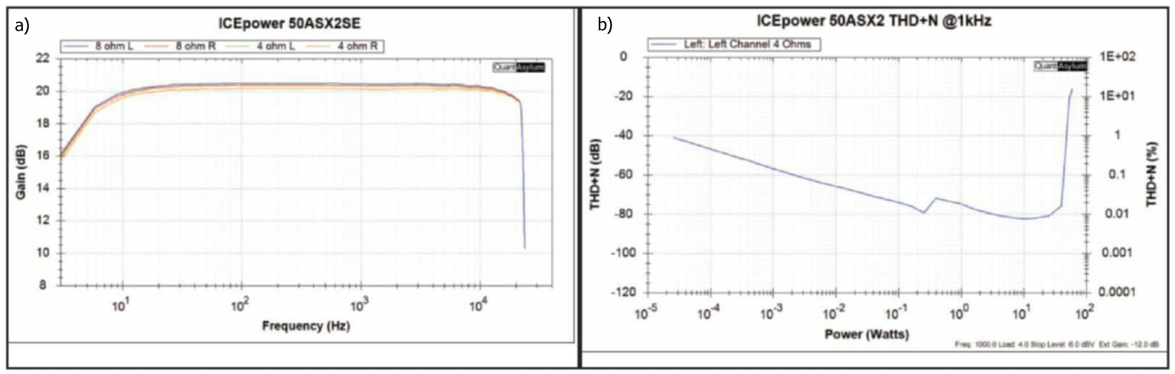


Figure 6: This is the measured ICEpower frequency response (a). This is the measured ICEpower THD vs. output power (b).

16065656

that QuantAsylum provides to its customers at no charge. The myriad options offer great flexibility for manufacturers large and small, especially those looking at enhancing their data and production record keeping needed for audit or certification purposes. One further bonus, is that the cloud database already provides typical statistical analysis, calculating and logging the mean and standard deviation values automatically. If you require more sophisticated analysis, then simply download and post-process the data as needed.

Comparison on Automation—Take a REST, Ride a Tractor

If you plan on using the QA401 and the QA451 in a production environment, there are a plethora of tools available right out of the box, from easy scripting options, to simple "point and click" GUI options, plus modifications to the open-source code for the core applications, if you want to go for a truly customized and unique build.

When the product you are testing has non-audio functions, it could prove useful to utilize a standard programming or scripting language so that the QuantAnalyzer can be integrated into a fully-automated test program. The QA401 and QA451 both

Photo 4: This is the Fosi amplifier DUT setup.

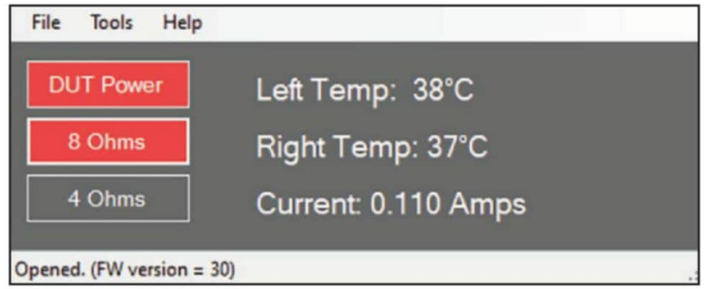


Figure 7: Load temperature and current draw is monitored by the QA451 app. It operates independently of the QAAnalyzer app and provides control over impedance load and DC DUT power.

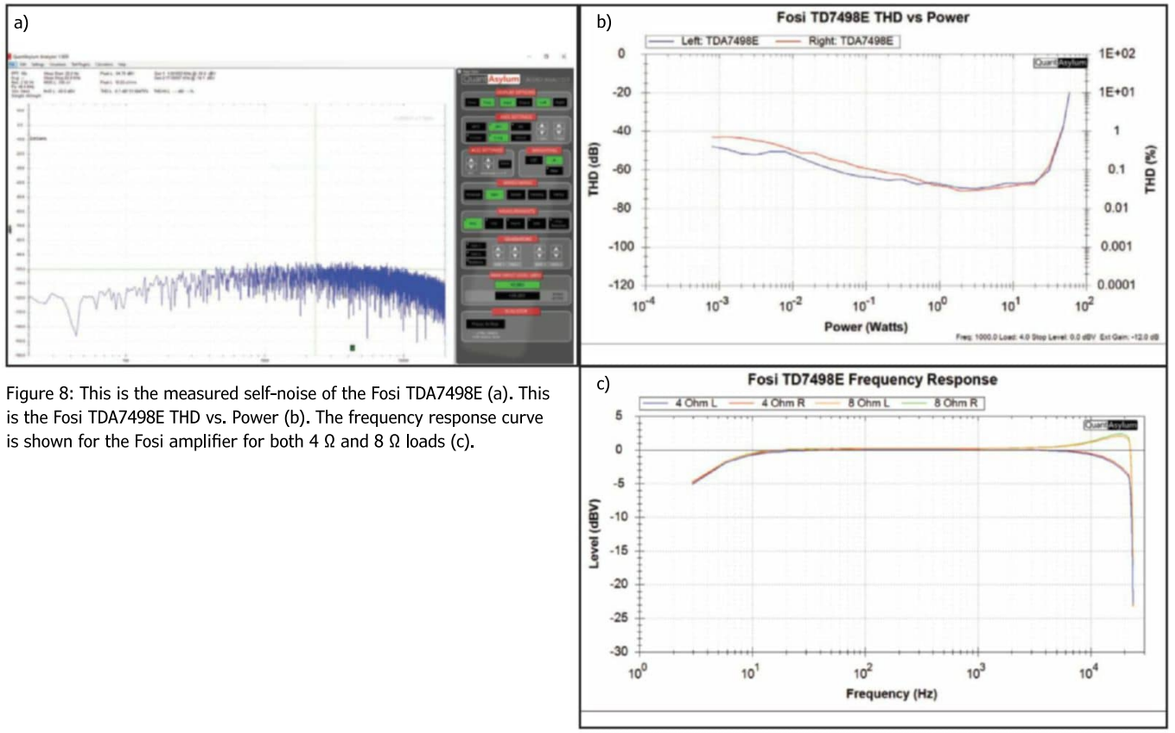


Figure 8: This is the measured self-noise of the Fosi TDA7498E (a). This is the Fosi TDA7498E THD vs. Power (b). The frequency response curve is shown for the Fosi amplifier for both 4 Ω and 8 Ω loads (c).

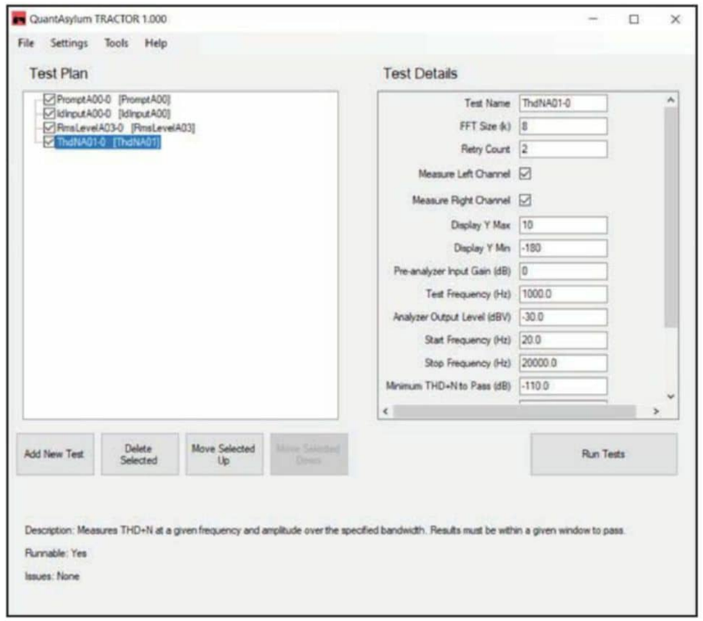


Figure 9: A sample test plan created in the free Tractor app provides an operator prompt, operator info, RMS level check, and THD+N measurement at 1 kHz in a single automated sequence, with no code or scripting required.

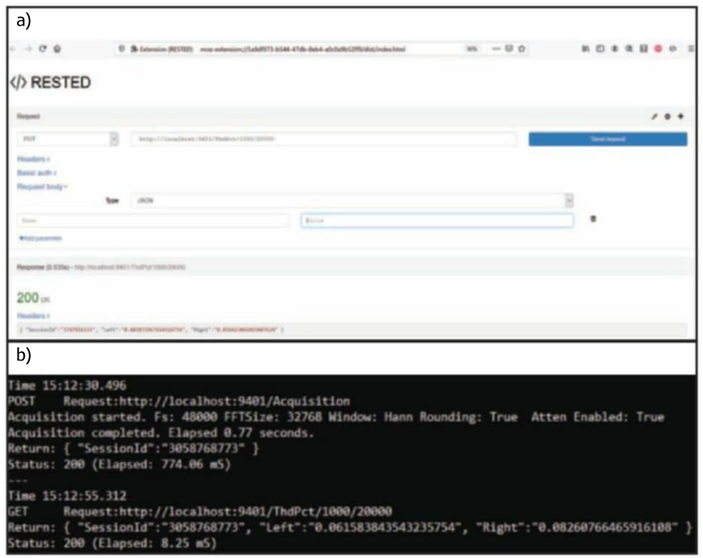


Figure 10: THD percentage has been measured and displayed over a REST interface in a Firefox window. The computed values are displayed at bottom as left =.0438% and right = .03642% for ICEpower 50ASX2SE (a). This command prompt is echoing the commands issued over the REST interface (b).

support the Representation State Transfer (REST) protocol for control over a web-based interface. To investigate this functionality, I first downloaded the "headless" (i.e. non-graphical) version of the QA401 software, in order to interact via a Command Line Interface (CLI).

Once I installed that package, I opened a Command Prompt in Windows 10 and enabled a "listening session" over the USB prompt and the headless control software. Next, I opened a FireFox window with a REST extension enabled and began entering the various control parameters using the standard functions over the web interface with commands such as PUT, GET, and POST (see **Figure 10**).

Final Thoughts

Ordinarily, audio laboratories are filled with dedicated hardware for measuring all manner of electroacoustic parameters. It is atypical, however, for the ratio of gear to engineers to be 1:1. That means there is often a backlog or wait list for access to lab equipment. For critical measurements for data sheets or characterization before sending a sample out to customer or certification, the proper paradigm is the utilization of the dedicated hardware systems.

But if you need to check a new sample and provide same-day feedback, or make measurements in the field or a home office, then the opportunity to have an audio test system in the cubicle or desk of every engineer becomes compelling. This appears to be the raison d'etre of the QA401 and the QA451. These are powerful and compelling tools that enable and accelerate the development process.

In a world of increasing measurement demands, decentralizing the test platforms and deploying them throughout a team is a positive productivity enhancement. And while a lab full of dedicated analyzers and instruments is still a must, getting more tools out to all of the team members is a recipe for organizational success.

Whether a development engineer needs to quickly address the latest iteration of a new power amplifier, or a production engineer needs to develop and scale a production-ready test solution without the support of a software and firmware team, the combination of the QA401 and the QA451 offer enticing capabilities.

From the core application functionality to the availability of Test Plugins, production tools for automation, and free cloud storage for data access, the combination of the QA401 and the QA451 resonate with a satisfying sense of cohesion, thanks to the well-rounded system-level architecture and product support. Too often, test and measurement are not given the attention that they deserve in corporate environments, due to the common constraints of time and money.

This problem is particularly notable on the production test floor, where speed, accuracy, and cost are the critical factors, beyond any advanced capabilities that might be needed for product design, development, or troubleshooting. If you are looking to save money, minimize setup times, and remove obstacles to shared resources in the laboratory and production environments, QuantAsylum has developed an impressive toolbox of hardware and software.

Resources

"QA401 Quick Start," QuantAsylum, https://cdn.shopify.com/s/files/1/1631/5609/files/QA401_Quick_Start_RevB.pdf?v=1588785095

"QA401 Product Brief," QuantAsylum, https://cdn.shopify.com/s/files/1/1631/5609/files/QA401_Product_Sheet_RevD.pdf?85

"QuantAsylum QA401 User's Manual," QuantAsylum, https://cdn.shopify.com/s/files/1/1631/5609/files/QA401_Users_Manual_0ea33c5f-a8c5-449a-92d3-7927893485a9.pdf?v=1588785336

"QuantAsylum QA450 User's Manual," QuantAsylum, https://cdn.shopify.com/s/files/1/1631/5609/files/QA450_User_s_Manual.pdf?12998937473040546318

"QuantAsylum QA450 Product Sheet," QuantAsylum, https://cdn.shopify.com/s/files/1/1631/5609/files/QA450_Product_Sheet_RevB.pdf?4450960313629741042

"Introducing the QA451," QuantAsylum (July 22, 2020 blog post), https://quantasylum.com/blogs/news/introducing-the-qa451?_pos=3&_sid=c3e3f6935&_ss=r

"QA401, REST, and Python," QuantAsylum (May 17, 2019 blog post), https://quantasylum.com/blogs/news/qa401h-rest-and-python?_pos=1&_sid=2936e996c&_ss=r

"Automated Amplifier Testing with Tractor," QuantAsylum (April 20, 2019 blog post), https://quantasylum.com/blogs/news/automated-amplifier-testing?_pos=2&_sid=92631868a&_ss=r

"QA401 Headless & Linux Support," QuantAsylum (December 6, 2018 blog post), https://quantasylum.com/blogs/news/qa401-headless-linux?_pos=1&_sid=c557a7254&_ss=r

"First Look: Evaluating Class D Power Amps with the QA401," QuantAsylum (April 30, 2016 blog post), https://quantasylum.com/blogs/news/first-look-evaluating-class-d-power-amps-with-the-qa401?_pos=1&_sid=f13498c85&_ss=r

Mastering Wireless Multi-Tone Testing

This article explains the issues resulting from measuring a wireless audio connection and how wireless multi-tone testing can help us to overcome those obstacles.

By
Andrew Taylor
(Klippel GmbH)

Standard audio testing is dominated by classic sine wave stimuli such as stepped-sine, chirp, and two-tone measurements. While there is always a place for these measurements, multi-tone testing has many advantages, and while not new, it could be considered overlooked or underutilized. Nowadays, more and more audio devices include a wireless connection. Unfortunately, this wireless connection introduces new challenges and issues that are not present in a classic wired connection. For manufacturers and designers of these modern audio devices, it is imperative that audio test and measurement solutions overcome these issues and provide accurate and reliable results.

Multi-Tone Testing

Multi-tone testing, which has been around for many years, starts with the generation of a sparse stimulus consisting of multiple tones, which will be expanded upon later in the article. The response is then captured and analyzed. The post-processing includes the separation of the bins containing energy corresponding to excited frequencies of the stimulus. The remaining bins contain the audio system's distortion plus all noise. This process is illustrated in **Figure 1**.

Stimulus

The multi-tone stimulus, such as defined by IEC 60268-21[1], consists of three or more tones, each at a unique and fixed frequency. As seen in **Table 1**, the crest factor for the multi-tone signal is much larger than the sine wave stimuli. In order to compare signals, crest factor, or kurtosis, which is a measure of the probability of any given data point being very close to the mean value vs. very far away from the mean, can be used. In both cases, a higher value means that the signal energy is more strongly concentrated around the mean and less at the peaks. However, kurtosis is preferred to crest factor because it provides similar information, but is a more robust statistical characteristic. This is illustrated in Table 1, where pink noise theoretically has an infinite crest factor.

In real systems, there is always an upper bound. However, the crest factor of a real system is less reliable because it is dependent on the peak value of the analyzed signal, which may be changing over time and could be different depending on the length of the captured signal. When comparing kurtosis values and stationarity, the multi-tone stimulus is very similar to pink noise.

The pseudo-randomness of the multi-tone stimulus is also important. For a noise-like signal, the phase of the frequency components should be randomized. However, the exact properties are based on a few input (seed) parameters, which ensures that the exact stimulus can always be repeated, leading to reproducible and comparable measurement results. The signal is also flexible depending on the particular measurement application.

The stimulus is usually defined by a frequency range with corresponding amplitude and density, or resolution of excited bins, with, for example, the unit tones per octave. Frequency dependent amplitude shaping can also be applied so that, for example, it has the spectrum of typical program material as defined in IEC 60268-21.

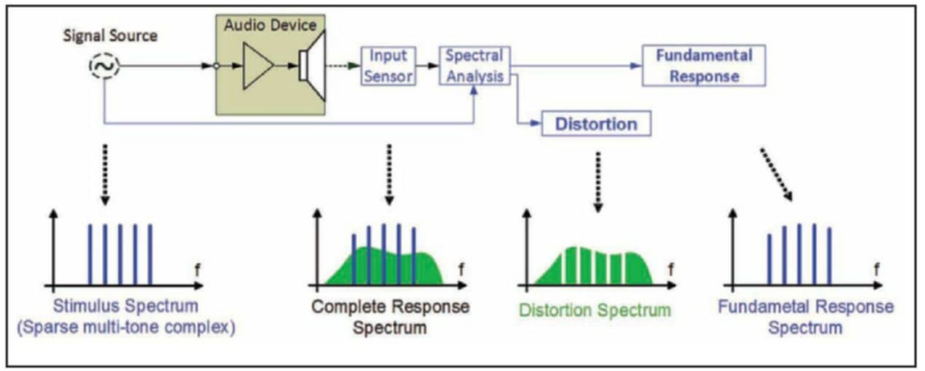


Figure 1: Overview of the multi-tone testing method

Advantages

As previously discussed, the sparse spectrum of the multi-tone stimulus makes it possible to separate fundamental components from distortion and noise components. Therefore, both the fundamental frequency response as well as a comprehensive distortion evaluation of the audio device can be extracted. This distortion measurement includes both harmonics and intermodulations and can be thought of as a distortion "fingerprint."

When applied over a sufficiently wide frequency range, with a sufficient density (e.g., ≥ 10 tones per octave), and with enough amplitude (large signal domain), this stimulus activates all transducer nonlinearities, causes typical heating and force convection cooling and activates DSP protection features (limiter, compressor, etc.), if present.

In the end, the multi-tone measurement gives some great practical benefits. It can be used to quickly evaluate both an audio device's fundamental and distortion responses in the same measurement and can also be used in long-term power and life testing. When used to compare and benchmark different audio devices, it gives more realistic and complete results than through sine wave testing alone. Multi-tone testing is also recommended for several aspects of IEC 60268-21 including rating maximum SPL, 100- hour testing, and input calibration.

Disadvantages

The comprehensive distortion evaluation of the multi-tone measurement gives a distortion curve in which the individual components cannot be separated. This means contributions from harmonics cannot be differentiated from intermodulations. Likewise, even-order and odd-order distortion components cannot be distinguished, so there is no indication of whether the nonlinearities are symmetrical or asymmetrical.

Finally, the multi-tone measurement is an energy-based measurement. Therefore, it is not very sensitive for impulsive distortion, also known as rub and buzz. For sensitive impulsive distortion detection, time domain analysis is recommended [2].

Wireless Multi-Tone Testing

While the benefits of multi-tone testing are apparent, testing modern audio devices with a wireless connection generates several additional issues that need to be overcome for accurate results.

One issue found in a wireless connection that is not present in a classic analog connection is a larger latency or delay. Depending on the specific type of wireless connection and the total setup, delays could be larger than 0.5 seconds. And while

	Full-Band Crest Factor	Full-Band Kurtosis	Stationary	Sparse
Chirp	3 dB	3.5 dB	No	Yes
Steady-State Sine	3 dB	3.5 dB	Yes	Yes
Pink Noise	∞	9.5 dB	Yes	No
Typical Music	>18 dB*	10 dB*	Yes	No
M-Noise [4]	18 dB*	10 dB*	Yes	No
Multi-Tone [1]	12 dB	9.5 dB	Yes	Yes
*Frequency dependent				

Table 1: Comparison of several common audio test stimuli

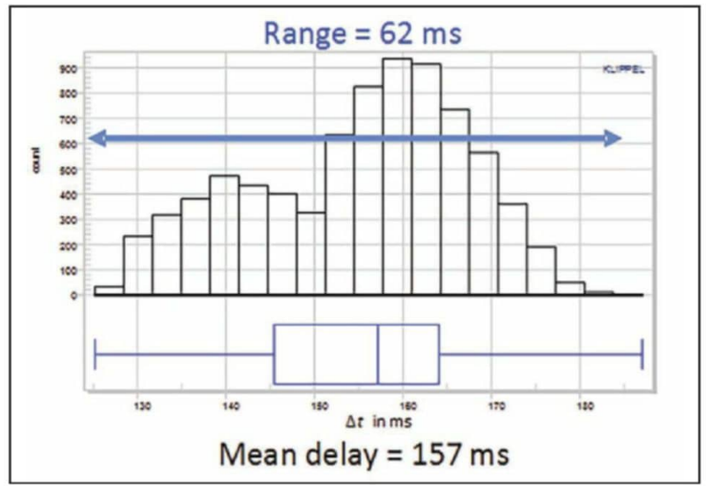


Figure 2: Distribution of delay from a wireless connection to Bluetooth headphones after re-initialization

a constant and stable delay would be easier to deal with, the delay of wireless connections can vary significantly. Marian Liebig previously investigated the delay of a Bluetooth audio connection after 7,800 re-initialization processes, with the results shown in **Figure 2** [3]. With a mean delay of 157 ms, the delay has a range of 62 ms, which corresponds to an acoustic delay with a range of more than 21 meters.

Another issue is sampling jitter, which is generated by a mismatch between the sampling rates of the audio device and the measurement device(s). This causes more severe problems for multi-tone testing than delay issues. This same issue can sometimes be seen when testing an audio device using an analog connection if the stimulus generator and the analyzer capturing the response do not share a common clock and are not completely synchronized. The practical consequence is a frequency transformation of signals, leading to the response being either slightly shorter or longer than the stimulus. If the response is shorter than the stimulus, the response contains higher frequencies (shift upward). Likewise, if the response is longer than the stimulus, the response contains lower frequencies (shift downwards). This leads to a non-coherent relationship between stimulus and response that rises with frequency, degrading the accuracy of the measurement results. This dynamic frequency transformation is a frequency modulation similar to the Doppler effect but more random. While this generates distortion, it only has a minor effect on perceptual audio quality, unlike amplitude modulation, which may cause fluctuations and roughness [5]. Although these effects are less perceptually relevant, they can cover up the more critical distortions of an audio device if the measurement system does not compensate for them. There is a lower impact on the fundamental frequency response but a high impact on the distortion and phase measurements, which are especially corrupted at high frequencies (≥ 1 kHz).

Solutions

Dealing with delay is not a new problem in audio measurements. For example, acoustic delay, dependent on the distance between the audio device and the microphone, must be considered. However, the delays in a wireless system can be much longer or vary much more than normal electrical or acoustical delays. In general, there are a few conventional ways to deal with long and/or varying delays. The first is using a pre-loop, or an iteration of the stimulus, before the actual stimulus that is used to capture the response. If the stimulus length is longer than the longest expected delay, then a single pre-loop is sufficient. If the stimulus is very short or the delay is very long, more pre-loops can be added.

Klippel's R&D multi-tone measurement module MTON, like most Klippel modules, allows for the addition of pre-loops. Other common methods of overcoming these delays, which are faster and may be required for production line testing, include finding the energy peak in the signal's impulse response or using a synchronization signal such as Klippel's External Synchronization (SYN) software add-on for use with Klippel's QC (end-of-line) testing suite [3].

Sampling jitter is also not new in the context of audio because it is inherently present in all audio systems that incorporate a clocked device such as an analog-to-digital or digital-to-analog converter. However, these effects are usually negligible when assessing the audio quality of most complete electroacoustic systems with a wired connection. However, when using a wireless connection, these effects can become much more severe and may corrupt measurement results. It is good practice to avoid long measurements, because the coherence between the stimulus and the response usually becomes worse over time. Also, averaging should be avoided because the sampling jitter is changing over time, meaning that it is unlikely that consecutive responses will have the exact same clock relationship and thus sampling jitter effects. However, even if long measurements and averaging are avoided, the results still may not be accurate.

A more robust technique to reliably and accurately overcome sampling jitter effects is needed. Klippel's MTON solves this by implementing a "Clock Drift Tolerance" feature. By activating this, the fundamental components, which undergo a smearing effect due to the sampling jitter, can be correctly identified and separated from the distortion components to provide accurate fundamental frequency and total distortion responses. **Figure 3** shows differently processed distortion curves of multi-tone measurements using MTON on a Bluetooth audio device over a wireless connection. The red curve did not compensate for the sampling jitter, and the frequency modulation caused by the wireless connection is interpreted as distortion. The blue curve used the "Clock Drift Tolerance," correctly identified the dominant distortions generated by the loudspeaker in the audio device, and is indistinguishable from the audio device's distortion measured with a wired (analog cable) input connection.

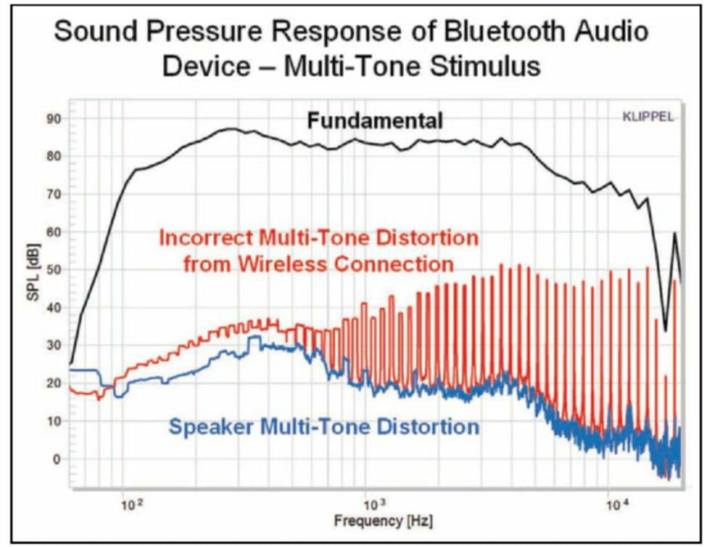


Figure 3: The response curves of a correctly processed (blue) and an incorrectly processed (red) multi-tone measurement for a wirelessly connected Bluetooth audio device

Conclusion

Multi-tone testing has many advantages and is a fast and powerful tool for evaluating overall performance and audio quality. And because of the increasingly wireless nature of modern audio devices, audio test and measurement solutions must be able to reliably overcome the issues generated by a wireless connection. Klippel's MTON method can quickly and accurately assess the fundamental and distortion responses of an audio device, even if a wireless connection with long and varying delay and sampling jitter is used.

Klippel has a wealth of freely available information regarding multi-tone testing, testing over a wireless connection, benchmarking, and much more. See the Resources for several relevant links including application notes, web-seminars, and measurement modules.

About the Author

Andrew Taylor received his bachelor's degree in electrical engineering at the University of Kansas in Lawrence, KS in 2015. As he finished his master's in electrical engineering with a concentration in musical acoustics and signal processing at the University of Rochester in Rochester, NY, he interned at Klippel GmbH in 2016. He has been a full-time employee at Klippel since 2017, where he now splits time between audio analyzer hardware development and technical writing and content creation.

References

[1] IEC 60268-21, "Sound System Equipment—Part 21: Acoustical (Output Based) Measurements," International Electrotechnical Commission (IEC), 2018.

[2] W. Klippel, "Measurement of Impulsive Distortion, Rub and Buzz and other Disturbances," presented at the 114th Convention of the Audio Engineering Society, March 2003, preprint 5734.

[3] M. Liebig, "Challenges of Testing Mobile Devices and Mobile Testing," *Voice Coil*, January 2017.

[4] R. Schwenke, "A New Signal for Measuring Loudspeaker Maximum Linear SPL", SMPTE 2019 Annual Technical Conference, Los Angeles, October 2019 (submitted).

[5] E. Zwicker and H. Fastl, *Psychoacoustics—Facts and Models*, Springer, Berlin, 1999, ISBN 3-540-65063-6

Resources

AN72, Testing Wireless Audio Devices with Klippel R&D System, Klippel application note, www.klippel.de/know-how/literature/application-notes.html

AN73, QC Headphone Testing, Klippel application note, www.klippel.de/know-how/literature/application-notes.html

AN76, QC Testing of Wireless Audio Devices, Klippel application note, www.klippel.de/know-how/literature/application-notes.html

External Synchronization (SYN) Option for Klippel QC System, www.klippel.de/products/qc-system/modules/syn-external-synchronization.html

Klippel QC System, www.klippel.de/products/qc-system.html

Klippel GmbH, Multi-Tone Distortion, www.klippel.de/know-how/measurements/nonlinear-distortion/multi-tone-distortion.html

Mutli-Tone Distortion (MTD) Option for Klippel QC System, www.klippel.de/products/qc-system/modules/mtd-multi-tone-distortion.html

Multi-Tone Measurement (MTON) for Klippel R&D System, www.klippel.de/products/rd-system/modules/mton-multi-tone-measurement.html

Session #5: Maximum SPL – Giving This Value Meaning, Acoustical Measurement of Sound System Equipment according to IEC 60268-21, KLIPPEL LIVE – Web-Seminars, www.klippel.de/know-how/education/klippel-live-web-seminars/iec-60268-21.html

Session #7: Amplitude Compression – Less Output at Higher Amplitudes, Acoustical Measurement of Sound System Equipment according to IEC 60268-21, KLIPPEL LIVE – Web-Seminars, www.klippel.de/know-how/education/klippel-live-web-seminars/iec-60268-21.html

Session #9: Intermodulation Distortion – Music is More Than a Single Tone, Acoustical Measurement of Sound System Equipment according to IEC 60268-21, KLIPPEL LIVE – Web-Seminars, www.klippel.de/know-how/education/klippel-live-web-seminars/iec-60268-21.html

Session #11: Pitfalls in Testing Wireless Audio Devices, Acoustical Measurement of Sound System Equipment according to IEC 60268-21, KLIPPEL LIVE – Web-Seminars, www.klippel.de/know-how/education/klippel-live-web-seminars/iec-60268-21.html

Session #12: Benchmarking of Audio Products Under Standard Conditions, Acoustical Measurement of Sound System Equipment according to IEC 60268-21, KLIPPEL LIVE – Web-Seminars, www.klippel.de/know-how/education/klippel-live-web-seminars/iec-60268-21.html

The Puffin Phono DSP by Parks Audio

By
Mike Harkins
(United States)

In this product review article, Mike Harkins puts the Puffin, a next-generation phono preamp with a wide range of gain settings, under the microscope. This design by Shannon Parks adds a programmable DSP and truly takes the concept of a phono stage to a different level.

Photo 1: The Puffin is a next-generation phono preamp with a programmable DSP to allow a wide range of gain settings.

I had already completed this article on the Puffin Phono DSP when I found out that it was almost completely obsolete. Did I get a new, improved device from the manufacturer? Nope. A firmware update, which took about 10 minutes, was all it took to do the trick. And any future updates will be just as easy and with no investment in hardware required.

The business model for Shannon Parks, founder and chief engineer at Parks Audio LLC states: "Buy my phono stage, and it will be updated and improved constantly with simple firmware updates." However, the upgrades were substantial, so I felt it best to start from scratch and re-evaluate the Puffin.

I knew that this was a possibility when I purchased this phono preamp last year. But I was surprised at how drastically the function and performance of this device could change with a single firmware update. There were six additional controls added and several improvements to the 16 previous ones.

The Puffin

But let's start at the beginning. The Puffin is basically a phono preamp designed to provide gain and equalization for any phono cartridge plugged into it, whether moving magnet, moving iron, or moving coil (see **Photo 1**). The purist approach would be to design a tube or solid-state phono preamplifier with components specified to reproduce a flat frequency response with low noise and distortion and good matching of impedance for cartridge loading and output. In some cases, however, the end user would be unsatisfied with such a design. For example, the preamp's gain may not be adequate for the cartridge output. This is especially important if the user wants to transfer records into a digital format. Inadequate gain would cause a recording to be made at suboptimal volume levels decreasing the potential signal-to-noise available in a digital recording and reducing the overall volume compared to other digital audio files.

In the past, personal experience with this problem caused me to resort to supplemental amplification to achieve good recording results with my system. Many inexpensive (less than $500) phono stages don't have moving-coil (MC) cartridge capability and even moving-magnet cartridges and high output MC cartridges have many different output levels. Even if a phono stage has a gain control, it may still not have sufficient gain for a given cartridge.

What about equalization? Most phono preamps do an adequate job applying RIAA equalization to the signal path. But some records that are overly accentuated in the high-frequency area, or whose bass has been unduly compressed for vinyl recording may need some help. Or it might be that a vintage record is somewhat noisy or worn and would benefit from some high-frequency noise attenuation. Some listeners, who like that warm tube sound, would love to have a preamp that delivers such sonic joy to the ears. Collectors of antique records or 78s may also experience issues with properly equalizing them for the best sound. The same goes for anyone who likes to swap cartridges occasionally. And wouldn't it be nice to have the ability to grade records as to their quality and to improve their sound by diminishing the noise typically found on vinyl records?

To address these issues, the Puffin Phono DSP by Parks Audio takes an entirely different approach. It sells for $489 online from the company's website (http://parksaudiollc.com). Purists might cringe at the idea of converting analog to digital, but that is exactly what this phono preamp does. It is, in fact, a digital signal processor (DSP). It takes audio from the cartridge after one stage of amplification and converts it to digital through an onboard analog-to-digital converter (ADC) operating with 24-bit resolution and 96 kHz sampling rate. The signal is filtered in the digital domain using digital signal processing to render the correct RIAA equalization and then it is converted back to analog and amplified again with a TI digital-to-analog converter (DAC) before being sent to unbalanced RCA output connectors (see **Figure 1**). Parks Audio also offers a DIY modification, which enables users to use their own DAC if desired.

The Puffin's Features

Before discussing performance, let's look at the extreme versatility of this product. The Puffin is truly the "Swiss Army Knife" of phono preamps. In fact, it isn't even strictly a phono preamp. If you look at the introductory video on its website, you will see what I mean. But meanwhile, let me enumerate and comment on what this thing does by reviewing its mechanical features and control functions, some of which were recently enabled or improved with the new firmware update (v. 1.20):

- Typical "on-off" switch, which is called a "Standby" button. User presets are stored.
- Rotary selection switch (Knob) which allows a multitude of adjustments to the DSP preamp.

Once a selection is made (see "Select (arrow) button"), rotating the knob will allow selection of any of the items under that menu. A list of the various numbered menu items, which can be adjusted once selected include:

Volume: This is typically set to maximum (100) but can be adjusted to any level. Users can set this control to mute between records with an additional clockwise click from 100, or to adjust playback volume to any level desired by rotating counterclockwise to the desired level.

Gain: This control should be set to the level suitable for your cartridge or input device. The Parks Audio website has recommended settings for many popular cartridges. This is an extremely flexible control allowing from -4 dB to +72 dB of gain. In addition, a tape deck, CD player, or other source can be attached with the gain set at one of the lower settings (-4 dB or 0 dB). This is a remarkable amount of gain flexibility unmatched by nearly anything I have seen.

Tilt (see **Figure 2**): A tone control hinging at 800 Hz, which increases treble while concurrently decreasing bass in 0.5 dB increments or vice-versa, resulting in subtle to somewhat dramatic changes to the output. It is a quick way to change overly bright or overly warm sound coming from phono playback. You can shave over-emphasized high frequencies from the response of some cartridges, or you can

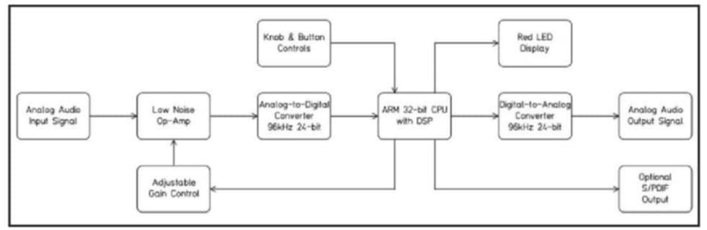


Figure 1: Puffin DSP Phono Stage block diagram. The signal is filtered in the digital domain using digital signal processing to render the correct RIAA equalization and then it is converted back to analog and amplified again with a TI digital-to-analog converter (DAC) before being sent to unbalanced RCA output connectors.

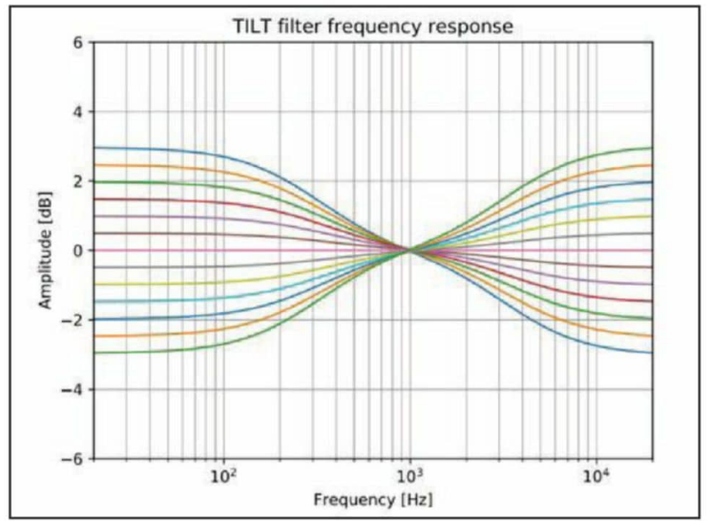


Figure 2: This graph shows the effects of using Tilt, a tone control hinging at 800 Hz, which increases treble while concurrently decreasing bass in 0.5 dB increments or vice-versa. Note that the "Tilt" hinge point was changed slightly to 800 Hz instead of the 1,000 Hz hinge point.

brighten a dull high-frequency response. The steps of adjustment, as shown in the response curves for Tilt, are just 0.5 dB per step. However, the ear is quite sensitive to even slight changes in response over a broad range of frequencies. Thus, it is advisable to use this diligently. Tilt causes the low frequencies to attenuate as the highs are emphasized. Thus, if one tries to emphasize the highs, the lows may be cut more than desired. However, the Bass and BassBoost control can make the sound right again if needed.

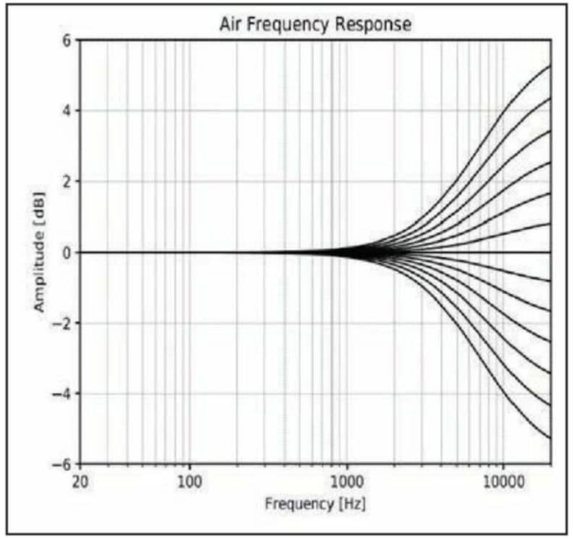


Figure 3: Air is a tone control that adjusts the highest frequencies (mostly above 8 kHz) in increments of about 1 dB at 20 kHz.

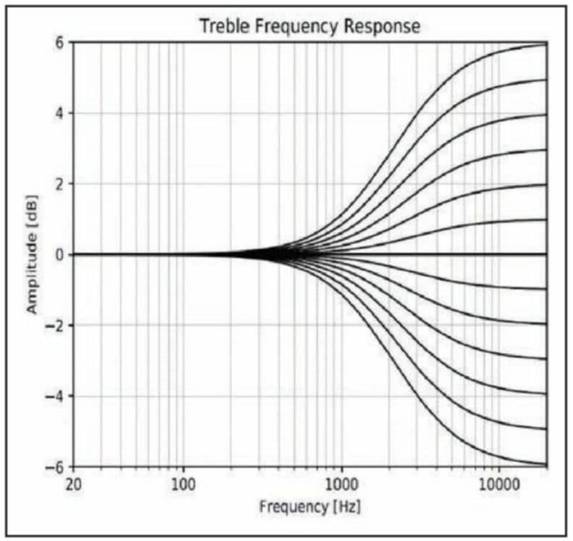


Figure 4: This graph shows the effects of using Treble, a tone control which operates in the broad band mainly above 3,000 Hz.

Air (see **Figure 3**): A tone control that adjusts the highest frequencies (mostly above 8 kHz) in increments of about 1 dB at 20 kHz. In addition to improving the sound of some recordings that lack those super high frequencies, or which are painful to listen to because of too much high frequency energy, the Air control is best used to tame phono cartridge anomalies. Some phono cartridges and their loading can detract from, or over-emphasize the detail found at the highest frequencies of the audio spectrum. The Air control is used to make fine adjustments which can reduce the very high frequencies or alternately, emphasize them when needed. Since the Puffin does not offer multiple loads for MC cartridges beyond the standard 200 Ω, some cartridges may benefit from adjusting Air slightly to accommodate any loading mismatches. Notice that Air has practically no effect on fundamentals, which are mostly below 4 kHz, yet high frequency harmonics and the "air" around cymbals and percussion instruments can be affected by this control.

Treble (see **Figure 4**): A tone control which operates in the broad band mainly above 3,000 Hz.

Bass (see **Figure 5**): A tone control that operates in the broad band mainly below 300 Hz.

Bass Boost: Adds (or subtracts) additional bass with a corner setting of 100 Hz, based on the setting of the bass control. Bass boost is set to off but can be turned on by the user.

Hi: The high-frequency cut-off is selectable using this control. It is a fourth-order Butterworth filter. This control is particularly useful at reducing noise on old LPs and 78s. Some recommended settings and experimental settings are given in the manual. It can be set to any of 12 different settings from 5 kHz to 48 kHz with default at 20 kHz.

Lo: This is also a fourth-order Butterworth filter with selectable cut-off frequencies. It can be useful in eliminating such noises as rumble or low-frequency effects caused by record warps. Its default is 25 Hz, but it can be set to any of 14 settings from 5 Hz to 500 Hz.

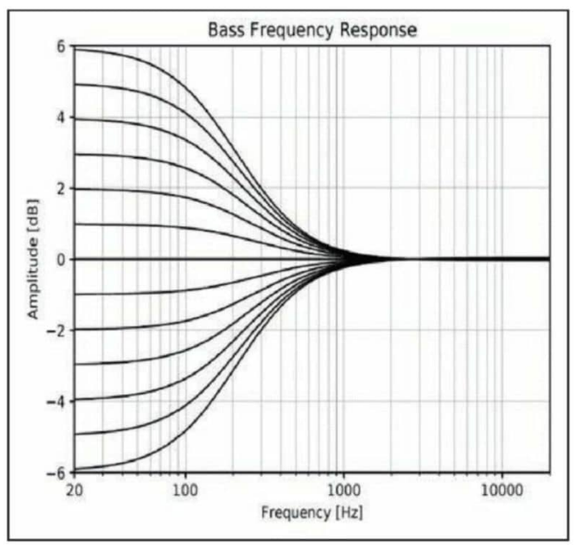


Figure 5: Here, the effects of Bass, a tone control that operates in the broad band mainly below 300 Hz, are shown.

Mode: Select Stereo, Mono (sum of left and right), Left, and Right. Recommendations for use of this control are also in the manual.

Grade: This is a new function of the Puffin. It enables the user to determine the quality of the recording being played. The Puffin uses a click detector, which finds suspected noise of this type on the recording. The Grade function then counts the number of these defects encountered after it is activated. It determines a letter grade, from A+ to F, just like you got for your homework in school, for quality of the record. It looks for an average over time, so it's best to engage after the record starts playing and to stop it just as the record ends. You can watch the grade change as the record plays, getting better or worse during playback.

Magic: Another new function, Magic is a means to improve the playback sound of your records by eliminating or reducing those pops or ticks used to Grade the record. It uses an algorithm to determine if the detected click is noise or part of the musical program to decide whether to apply a mild corrective change to the waveform. It is recommended to leave this function on since it should not audibly detract from any musical signal.

fx: This control enables the user to simulate a sound like that of tube amplifiers or tape decks. This is interesting to play with and could be helpful with some recordings or if you really like that warm tube sound. Tubes generally produce even-order harmonic distortion, with a dominant second harmonic. Setting the Puffin to "tube" adds a significant amount of second-harmonic distortion, which is what consensually generates a more "tube-like" sound. Tape decks tend to accentuate third harmonics and other odd harmonics for a slightly harsher sound and the Tape setting simulates this effect. Another setting allows the generation of pink noise. This enables the user to test room characteristics using the Puffin as the source and an appropriate real-time analyzer. Users might find this to be a useful control, which can help determine the best setting for the RoomEQ function.

DeRumble: This is a newly added function that converts frequencies in the bass range (below 200 Hz) into mono when mode is set to stereo. This function is said to eliminate rumble, warps, footfalls, undesired mechanical feedback, and other undesired artifacts from record playback. It is recommended for thin and warped LPs, which would otherwise

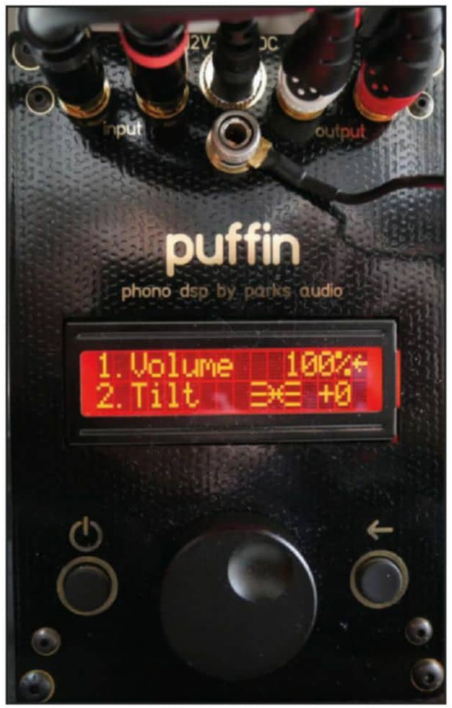


Photo 2: The Puffin has three controls: the Standby button, the Knob, and the Select (arrow) button. The Standby turns the Puffin on and off. The Knob quickly rotates through the settings. The Select button chooses the function or setting.

suffer from degradation due to undesirable vertical movement of the stylus during playback.

RoomEQ: This is another new function for the Puffin. It attempts to eliminate room peaks that might plague the listening experience in many listening rooms. It places a narrow-band 6 dB cut filter at the default setting of 300 Hz or at any of the other selectable frequencies from 200 Hz to 400 Hz. There is an "off" setting between each selection so that the user can easily see the "before-and-after" effect of using this control.

Load: This is the Puffin's input impedance and is normally set to 47 kΩ. Low output moving-coil users can reset this load to 200 Ω. It is recommended that subtle adjustments for MC cartridges be done using the extensive set of controls, primarily Air and Tilt, for cartridge compensation.

Balance: Provides balance control between left and right channels in 2 dB steps.

Fine Balance: Adjusts the left channel in ± 0.3 dB steps to match the right channel.

Phase: Has four phase settings, Normal, Invert, Mixed, and Null. Normal is the same phase as the incoming signal. Invert reverses the phase on both channels. Some experts believe that proper absolute phase is needed to experience the best effect for stereo listening and this control enables users to easily do that experiment. Mixed phase has the two channels 180° out of phase with each other and is useful for determining good stereo imaging (the more diffuse the better). Null subtracts the two channels from each other and can help set "Fine Balance."

Out: This menu option switches between the usual analog RCA outputs and an optional digital coax SPDIF output for those wishing to use their own DAC with the Puffin.

EQ: This is an equalization setting control that sets the Puffin to the standard RIAA equalization or to any of several selectable standards. "Line In" is a flat response for use with CD players, tape decks, and the like. Enthusiasts with older format records are welcome to play in this sandbox of 12 selectable EQ standards.

Set: This control is a preset gain control that has four factory settings for two different cartridge input types, computer audio input, and CD player input. There is room for four additional user settings which can be saved as presets.

Two additional menu items on the front panel display the signal levels for each channel into the ADC and the DAC. They can be set to display real-time average signal levels over a non-specified, but brief period, or peak signal levels, which once displayed stay there unless a higher peak is detected. Although, it is important to make sure that the peak of 0 dB is not exceeded, the Puffin can handle minor clipping without adverse effects. A newly added feature displays an asterisk next to the Gain and Volume control settings if any clipping is detected. These controls should be reduced if clipping is a problem.

Continuing with the mechanical and display items on the hardware (see **Photo 2**):

- Red display showing each of the mentioned functions and allowing selection and adjustment of each.
- Select (arrow) button: This button enables the user to select items from any of the above functional menus for adjustments and settings.
- Input and output RCA receptacles and a ground wire attachment lug for use with a turntable.

Performance

Designer Shannon Parks has the engineering chops to design a preamp like this. He got his start making DIY kits part-time, but now dedicates all his time to designing and building electronic products for audio enthusiasts. One of his first such products is this Puffin DSP preamp. He is no stranger to tube amplification either, having developed several amplifiers and preamps with low-noise tube stages. The Puffin Phono DSP by Parks Audio looks to be a well-designed high-quality product. The enclosure is a solid metal case, which should hold up well under normal operating conditions. If you use the Puffin, you will undoubtedly be handling it a lot. The power supply is a Phihong 15 W, 12 V switcher mounted in-line with the line cord, providing a steady source of power for the Puffin.

Performance specifications, such as frequency response, channel separation, signal-to-noise ratio (SNR), distortion, and other technical information are omitted from the specs. However, most of those are dependent on the performance of component devices, such as the Texas Instruments ADC and DAC and the ARM DSP.

That said, flat frequency response is easily achieved by most phono preamps and the distortion levels are usually well below audible levels. And, of course, the Puffin enables the user to make fine adjustments to compensate for cartridge deficiencies. One can ascertain approximate SNR by looking at the ADC and DAC signals on the display. On my system with the turntable and cartridge plugged in, I get a reading of -86 dB at the ADC input and -81 dB at the DAC input. This indicates that minimum SNR is approximately, 81 dB. This is quite good considering most LPs cannot do more than about 40 dB to 50 dB and the absolute best one could hope for is about 60 dB or so on the finest vinyl.

Setup

The Puffin is easy to set up. Simply plug in the RCA phono leads from the turntable and attach the ground wire securely (if so equipped). The output is also a plug-in from unbalanced RCA connectors to your amplifier, external preamp line input, or other device. Then plug in the power supply to a wall outlet and turn on the push button power switch. The Puffin will power on with a display showing the firmware level stored in the DSP memory. If not the latest, as can be found on the Parks Audio website, it can be easily updated. Currently, I'm using the Puffin for recording vinyl records into FLAC files for later playback on my hi-res player.

Once the Puffin is turned on, it boots quickly to a screen as shown in Photo 2. The various settings can be accessed via the rotary selection switch knob on the unit.

Functional Performance

My current setup uses the Puffin DSP preamp with a Clearaudio Concept turntable with a Satisfy tone arm and a Hana EH high output moving coil cartridge. I use this as input to Puffin at 48 dB gain, which was the recommendation for this cartridge. The Puffin's output is connected to my PC using a Creative Labs Sound Blaster ZX as a recording device. When used in conjunction with VinylStudio software, I can record with a nearly silent noise floor. Using this software's click removal function nearly eliminates all pops and clicks from the recording, making this the closest to the original master tape that you can get in any vinyl recording process. But I noticed that VinylStudio software is not perfect. It can eliminate some high-frequency content, particularly that of synthesizers and some percussion and brass instruments. So, it will be interesting to see if the new "Magic" feature of the Puffin can do nearly the same thing without eliminating such artifacts.

I did not evaluate some controls, such as the Mode control. Controls such as this might be useful to some users, but I would only use Mode, EQ, Hi, and Lo settings, for special needs or experimentation. Some users, for example, might want to eliminate excessive noise in badly worn recordings or perhaps to balance a noticeably different left and right channel output.

The Puffin's noise floor is quite low. I could detect no hum or other extraneous sound when program material was not present. Adjustment of any of the settings did not affect the noise floor of the device. I could only hear some slight hum and noise when the Puffin, Sound Blaster, and my computer were set to maximum levels ... obviously not suitable for playback without destroying transducers and/or one's hearing. Subjectively, the adjustments that I made in testing the device seem to do just what they are intended to do.

Sound and Listening Performance

There are so many possible functions and settings for the Puffin, that to do listening tests at every setting would be nearly impossible. So, I will only comment on a few tests that I have personally done and give a general overview of how well the Puffin works in my system. Listening tests are a good way to ascertain the effects of each control on the Puffin. Some are quite subtle, but when used to extreme, can be quite pronounced in affecting the overall sound. To get the best impression of how the system sounds, I'm using a simple pair of headphones, Grado SR125. The advantage of headphones is that they provide the ability to hear very subtle details without room effects disturbing audibility.

Subjective tests on musical program material are a good way to evaluate how well a product, such as this Puffin, performs. This is especially true if one already knows what the cartridge and other gear sounds like with the program material in use. With that in mind, I recorded and monitored a couple of pieces of program material that I thought would provide some insight into the Puffin's performance. I used John Klemmer's "Man and Woman" from the

Finesse LP and Jethro Tull's "Locomotive Breath" from the *Aqualung* LP to test some of the specialized tone and equalization controls.

I listened to John Klemmer's recording with several settings of tilt, air, and fx set to "tube," and "tape." I have listened to this recording often and I think the sax comes across a bit bright and frankly, too smooth. This may be partially caused by the reverb effects. The track, "Man and Woman" has a variety of percussion instruments as well as significant bass, enabling me to evaluate much more than the solo sax's sound reproduction. Tilt, when set at +2 brightened the sound a bit and brought out some tinkling of the chimes but when set at -2, the effect boosted bass a bit too much for my taste without really detracting too much from the percussion.

When I listened on speakers, I found the -2 Tilt to be quite enjoyable on this recording. The Air control also brightened the upper registers a bit, but not obtrusively. Cymbals and percussion instruments sounded quite natural to my ears. The fx control has a fascinating effect. When I turned to the tube setting, the sax got warmer and smoother with little effect on percussion or bass. Trying the tape setting caused the sax to become a bit more raucous, but not to the point of being annoying. You can play with this to your heart's content, creating a subtle and often pleasing difference in the sound.

I didn't try the fx settings with Jethro Tull's recording, preferring to stick with the Tilt, Air, and Bass controls. I did this because this recording has some aspects that seem useful for testing the Puffin. For example, dynamic range is quite large for a rock recording. The song "Locomotive Breath" starts out with an unusually quiet piano passage that eventually explodes into heavy rock guitar band sound. I found this quiet piano passage to be noise free except for a slight vinyl whoosh that told me I was listening to real vinyl. Ian Anderson's flute and vocals come across well, but the vocals and percussion seem a bit dull on this recording. Sibilance in the vocals seem almost nonexistent and some percussion instruments, like the tambourine near the end of the song, seem way in the background and almost sonically invisible. Of course, the producer and the recording engineer had a lot to do with this, but I wanted to see if I could undo some of the things they did to make the recording sound this way.

I played the song with four different settings of the controls:

- Flat (phono RIAA with no adjustments of other controls)
- Tilt: +2, Bass: +2;
- Tilt: +2, Bass: +2, Air: +3
- Tilt: +4, Bass: +2, Air: +3

The first two settings sounded nearly identical to my ears. The kick in Bass response offset the negative bass effect of the tilt control so that no significant change in the fundamental frequencies was evident. A very slight, but nearly imperceptible change in highs could be heard with careful listening. Moving to setting 3 also made little difference to my ears except for a slight improvement in sibilance and a little more audible visibility of the tambourine. Finally, boosting the tilt control a bit more, as in setting number 4 made the difference a little more apparent. I could hear considerably better sibilance, and the tambourine was starting to come out of the background. Even then, I couldn't hear much difference in the sound of Anderson's flute except that the vocalized breath effects he used were more apparent. Little change in bass response was evident with this setting, and honestly, I thought the last setting made this recording more enjoyable and natural to me. Of course, maybe that setting makes up for my less-than-perfect hearing, but so be it!

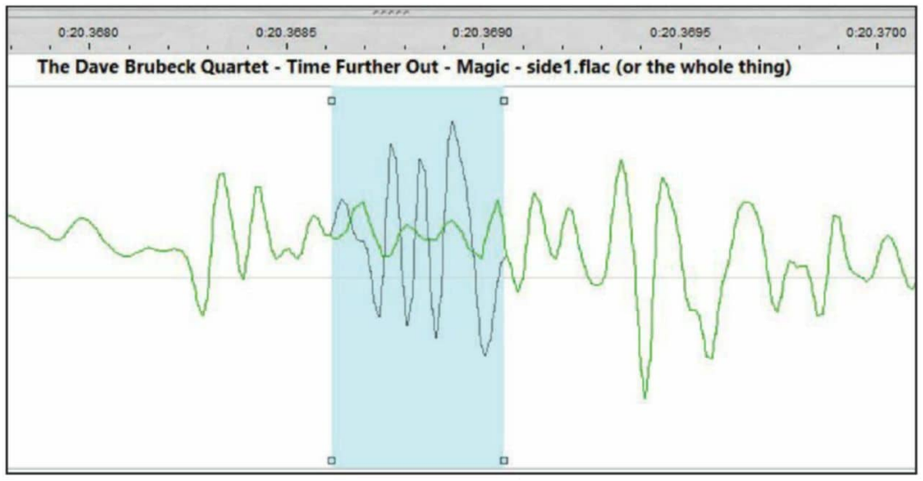


Figure 6: A VinylStudio screenshot of the Puffin's output is displayed with Magic on. The Puffin preserved the original waveform (in black).

Grading the Magic

I tried the "Magic" control to see what it would do. This control can't really do magic. But, according to the manufacturer, it can make a record that is already particularly good into one that is superb in terms of noise-free sound. To hear the dead silence between tracks, even at high volume, is awesome. But I was concerned that some musical information might be removed from the signal if this control is too aggressive in its performance. So, I tried listening to The Dave Brubeck Quartet's album *Time Further Out*. This 1961 recording is typical Brubeck. It has a few tracks with heavy drums. I used the Impex audiophile version of this recording for test purposes since they are considered some of the best vinyl available.

I used the Grade function to determine how good the quality of this album is based on the number of pops and ticks in the recording. It scored an A on side 1, but a B on side 2. I did notice that two of the tracks on side 2 had a lot of heavy percussion. In fact, the first track, "Far More Drums" was one of the heaviest percussion tracks. "Unsquare Dance" was also quite heavy on percussion. The bugaboo of software that tries to find and eliminate clicks is that percussion looks like a lot of clicks to the software. So, I should not be surprised to find out that after playing that first track on side 2, the software was scoring it at a grade of C+. But the Puffin tries to prevent true percussion sounds from being destroyed by the software.

To see if it does that, I used VinylStudio after processing the album through the Puffin's Magic function to see if it found additional noises that the Puffin could not. I set VinylStudio at its lowest click elimination setting and with maximum percussion protection. At that setting, I found several places where VinylStudio called a percussion sound a click, proving that the Puffin was correct in allowing that portion of the waveform through (see **Figure 6**).

Note in Figure 6, that the VinylStudio software would have replaced the black line with the tamer green line inside the light blue colored box. The Puffin let the whole thing through, so I have some confidence that the Puffin is not destroying high-frequency program material sounds. In fact, this proves it does a better job in that regard than VinylStudio. The affected waveform in this example is about 4 ms in length, so it is doubtful if any human ears could hear even the more aggressive click removal from VinylStudio. But the good news is that the Puffin preserves the sound.

Listening to inter-track noise is a good way to see if the software is doing anything. Indeed, the Puffin did reduce the noise substantially compared to my original playback, which had no noise-removal processing (see **Figure 7**).

Listening to the version of Brubeck's "Blue Shadows in the Streets" recorded without Magic, I noted clear sound, a slight 60 Hz hum in the left channel, and a slight crackle in the sound prior to leadout. There was also a very minor tick in the first few measures of the music. I judged the instrument

Figure 7: An intertrack noise spike is seen in this waveform. The Puffin's Magic eliminated such glitches. Note from the scale that this noise spike is down by 35 dB and is, therefore, not very loud.

timbre to be near perfect and I attribute the slight hum to the recording studio.

Listening to the same track with the Puffin's Magic engaged, I detected no change in the timbre or overall sound of the recording. The slight 60 Hz hum was still there. The slight crackle at the end of the track was much subdued, but still slightly audible. The tick I heard in the unaltered recording was still there, but much duller and practically inaudible. I did note a slight hissing sound at the very end. It was there in the original unaltered recording too, but this playback with Magic engaged, showed a slight accentuation of the hissing sound. I cannot attribute this to anything other than possible slight record wear from the elliptical stylus tip. Someday, I will have to invest in Shibata or Microline.

Overall Sound Quality

In a word, it sounds natural. In another word, it sounds transparent. The music comes through without alteration. I first tested the Puffin with flat frequency response to determine if there was anything that stood out. Nothing stood out. It sounds perfectly flat in response. The noise floor was as quiet as I have heard listening to a vinyl recording. With proper gain setting (no clipping), there was no harshness or compression in the sound. It was an accurate sound to my ears and did not seem to accentuate or diminish any of the audio spectrum.

I have tried nearly every control on the Puffin, and they seem to work as intended. The most tantalizing controls are the new Magic and Grade functions. Magic doesn't seem to adversely affect program material as my tests showed, so leaving it engaged should not be a problem. Magic won't remove very egregious defects in the vinyl. Its effect is subtle and frankly, more useful on good quality vinyl than on worn, scratched, dirty, or old recordings.

The Grade function shows how good or bad vinyl is in terms of its surface noise. I was surprised to find some audiophile recordings scoring only a B, when a 45 RPM record I picked up at a Son Little concert showed A and A+ scores. But I would caution that high percussion might show the score lower than it should be. On high dynamic range recordings of orchestral works, the Grade function may also be somewhat critical, because it is hearing the pops and clicks more audibly during quiet passages. But then, so are we, the listeners! Grade gives a good idea of recording quality, but it is not foolproof. It is useful in deciding if you want to invest in a higher quality recording or if you want to determine which of two identical records to discard.

Some of the more interesting and useful controls were Tilt, Air, and the fx controls. I used the Tilt setting to reduce or boost highs. Small changes to the setting were audible if adjusted by two clicks or more. I did not find any of the settings to cause artifacts, which might detract from the music. I only used one cartridge for this test, but I can state that the effects of these control settings would certainly tame a cartridge that is over-emphasizing highs or lows due to impedance mismatch, resonances, or other issues. In today's world of fine cartridges, the controls can help fine-tune the cartridge for best performance.

The Air control didn't brighten the sound of any recordings or diminish the brightness due to its activity at higher frequencies. It does make the sound of cymbals a bit more airy or if turned down, diminishes their air and presence a bit. A little more sparkle can be added or diminished from the sound of tambourines and other percussion instruments. Sibilance can also be the beneficiary of adjustments to Treble, Air, and Tilt.

The fx function settings cause very subtle changes in the sound. The "tube" sound causes accentuation of second harmonics. Such accentuation can cause musical tones to be more pleasing for some listeners, just as harmonic resonance in music tends to make fundamental tones sound more interesting. The basic character of the instrument or voice sounds largely unchanged, however. You might very well find that you like the sound of tubes without experiencing their drawbacks (e.g., high power consumption per watt, excessive heat, and warmup time).

Conclusions

Parks Audio offers a 30-day return policy and a two-year warranty in case it should malfunction during that time due to manufacturing defects. Parks Audio also has a few refurbished Puffins for sale for anyone wishing to try it out. Some may balk at the idea of buying a product online and there are some convenience factors to weigh. Nevertheless, there are several good online companies manufacturing and selling particularly good gear. And with the return policy, you risk nothing by trying the Puffin.

Before offering my own conclusions, I thought the following remarks by Shannon Parks, designer of the Puffin, would be helpful.

"I see the Puffin in two parts. The first part is addressing the problems we vinyl lovers deal with. A major problem is supporting the great variety of different cartridges manufactured today, so a wide gain range is needed for a phono preamp. The Puffin's 76 dB gain range solves this problem. Then low noise performance is a necessity and that's been a key part of my analog work over the last decade. Finally, getting the cartridge and LP to sound 'just right' is a struggle due to factors like cartridge loading, sonic character of the cartridge and overall system, and then even the sound of the mix ending up on the LP. So, the Puffin controls are there for fixing these issues with a quick and easy twist of a knob.

The second part of the Puffin is the evolution of the Puffin itself. Since the 'audio work' is done digitally, Puffins can easily be upgraded with new features and functionality. I want to do some fun and exciting things with future updates, and these are available free to all Puffin customers. This also is a great creative outlet for me as an audio designer. Lots of engineers love to finish a project and move on to the next project—it keeps them stimulated to design new things. Since this evolution is built into the Puffin, I get to spend all my R&D with new ideas within the Puffin. So, the plan is to keep the hardware as unchanged as possible going forward and to spend all my R&D effort on the software instead."

In a world of high-priced high-end audio, the Puffin is a refreshing alternative that provides good sound and an amazing amount of flexibility for its price. It will accommodate any phono cartridge or line source flawlessly with a twist of the gain control, or by selecting a custom setting. It allows great flexibility in sound performance and equalization for nearly anything out there and for any record in existence, even the very old recording formats.

In addition, the manufacturer kept his promise, delivering significant functional upgrades to the product through firmware updates. After several months of use and serious listening, my conclusion is that even though you can go wrong buying a phono preamp, you will not go wrong if you buy a Puffin. If you are allergic to digital and you insist on nothing digital entering the playback chain, this product is not for you. But I highly recommend the Puffin for all but the most dyed-in-the-wool purist. Enjoy!

About the Author

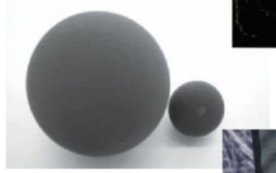
Mike Harkins loves audio and has been active in the hobby for more than 40 years. Since leaving full-time work in the tech industry, he was able to rekindle his interest in all things audio. He attended audio exhibitions such as AXPONA, Florida Audio Fest, and Rocky Mountain Audiofest when it was possible to attend such events. Last year, a letter to a well-respected audio publisher led to Mike's first articles about two of these events being published last year and early this year. He also writes an audio blog for a local audio/video store. He loves to write, almost as much as learning about and listening to high-end audio. Mike has an electrical engineering degree and has most recently taught continuing education courses at the local community college. He is a senior member of The American Society for Quality (ASQ) and in addition to audio, likes to dabble in amateur astronomy. He lives in Austin, Texas with his wife, Theresa.

Celebrating 40 Years!

ACOustics begins with **ACO**™

Measurement Microphones
Polarized and Electret

Community/Industrial Noise Monitors/Alarms Indoors and Out



Windscreens
3 to18 inch & Custom
Weather/UV Resistant

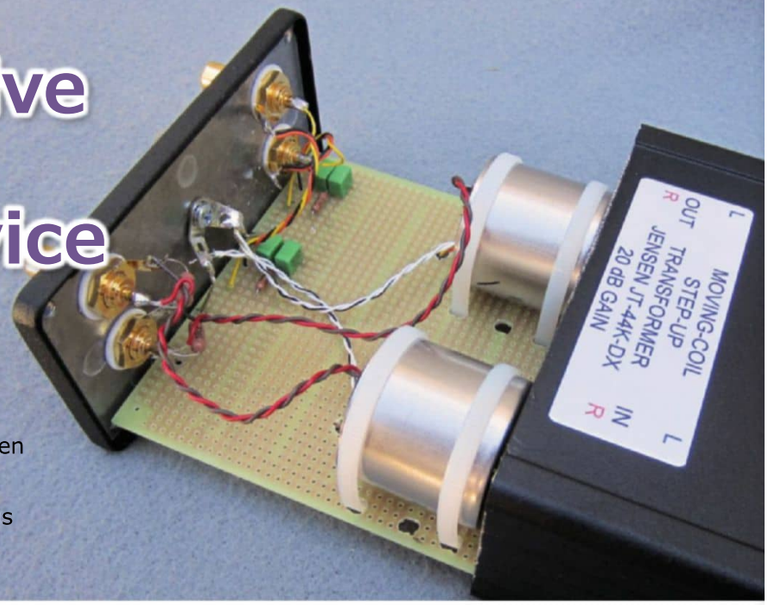
Microphone Systems
Standalone
IEPE Powered
Phantom Power
Custom

ACO Pacific, Inc
www.acopacific.com sales@acopacific.com
Tel: +1-650-595-8588

DO IT YOUR WAY
Cables/Panels/Boxes
WITH REDCO AUDIO'S
"Build Your Own" web pages
Full control for design, editing and more.
AND WE'LL BUILD IT JUST THE WAY YOU WANT!

REDCO AUDIO
www.redco.com 1-800-572-7280

Build a Passive Moving-Coil Step-Up Device



By
Gary Galo
(United States)

After reviewing other products using Jensen transformers, Gary Galo decided to build his own step-up device and now shares his experience in this project article.

Vinyl records are a lot like Mark Twain—rumors of their demise were greatly exaggerated. Vinyl playback is alive and well, with new audiophile LPs being released weekly, and a used vinyl market still thriving in specialty shops such as Academy Used Records in New York City and Princeton (NJ) Record Exchange, along with Internet dealers on eBay and elsewhere. For the vinyl collector, there's an almost dizzying array of playback equipment in all price ranges, covering turntables, phono cartridges, separate tone arms, and phono preamps.

Although there are many high-quality moving-magnet, moving-iron, and high-output moving-coil cartridges receiving favorable coverage in the high-end press, many audiophiles still prefer low-output moving-coils. These cartridges normally have very low output impedance and voltage, and typically require 20 dB or more additional gain compared to high-output cartridges. Output impedances are typically below 50 Ω, and output voltages are usually 0.5 mV or less.

Many phono preamps offer switchable gain for low- and high-output cartridges and, properly designed, moving-coil phono preamps with all-active circuitry can perform extremely well. Simply increasing the gain in a moving magnet preamp is inevitably a compromise; this solution is usually found only in very cost-effective products. The best all-active preamps for low-output cartridges have an input-to-output gain structure that's optimized for this purpose.

Invariably, the increase in gain comes with a corresponding increase in noise. When low-output moving-coil cartridges first became popular, back in the 1970s, there were a number of active, outboard "head amplifiers" that provided the additional gain. These head amplifiers—sometimes called "pre-preamplifiers"—are normally connected to the input of a standard RIAA moving-magnet phono preamp.

Today, outboard head amplifiers have all but disappeared. One of the few units still made is the DB Systems DB-4B, first introduced in 1976, and still selling for a very reasonable $270 US. Another is the British-made Graham Slee Elevator EXP Moving Coil Head Amplifier sold by LP Gear for $1115 US. Several active head-amplifiers were designed by W. Marshall Leach, Jr. decades ago—schematics for three of them are still available on his website. These days, nearly every all-active preamplifier system for moving-coil cartridges is designed as an integral RIAA phono preamp.

Step-up transformers have long been an alternative to the all-active gain solution for low-output moving-coil cartridges. There are still many models from which to choose, ranging in price from a few hundred dollars US, to several thousand. A step-up transformer is a passive device. Unlike an active preamp built with tubes, discrete transistors, or IC op-amps, a transformer won't add any electronic noise (i.e., background hiss) of its own. The greatest potential for noise with a step-up transformer is the introduction of hum. But, given proper shielding, layout, and grounding, hum levels can be kept to levels nearly as low as they are in a playback system designed for high-output cartridges.

In the April 2017 issue of *audioXpress* I reviewed several Hafler phono products, including the PH44 step-up transformer. At that time, the PH44 retailed for $999 US, but most Internet dealers are now selling it for $599. I was quite enthusiastic about the performance of this product, which is built around a pair of Jensen JT-44K-DX transformers with 20 dB of gain. These transformers were designed by the late Deane Jensen, specifically for MC cartridges with impedances around 45 Ω (e.g., the Denon DL-103 series). I evaluated the review sample with the DL-103R, an improved version with high-purity (6N) copper in the coils, and an output impedance of 14 Ω. With proper loading, the Hafler step-up proved to be an excellent match for the improved model.

Possible Improvements

Although it performed very well, my audio DIY side told me that I could roll my own for a lot less money. After looking at the layout of the Hafler step-up, I also realized that there were areas that might be improved. Jensen notes that the primary and secondary transformer should be separated as much as possible, to avoid capacitive coupling between them. The input and output leads should not be twisted together under any circumstances. They also advise keeping the secondary leads as short as possible (this is not so critical with the primary leads because of the low impedances involved). The layout in the review sample reflected some minor changes in cable routing and layout compared to the photo still seen on the Hafler website. In the current version, the transformers have been moved closer to the rear panel, but the output leads could be made even shorter with a new layout. The input and output connectors are closer than they need to be, so I decided to separate them as much as possible in my project.

Jensen still sells its own step-up device using these transformers, the model MC-2RR-L, which sells for $600. Jensen achieves maximum spacing between the primary and secondary leads by placing the input and output connectors on opposite ends of its enclosure. The disadvantage I see in this approach is that interconnect cables between the turntable and the step-up, and/or between the step-up and the phono preamp input, may be longer than I'd like—especially if the step-up has to be oriented sideways relative to your phono preamp. I prefer to keep the interconnect cables as short as possible. I'm baffled that so many high-end tonearm cables are sold at 1.5 meters in length. This makes no sense at all, especially with low-output cartridges. My own cables, which I make myself, are no more than 2' (approximately 61 cm) long.

Custom Layout

I decided to buy a pair of the JT-44K-DX transformers directly from Jensen and build my own step-up device. My goal was to use a layout that would place the input and output connectors on the rear panel, maintaining the necessary spacing between the leads, and keeping the output leads as short as possible. Hafler uses a steel enclosure, but I prefer to avoid permeable metal. Jensen's is aluminum, and Dave Hill, Jensen's General Manager, noted that unless the case is set directly on top of another piece of equipment with a strong magnetic field, induced hum should not be a problem. All of Jensen's moving-coil step-up transformers have double mu-metal shielding, which protects the transformer coils from stray electromagnetic interference (EMI). Just to be

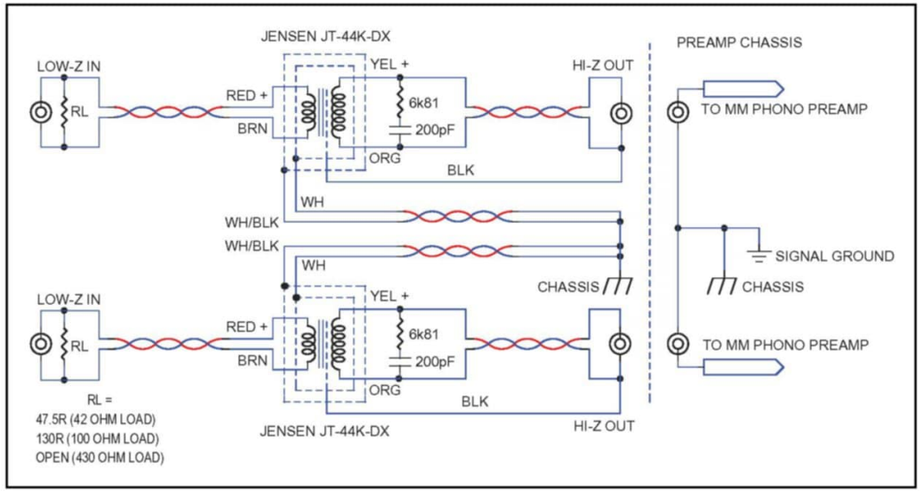


Figure 1: This is the schematic diagram for the moving-coil step-up device, built around a pair of Jensen JT-44K-DX transformers. Transformer shield leads connect with the electrostatic shield and output signal grounds at the preamp input. Interconnect and ground leads between the step-up and phono preamp input should be kept as short as possible. Values for RL are shown in the lower left.

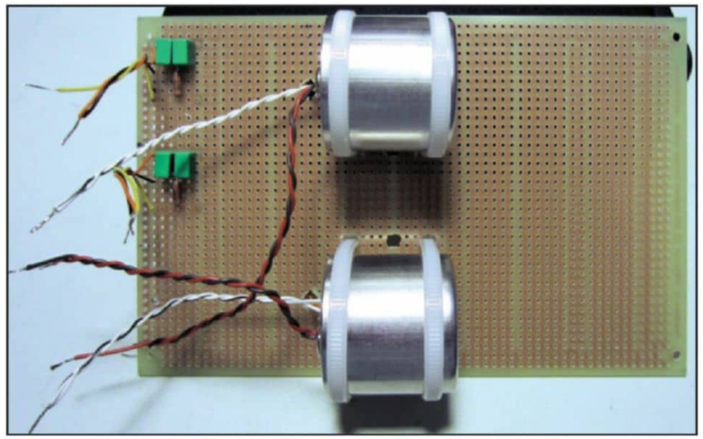


Photo 1: This is the top of the board with the transformers and Zobel components mounted. The yellow, orange, and black transformer leads are routed to the bottom of the board through holes drilled directly in front of the transformers.

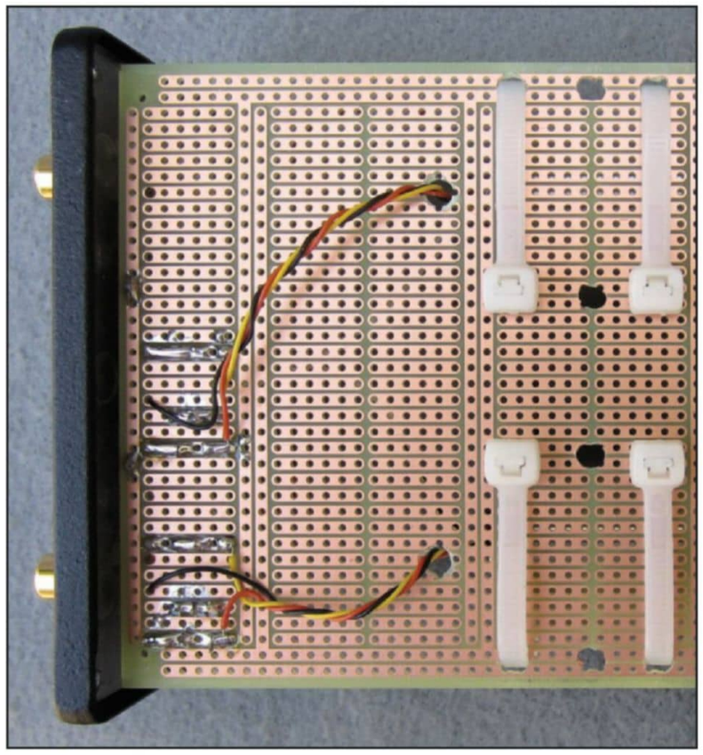


Photo 2: This is the the bottom of the board, with the rear panel secured, showing the yellow, orange, and black twisted bundles. The yellow and orange wires are cut to length and soldered across the Zobel components. The black wires are routed to the top of the board through unused holes in the board.

safe, I chose an aluminum enclosure with its own EMI/RFI shielding, the Hammond 1457N1601EBK. This extruded aluminum enclosure comes with conductive foam gaskets for the end caps, which make the enclosure water-resistant. For this project, they're not necessary and can be discarded. The enclosure measures 4.09" wide by 6.3" deep by 2.15" high (104 mm × 160 mm × 55 mm), dimensions that are very close to the Hafler enclosure.

Figure 1 shows the schematic for the complete stereo transformer circuit. RL is the load resistor needed to bring the input impedance down to that required by the cartridge. For the DL-103R, Denon recommends a 40 Ω cartridge load with a step-up transformer and 100 Ω with an active preamp. The difference is likely because there's a reactive component to the load with a transformer—an active preamp will have something close to a purely resistive load. With this circuit RL should be 47 Ω or 47.5 Ω. This yields a net input impedance of 42 Ω when the output is loaded with the standard 47 kΩ resistor found in nearly all moving-magnet phono preamps.

I discussed proper loading for the Denon cartridges in my Hafler review, and noted some considerably higher recommendations that I found to be bizarre. I use wide-bandwidth, very transparent solid-state electronics and I found that any loading substantially higher than that recommended by Denon resulted in a tonal balance that was too bright. Personally, I don't believe in balancing euphonic colorations. LPs and LP playback systems are admittedly colored, however wonderful they may sound. But, I want the rest of my electronics to tell me exactly what my LP playback system is doing. If you use electronics that soften the treble region, you're welcome to adjust RL to your personal taste.

Assembly

Mount the transformers and parts for the output Zobel network on a BPS PowerBoard-3U, as shown in **Photo 1** and **Photo 2**. This board is exactly the right width for the internal slots in the Hammond enclosure. One end of the PowerBoard has a single bus running across the entire width of the board. The transformers are mounted closest to this end (the other end is for attaching a connector and is not used for this project). You'll need to drill holes in the board large enough to accommodate the cable ties, as well as smaller holes in front of the transformer locations for the routing of the output leads. After drilling, secure the transformers with 8" nylon cable ties. The transformers should be positioned with the red and brown leads on top.

The Zobel components are supplied with the transformer. Since 200 pF is not a standard value, each of mine came with two 100 pF caps, which are connected in parallel. Photo 1 shows the locations of the Zobel components for each channel—solder them to the traces oriented toward the length of the board—don't connect anything to the bus running across the edge of the board just yet.

For each transformer, make three twisted bundles with the transformer leads, as follows:

- The red and brown primary leads
- The white and white-black mu-metal shield connections
- The yellow and orange secondary leads along with the black electrostatic shield lead

For now, twist each bundle for no more than 1" to 1.5"; adjust the lengths of the twisted bundles when the other ends are connected. Route the yellow/orange/black wires through the holes already drilled in the board. Connect the yellow and orange wires across the Zobel components, on the bottom of the PC board, trimming the leads so they are no longer than necessary. Save the excess wire and do not cut the black wires yet. Slip the black wires through unused holes in the board and pull them through on the top of the board.

Photo 3 and **Photo 4** show two views of the top of the board with the rear panel in place. The RCA jacks must be insulated from the chassis—the DH Labs and Parts Express connectors mentioned in the Parts List come with insulating washers. Mount the RCA jacks, the ground post, and the two ground lugs, and secure the rear panel to the board with two pieces of solid, 18 AWG bus wire from the ground lugs to the trace across the back of the board. After soldering the bus wires to the ground lugs, insert them through the holes in the PC board. I suggest doing a dry run, sliding the board into the slots on the bottom of the case. After you're sure you have the board in the right position to attach the rear panel to the case, solder the two bus wires to the board. Note that the connection of the bus wires to the trace across the back of the PC

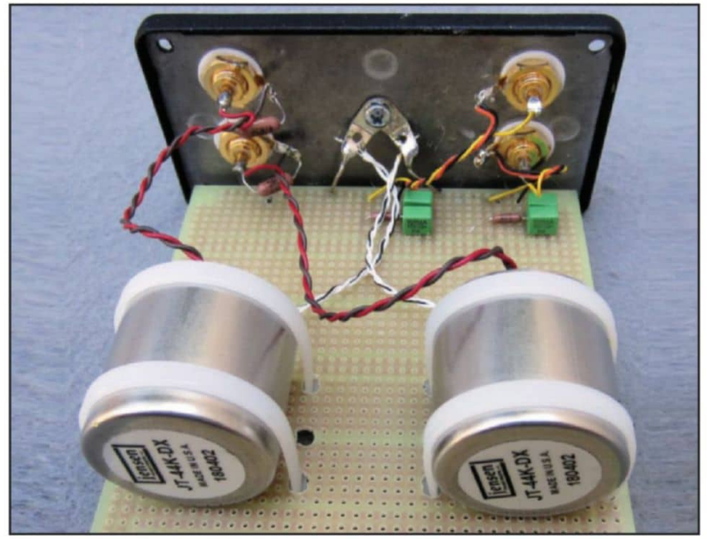


Photo 3: This is an inside view of the completed board and rear panel. The transformer input and output leads are spaced apart and oriented at angles, minimizing capacitive coupling. The two pieces of bus wire from the ground lugs to the trace across the front of the board secure the board in place, eliminating strain on the fragile transformer leads.

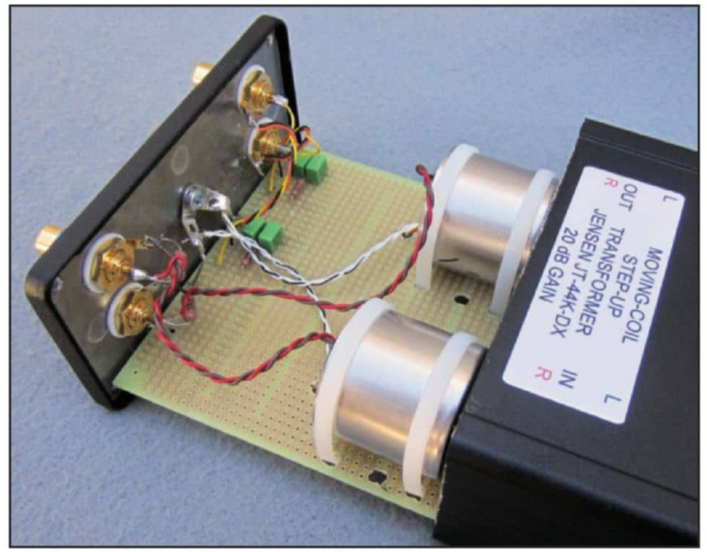


Photo 4: This view shows the completed project with the board partially inserted into the case. The board slides into the bottom channels on the sides of the case.

Parts List

(2) Jensen JT-44K-DX Moving-Coil Step-Up Transformers ($104.76 each from Jensen Transformers; includes mounting brackets and R/C Zobel parts)

(1) Hammond 1457N1601EBK Extruded Aluminum Enclosure, EMI/RFI Shielded (approximately $45.00 from Digi-Key or Mouser Electronics)

(1) BPS PowerBoard 3U Prototyping Board (about $11.00 from Mouser Electronics, 854-PR3UC, or Amazon.com)

(4) DH Labs CM-R1 RCA Jacks ($25.00 each from DH Labs or Parts Connexion)
-or-
(4) Parts Express 091-1120 RCA Jacks ($6.99/pair from Parts Express)

Miscellaneous: ground lugs, 6-32 machine screw, nuts, lock washer, knurled thumb nut, 8" nylon cable ties

board serves no electrical purpose. It's simply there to secure the rear panel and eliminate any strain on the transformer wires.

Use the remaining yellow and orange transformer leads to make the connections from the Zobel networks to the output RCA jacks. For each channel, the black, yellow and orange wires should be twisted together. Make these bundles only as long as they need to be and route as shown in the photos. Twist the red and brown input leads for each transformer and solder them to the input RCA jacks. Twist the white and white/black transformer shield wires for each channel together, and solder them to the ground lugs on the rear panel. When you're finished, the spacing and positions of the transformer leads should resemble the photos. The input and output transformer lead bundles are spaced far enough apart, and do not run parallel to each other—they are oriented at angles. When the wiring is finished, carefully slide the board into the enclosure and secure the rear panel.

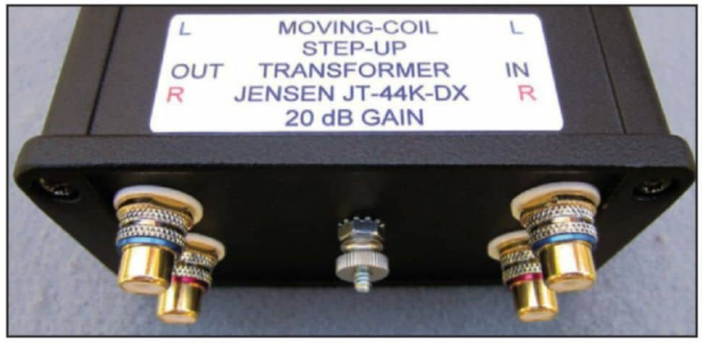


Photo 5: This is the rear panel of the moving-coil step-up device. Input and output connectors and the ground post are all located on the rear panel, to facilitate the shortest possible interconnects and ground leads.

Photo 6: This is the completed moving-coil step-up project. The compact case can be set directly on top of, or next to, the phono preamp, keeping interconnect and ground leads short.

Photo 5 and **Photo 6** show the assembled enclosure. Connect the ground post on the step-up box and the turntable ground wire directly to the ground post on your phono preamp. Don't daisy chain the ground wires from the turntable to the step-up box to the preamp. The ground wire from the step-up box should be as short as possible—mine is about 8". The interconnect from the step-up box to the preamp should also be short and low capacitance—my custom-made cables are 14", measured from the tips of the RCA center pins. With this arrangement, the transformer secondary and the electrostatic shield grounds connect with the transformer mu-metal shields and set-up device case ground at the preamp input, making the preamp input the star ground reference for the entire system.

Subjectively, with my preamp gain at maximum and my ear up to the speaker, hum and hiss are comparable to a moving magnet cartridge connected directly to my phono preamp—in other words, just barely audible. If you find that your phono preamp's grounding arrangement is not compatible with my grounding scheme, I suggest connecting the RCA connector shields in the step-up box to the ground lugs on the rear panel. Most phono preamps should work just fine with my arrangement. Jensen recommends removing any cartridge loading capacitor from the input of your moving magnet preamp.

Performance

The DIY moving-coil step-up device offers excellent performance for substantially less money than comparable factory-assembled products. The total cost of the parts for this project is less than $400. I chose the DH Labs premium RCA Jacks because I believe they make a difference in the sound on a high-resolution system. But, you could probably knock $85 off the price by using the affordable Part Express jacks. I strongly recommend gold-plating, in any case.

In my Hafler review, I showed the square wave response I measured for this transformer. Jensen's datasheet has several graphs and a complete list of specifications for the JT-44K-DX. The bandwidth at the -3 dB points is 2 Hz to 100 kHz, and deviation from linear phase is only 4° and 20 Hz with a 25 Ω source impedance. Jensen has another transformer in this series with 31 dB of gain—the JT-34K-DX. This transformer is suitable for very low impedance cartridges, including many of the Ortofon models.

For even better performance, Jensen offers a premium line of MC step-up transformers—the JT-346 and the JT-347 series. Five models are available to accommodate a variety of cartridge impedances and gain requirements. These premium transformers offer wider bandwidth in the bass (-3 dB at 0.3 Hz), and lower deviation from linear phase (around 1° to 1.5° at 20 Hz) than the JT-44 and JT-34 transformers. Distortion is also lower but, as expected, the improved performance comes with a price. The JT-346 models sell for $228 US each, and the JT-347 transformers cost $250 US each. I hope to try these premium transformers at some point. In the meantime, the JT-44K-DX transformers offer excellent performance with my Denon DL-103R cartridge and I expect them to meet my LP needs for some time to come.

Author's Note: I would like to thank Dave Hill, General Manager of Jensen Transformers, for his support and technical expertise during the preparation of this project.

About the Author

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

Resources

G. Galo, "The Hafler Revival: A Review of Hafler's PH50 and PH60 Phono Preamps and the PH44 Moving-Coil Step-Up Transformer," *audioXpress*, April 2017, https://audioxpress.com/article/a-review-of-hafler-s-ph50-and-ph60-phono-preamps-and-the-ph44-moving-coil-step-up-transformer.

W. Marshall Leach, Jr., "Head Amps for Moving-Coil Cartridges," https://leachlegacy.ece.gatech.edu/headamp

Sources

DB Systems DB-4A Head Amplifier
DB Systems | www.dbsystemsaudio.com

DH Labs | www.silversonic.com

Digi-Key Electronics | www.digikey.com

Graham Slee Elevator EXP Moving Coil Head Amplifier
www.lpgear.com/product/GSPELEV.html | www.hifisystemcomponents.com

Hafler PH44 Step-Up Transformer
Hafler | www.hafler.com/ph34ph44.php

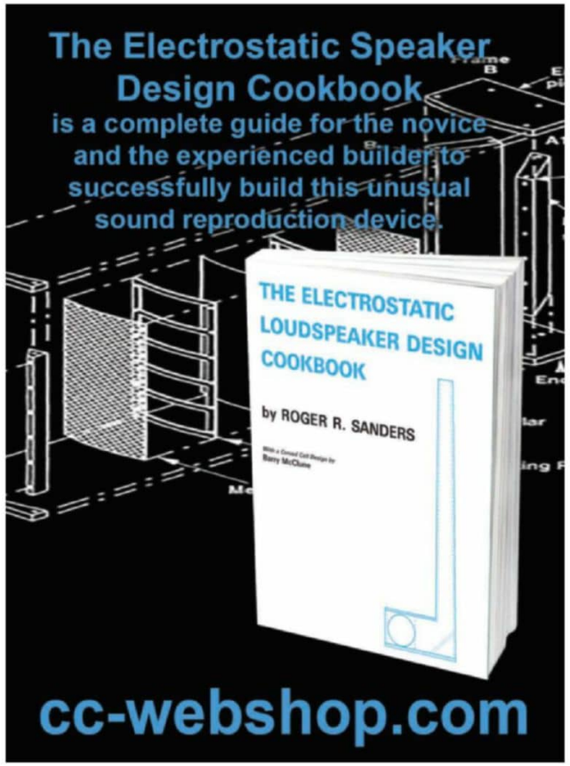
Hammond Manufacturing | www.hammfg.com

Jensen MC-2RR-L
Jensen Transformers | www.jensen-transformers.com/product/mc-2rr-l

Mouser Electronics | www.mouser.com

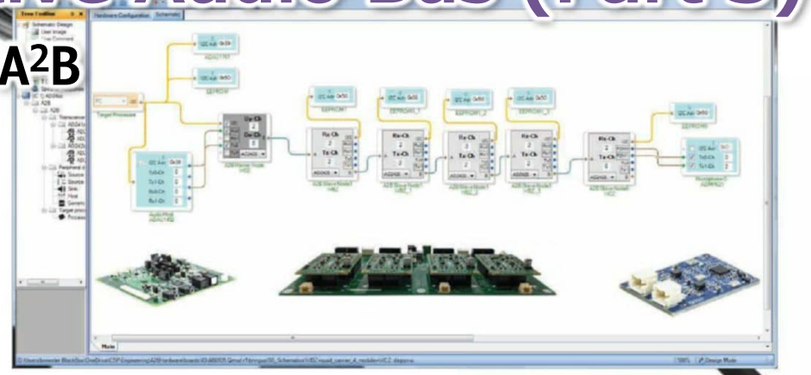
Parts Connexion | www.partsconnexion.com

Parts Express | www.parts-express.com



Getting Started with Automotive Audio Bus (Part 3)

Designing in A²B



By
Brewster LaMacchia
(United States)

The first two parts of this article examined the overall A²B system operation and how its features can enable new types of digital audio applications. This part takes a deeper look at A²B data transport and Analog Device Inc.'s (ADI) AD242x series of parts.

The normal A²B data rate is 49.152 Mbps. As noted in th first part of this article series, we are rounding the data rate up to 50 Mbps. This data rate corresponds to a 49.152 MHz fundamental frequency, which is also rounded off to 50 MHz.

ADI offers a number of IC variants that are cost optimized for particular use cases. The AD2428, which supports the entire feature set, will be the part chosen for most designs. We are using the AD2428 part number to stand in for any of the variants, with the understanding that features and circuits may need modification for the other versions. While the features are discussed in the context of the automotive use cases, we are also considering use cases anywhere consumer, commercial, or pro audio devices may be found.

A²B Bits at the Wire Level

The basic interface for the ADI transceivers is Low Voltage Differential Signaling (LVDS), with some differences in the details. Released in the mid-1990s, LVDS is used in many applications to move digital data over high speed serial links [1]. As with balanced audio, two lines 180° out of phase with each other are used to send data. This increases noise immunity, and in the case of high-speed digital signals, minimizes radiated electromagnetic interference (EMI) when used with twisted pair wire compared to single-ended configurations.

Another consideration for digital signals, which at the frequencies involved are better thought of as radio frequency (RF) vs. 1s and 0s, is impedance matching. At audio frequencies impedance is generally only important in terms of maximizing voltage transfer (audio interconnects). At radio frequencies, impedance mismatches cause signal reflections, resulting in distorted waveforms and increased EMI. Users of AES/EBU or SPDIF interfaces, which are only a few megahertz, have no doubt seen required impedance statements in datasheets. At the (nominal) 50 MHz A²B frequencies, particularly with longer wire lengths, impedance matching becomes part of the must-haves to correctly move data around.

LVDS only describes an electrical interface, not what it is used for. In the case of A²B the data is Manchester encoded, which allows the receiving end to easily recover the clock needed to know where the bits are. Manchester encoding encodes data on clock edges, rising for a 0 and falling for 1. Therefore instead of trying to determine the signal level at a specific instant in time, the receiver only needs to know the before and the after state—an easier proposition for which to build hardware. Manchester encoding is DC balanced, meaning systems can be AC coupled, which in turn simplifies phantom power design. It's also easy to extract a clock from.

Figure 1 shows an actual waveform captured from hardware. Though not captured on a very wide bandwidth scope (200 MHz in this case) with specialized differential probes (you know, the ones that cost more than your last car), we want to illustrate what actual waveforms look like.

Also of interest in understanding A²B is the spectrum of the signal, as shown in **Figure 2**. The peak is at the expected (approximately) 50 MHz and third harmonic at 150 MHz. The spike close to 100 MHz is actually a nearby FM station and illustrates the importance of knowing your test tools and system, otherwise one might assume in this case there's some sort of bad 2× clock signal ending up on the wire.

The peak is broad as the data causes differing patterns—the theoretical spectrum can be calculated from the signal properties but that is beyond what we need to know to use A²B. Below a few megahertz, there's not much signal energy, which will be helpful in understanding the filter components used in the ADI recommended circuit.

We can also start to understand the cable length limitations are not just a question of drive strength. The master node needs to send downstream all of the data (1024-bit periods are allocated per 48 kHz frame, one bit period is therefore 20.3 nsec) and receive all of the upstream data back within 20.83 µsec. Cat 5 propagation delay is about 5 nsec/m. The 40 m length limit corresponds to a delay of 200 nsec in each direction, or 400 nsec total—about 20-bit periods.

Each node also has to receive the data and reclock it out again and the nominal time for that is 9-bit periods. With 10 nodes that totals 180-bit periods. Between cable delay and node delay about 200-bit periods are needed to deal with delays related to the physics of the problem, leaving about 800-bit periods for headers, control information, and audio data. Those bit periods are allocated between the up and down stream directions.

A²B Analog Sections

With high-speed digital signals as well as analog RF signals, the source and load impedance must be matched [2]. Many differential digital systems use a 100 Ω impedance with twisted pair wire (e.g., a CAT5) can easily meet that requirement in a low-cost manner. While a single 100 Ω resistor at the IC input meets the matched termination requirement, most designs will use two 50 Ω resistors with the center connection AC grounded via a large value capacitor. This improves common mode rejection (CMR) and has no DC current draw—the details of the advantage of this scheme can be found in the prior reference.

Standard LVDS is normally unidirectional and A²B operates bidirectionally, which means termination resistors need to be on each end of the cable to accommodate data being sent in both directions.

Figure 3 shows the circuitry recommended by ADI. This drawing does not include the parts to extract phantom power from the bus, as well as other parts needed for local powered slaves. Full details are available from ADI [3]. Manchester encoding ensures no DC, therefore the system can be capacitively coupled. The two 10 nF capacitors serve that purpose; at 50 MHz XC < 0.3 Ω, and we can treat them as a piece of wire as far as the data signal is concerned. In this discussion, we'll ignore the DC offset needed by the AD2428 A²B transceiver.

The first thing the astute reader probably noticed is two 120 Ω termination resistors and said "wait, you just told me the termination impedance is supposed to be 100 Ω." And indeed it (sort of) is. The other components in Figure 3 also change the impedance in the frequency range of interest.

The critical consideration is the AD2428 isn't a standard LVDS part. It includes two 100 Ω resistors internally, and is center AC grounded through a capacitor external to the chip. For each side, this

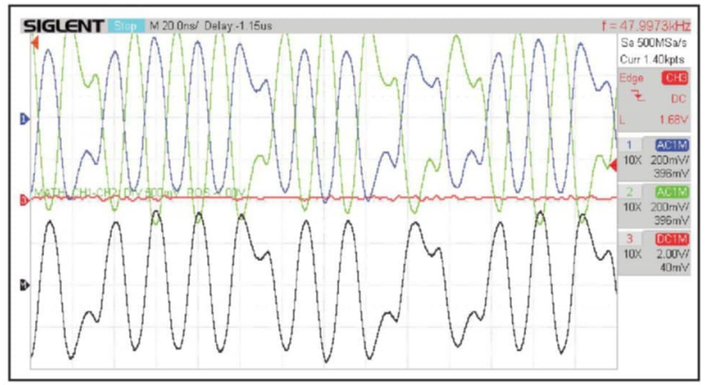


Figure 1: A²B data (black trace) is computed from the captured P (blue) and N (green) signal lines at the connector (approximately 200 MHz BW).

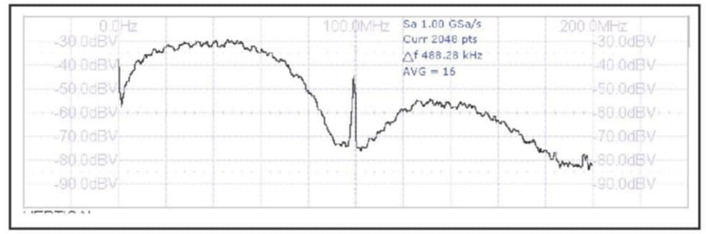


Figure 2: Measured A²B spectrum (random data). The spike at 99.5 MHz is a nearby FM transmitter. Linear frequency scale.

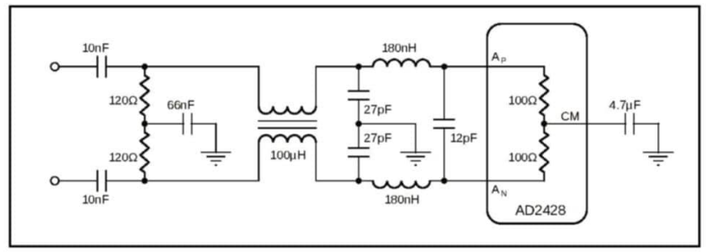


Figure 3: Recommended ADI design for A port for bus powered slave (excluding power circuitry). The 27 pF/180 nH combination is to reduce emissions in the FM/DAB band. This circuit, when combined with the AD2428s, offers 12 kV contact and 15 kV air discharge protection.

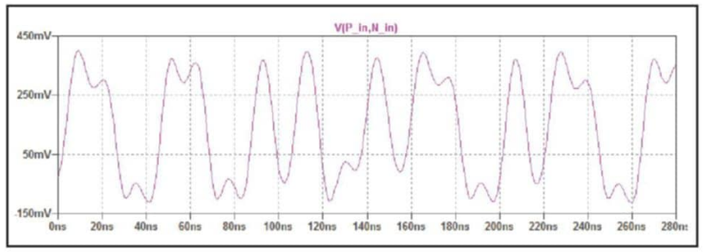


Figure 4: The LTSpice simulation of A²B system is measured at the input connector of an A port. Compare the general waveshapes to Figure 1.

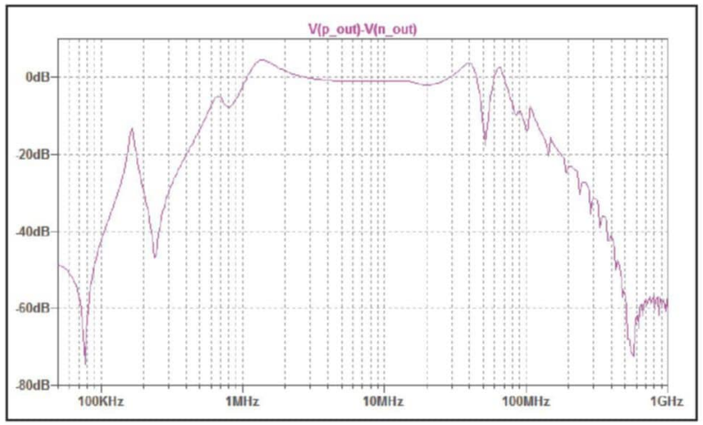


Figure 5: Filter output vs. frequency for transceiver output as seen at the connector with normal A²B load. This is not the same as the output spectrum.

References

[1] "Low-voltage differential signaling," Wikipedia, https://en.wikipedia.org/wiki/Low-voltage_differential_signaling

[2] H. Johnson and M. Graham, *High-Speed Signal Propagation, Advanced Black Magic*, Prentice Hall, 2003.

[3] AD241x, AD242x, Reference Schematics, Rev 2.1, Analog Devices, August 2019.

[4] Tech Note and LTSpice files, Clockworks Signal Processing, https://clk.works/2020/04/ltspice-files-for-a2b-simulation

[5] B. LaMacchia, "Getting Started with Automotive Audio Bus (Part 2): Understanding A²B Features," *audioXpress*, November 2020.

[6] J. Dunn, "Jitter Theory," Audio Precision Tech Note TN-23, 2000.

[7] "Tech Note008 A2B I2S Clock Jitter," Clockworks Signal Processing, https://clk.works/information/technotes

Sources

ADI's main A²B web page provides links to part information, ADI tools, and design information, www.analog.com/en/applications/technology/a2b-audio-bus.html

means there is 100 Ω || 120 Ω. There's also a parallel impedance from the 12 pF and 27 pF capacitors, and when you put this all together it comes out to around 50 Ω at 50 MHz—since this is a differential circuit, around 100 Ω differential impedance.

The port is bidirectional. Since the differential source impedance when the AD2428 is transmitting is also 200 Ω, and the filter network's characteristics are the same, this creates the same 100 Ω source impedance.

For those interested in understanding more about this circuit's behavior in the time and frequency domain, as well as LTSpice files that can be downloaded for your own experimentation, please see the materials described in reference [4]. Due to size constraints the LTSpice model is not included in this article.

While simulations can fall prey to a garbage in, garbage out (GIGO) model, the reasonable frequency range and the use of (mostly) ideal components with first-order parasitics gives useable results. **Figure 4** shows an A²B signal example from the simulation of the same design with which Figure 1 was measured. **Figure 5** shows the frequency response for the A²B signal path from the connector to the transceiver pins. The simulation uses two copies of Figure 3 (one for Tx, the other for Rx) and a simple model for the interconnecting CAT5 cable.

ADI's A²B design guide offers layout advice for the A²B signals, which includes guidance that it should be treated as a controlled impedance differential design. While the design of the analog filter of Figure 3 limits the high-order harmonics it's prudent to assume bandwidth extends out to at least the third harmonic of the 50 MHz data, which with the Manchester coding spread would be up to 200 MHz.

In FR-4 PCB material, 200 MHz has a wavelength of 700 mm, using the wavelength/10 rule of thumb for critical trace length, we find that traces less than 70 mm shouldn't be of concern. This is a long length relative to most A²B layouts. If we consider EMI, we might want to look out to a 500 MHz bandwidth and that puts the critical length at 27 mm, which does fall in the range of many A²B designs. Though the distributed termination (i.e., the 120 Ω resistors in Figure 3 and the 100 Ω internal to the AD2428) raises the question of exactly what impedance the layout should target, ADI recommends 100 Ω and without extensive alternate analysis 100 Ω would be good advice.

Regardless of the length rules, as a differential signal, both the positive and negative differential signals should be matched in all aspects of components, layout, and routing to prevent common mode to differential mode noise conversion and to minimize EMI.

A²B Transceiver Device

Having worked through an agonizing amount of background material, we finally reach the point of looking at the actual chip that ADI sells to build A²B systems. Full details are in the datasheet so only a quick summary is offered here. **Figure 6** summarizes the internal chip functions and outlines some of the connections. The resources available from the ADI website should be consulted for actual design details.

The chip operates across a wide voltage range, the low-end datasheet limit is 3.7 V but to be able to supply phantom power, 5 V is a more practical lower limit. We've ignored the Schottky diode drop and for phantom powered slave nodes, VSS will not be 0 V [5]. The chip includes two internal regulators, 3.3 V for the LVDS section and 1.9 V for the chip's internal logic. The I/O section (I²S, I²C, GPIO) can be powered by 1.9 V or 3.3 V to match the voltage needs of peripherals. The regulators supply enough current to power the chip and provide about 25 mA for a peripheral chip but a more rigorous computation is required for any actual system design.

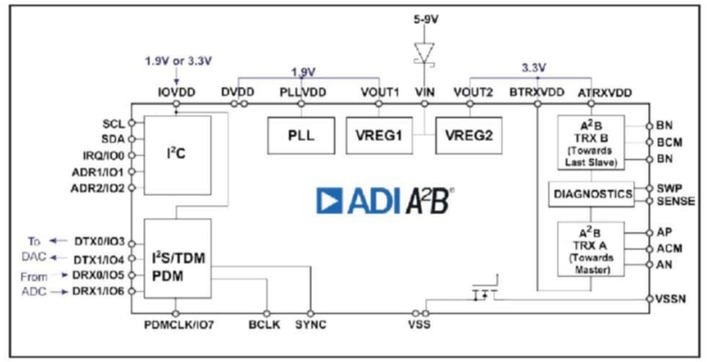
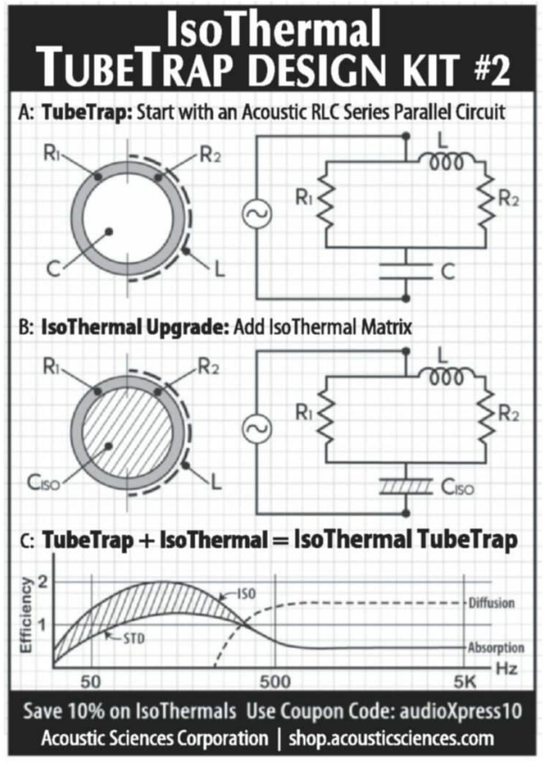


Figure 6: This is the AD2428's block diagram and connections. (Source: Analog Devices Inc., used with permission).

The signals for the A²B bus (shown on the right side of Figure 6) were detailed in the second part of this article (November 2020). The CM pins each connect to a 4.7 µF to ground (along with smaller values to reduce ESR at high frequencies) and are part of the AC termination scheme inside the chip that uses two 100 Ω resistors, the other half to get to the required 50 Ω is provided by the network shown in Figure 3.

The I²C port can be an I²C slave when the chip is operated as an A²B master node or an I²C master



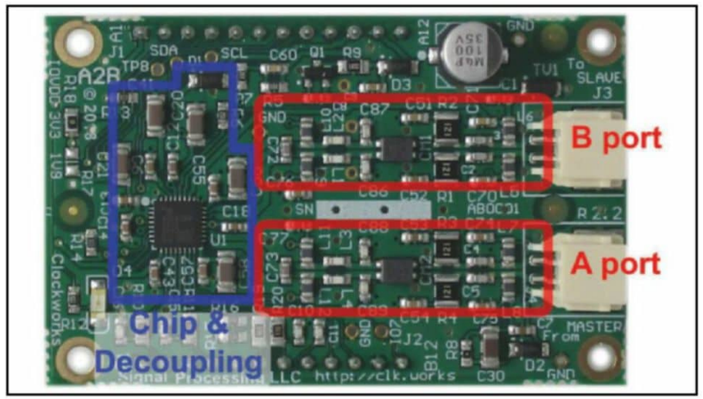


Figure 7: The layout of an A²B module (63 mm × 40 mm) is constructed as a single component side PCB and no 0402 components. Note the layout symmetry required by the high-speed LVDS signals.

when the chip is operated as an A²B slave node. In a slave node, it can also operate in I²C multimaster mode—simplifying the hardware design if the node requires a processor that needs to send/receive messages from the A²B host.

Like most chips, signal functions are multiplexed to try and support a large number of use cases without additional I/O expanders. GPIO lines can serve as interrupts back to the A²B host, as well as support a virtualized GPIO connection between the A²B master node and one or more slave nodes.

There are two sets of I²S data lines—one pair for input and the other for output. All support up to TDM32 mode with 16-bit or 32-bit word sizes. The input pair can also be used for PDM input. The AD2428 has dedicated bit clock and frame sync pins. For ADCs or DACs that need a master clock, the IO7 pin can be used. It also supplies PDM clock. ADI recommends 33 Ω series termination resistors on the I²S output pins, along with 27 pF to act as a low-pass filter to reduce EMI. This advice must be considered in the context of entire system design if the I²S or PDM connections are not point to point or traverse large impedance discontinuities (e.g., ribbon cables).

There is one specification on the AD2428 datasheet that will be of concern to anyone planning to use A²B for higher performance analog input and output: the unusually high I²S period jitter [6]. It ranges from typical values of 1.57 nsec RMS (one node hop) to 2.70 nsec RMS typical/5.5 nsec RMS maximum at the end of a 10-node chain. ADCs and DACs with good internal phase-locked loops (PLLs) will be able to handle some level of jitter, but determining the actual impact [7] is difficult as the jitter spectrum from the AD2428 is not defined by the datasheet. It's also reasonable to assume the jitter has data and use case dependencies since the A²B bus is idle for different amounts of time based on the data slots used. A slave node's PLL can only sync to the master when downstream data is going by.

To avoid extra design spins, high-performance audio applications should be prototyped and fully characterized to determine if the I²S jitter will need remediation. There are readily available ICs that can provide the needed clock functionality.

The AD2428 includes two "add jitter" features—one for the A²B bus and the other for the I²S interface—as a way to cheat on EMC testing by spreading the energy from periodic signals across a wider bandwidth. Use of this feature is not recommended in high-quality audio applications as: 1) the last thing any quality digital audio systems needs is more jitter; and 2) the total radiated energy is the same so it contributes to parasitic EMI pickup up in low-level analog circuits, regardless of how much jitter (bandwidth) spreading is used. Though the audible effect may be somewhat changed, it's always best to not have the problem in the first place.

And In Real Life?

We'll conclude the third part of this article with a look at an actual PCB to see how the various bits of theory, design rules, and circuits all come out in an actual product. **Figure 7** shows one of Clockworks Signal Processing's A²B modules with key sections highlighted. The module was designed to make A²B experimentation easy—all components are one side and no 0402 parts are used (mostly 0603 and 0805) so represents the upper bound for PCB real estate. A board with components on both sides and using smaller components with a compact layout can be done in half of the space. This module can act as a master or slave. A design that targets a specific mode of operation (along with the local or bus power choice) would need less parts.

Summary

In this article, we looked at the details of the electrical interfaces to the ADI AD242x chips. Guidance on the passive components needed for the interface was presented. In the final part of this article series, we'll look at the tools and hardware available to create systems for development and evaluation.

About the Author

Brewster LaMacchia started in signal processing working on military satellite communications systems. Since then he has worked on a wide range of signal processing systems, including a number of years on video processing and surround sound audio for consumer home theaters. Recently, he started Clockworks Signal Processing to develop open-source hardware for multichannel signal processing. He also dedicates time to public astronomy events to expose people to science and critical thinking. In the past 10 years, he's helped more than 7,000 people look through a telescope for the first time.

Circuit Designs for a Hollow-State Mixer

The Development of Recording Mixers

By
Richard Honeycutt
(United States)

Photo 1: New Zealand artist Johnny Cooper and his band are shown using a single-mic technique.

In this article, Richard Honeycutt discusses the beginnings of hollow-state mixers and shares some tips for DIYers who might want to design their own hollow-state recording mixer or assemble one from 500-series preamps and a home-brew summing amp.

The first commercially produced phonograph records made using electrical (rather than purely acoustical) recordings were released in the early 1920s. These records contained a single monophonic track. "The Symposium on Audio Perspective," printed in 1934, by J. C. Steinberg and W. C Snow, detailed the use of a three-channel audio system to pick up the Philadelphia Symphony Orchestra performance in the Academy of Science Hall in Philadelphia, PA; transmit it to Constitution Hall in Washington, D.C.; and reproduce it in that room. While this performance was not a music recording venture, it did employ the tools of the music recording art. Mixers were not employed, since each of the three microphones fed its own line amplifier and, ultimately, its own speaker.

The first three-channel commercial audio recordings were produced by Mercury records, beginning in late 1955. Again, no mixers were needed, since three omnidirectional mics were used. Recording was originally done on three-track magnetic tape, but Mercury later upgraded to three-track 35-mm magnetic film. The three-track recordings were mixed down in the studio for two-track stereo LP records, sold under the Mercury Living Presence brand. Incidentally, when CDs of these performances were released, they were mixed down from the original three-track masters. The 3:2 mixdown was accomplished on a Westrex mixer modified by Mercury engineers.

Surround-Sound Techniques

A variety of surround-sound techniques were developed in the movie industry, beginning with Disney's "Fantasound" process used in creating and exhibiting the movie *Fantasia* in 1940. No doubt some of these surround-sound experiments used multiple microphones for some or all tracks, and thus would have used mic mixers.

As late as the early 1960s, debate still raged in the pro-audio press about the relative merits of single-mic (see **Photo 1**) vs. multi-mic (see **Photo 2**) techniques for recording audio. As musical groups began to involve different instruments having more

Photo 2: The Lawrence Welk Orchestra often used multi-mic techniques.

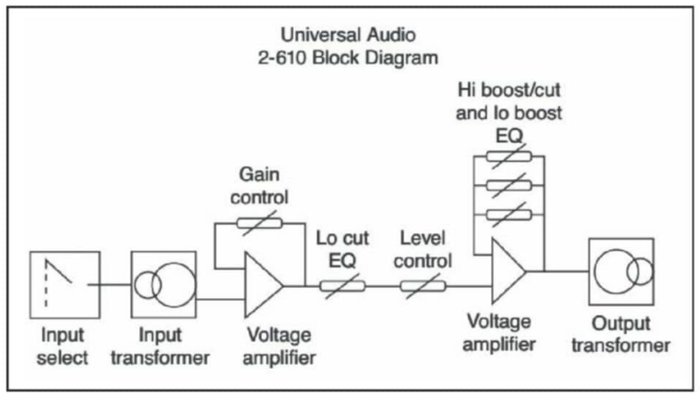


Figure 1: The block diagram of a UA 610 mixer channel strip shows the separate function blocks of the mixer.

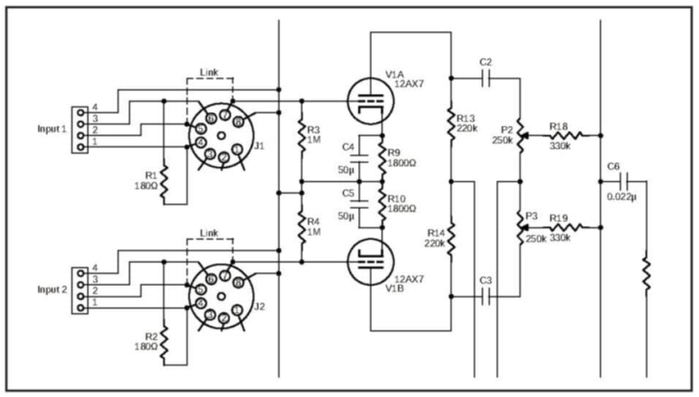


Figure 2: The preamps for channels 1 and 2 of the Altec 1567A mixer are triode-based. The junction of R13 and R14 is fed by 255V B+. The junction of P2 and P3 goes to the input capacitor of the program amplifier.

widely varying sound output levels (electric guitar, string bass, drum "trap" kits, brass instruments, and vocals), all needing to be balanced, multi-mic techniques essentially took over for many types of recording. Large orchestras might still use a single mic per channel, but rock, country, and jazz groups often needed more mics, and so mixers were required. To make a stereo recording of such a group, the mixer came to involve multiple output channels—for years, three was the standard number—with individual channel strips being assignable to the right, left, or center channel, or perhaps panned somewhere in between.

Since the late 20th century, most commercial recordings have also used multi-track techniques in which a single instrument, voice, or group of instruments or voices was assigned to a single track of a magnetic tape or a single digital bitstream. These multiple tracks are then mixed to the desired number of output tracks—two for a CD or MP3, more for surround sound. Following a dubious flirtation with quadrophonic (four-channel) LPs in the 1970s, and many experiments for SACD in the 2000s, surround sound remained predominantly the province of movies.

In multi-track recording, a mixer accepting mic and line inputs can be used to mix and route all inputs to the appropriate output track, or individual preamps can be routed to each track, with the completed multi-track recording then being mixed down to the desired number of output tracks. If multiple inputs will be mixed to some or all tracks, there could be several small mixers used, rather than a single large one. As you can see, depending upon the preferences of the recording engineer, a session can employ any of various approaches, involving one or more mixers.

As discussed in the October 2020 Hollow-State Electronics article, an engineer may prefer the clean sound and versatile control/EQ capabilities of a solid-state mixer, or the warmth and feel of one or more hollow-state mixers. In the latter case, the engineer might opt for a classic mixer (e.g., a UA 610), or perhaps prefer to "roll his/her own" hollow-state or hybrid mixer. **Figure 1** shows the block diagram of a UA 610 channel strip, in which we can see (from left to right) the input transformer, preamp ("voltage amplifier"), EQ, and output transformer. The output of the channel strip would feed an attenuator driving the three "mix buses," in which all the combined input signals would appear. The 610 had three mix buses, similar to those of an L-C-R mixer for non-surround cinema or some live-sound systems.

Preamps in Hollow-State Mixers

The preamps of most classic mixers used triode gain stages. **Figure 2** shows a typical mixer preamp stage from an Altec mixer. These preamps used input transformers to provide noise-free gain from the typical 50 Ω to 150 Ω mic element to the high-Z triode grid, and also to provide a balanced input to prevent hum from common-mode-to-transverse-mode conversion. Typically such input transformers had 38 Ω and 150 Ω input windings, and 50 kΩ secondaries, providing gains of about 36 and 18, respectively. The 1 MΩ grid resistor of each triode made the input resistance of the preamp stage about 1 MΩ, so the actual input resistance to the transformer was about 760 Ω or 3 kΩ, depending upon which winding was fed. Despite decades of misinformation, no attempt was made to match input impedances. Instead, circuits were designed for good efficiency, which requires the load impedance to be considerably higher than the source resistance.

As discussed in the September 2014 Hollow-State Electronics article, modern hollow-state mic preamps may use more advanced circuitry than the simple common-cathode stage employed in the classic mixers. An example is the cascode circuit shown in **Figure 3**, which provides extended high-frequency response by eliminating Miller effect.

Photo 3 shows a Roll RMS5A7 current-production hollow-state mic preamp using a two-

Photo 5: The Aphex Dual RPA 500 preamp uses a patented new hybrid tube technology.

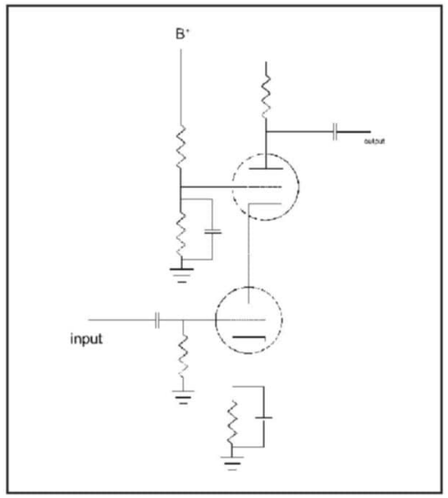


Figure 3: The cascode circuit is essentially a common-cathode amplifier feeding a common-grid amplifier.

Photo 3: The Roll RMS5A7 tube preamp can be used in a 500-series rack. (Photo courtesy of Roll Music Systems, Inc.)

Photo 4: The LaChappell 583s MKII, 500 series tube preamp is another excellent performer. (Photo courtesy LaChappell Audio, Inc.)

Resource

J. C. Steinberg and W. C. Snow, "The Symposium on Audio Perspective," 1934, www.aes.org/aeshc/docs/bell.labs/auditoryperspective.pdf.

16065656

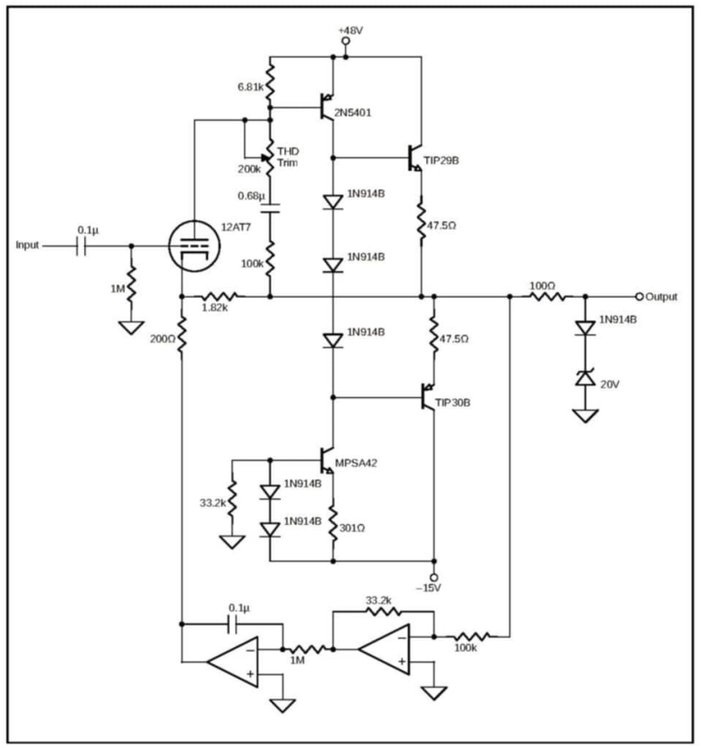


Figure 4: This schematic of the patented Aphex RPA shows the unusual design features. (Image courtesy of Aphex)

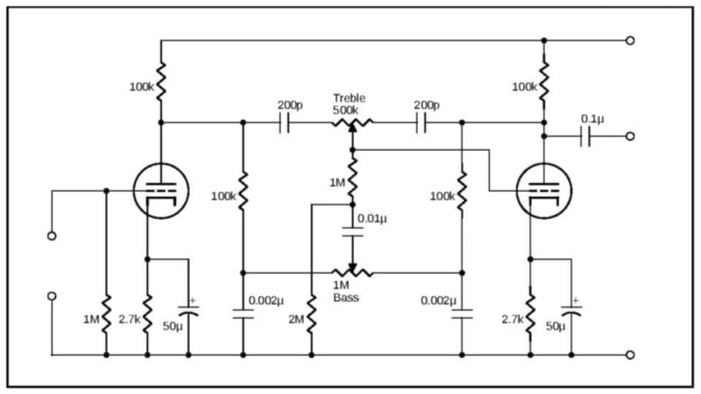


Figure 5: This Baxandall tone control has a center frequency of about 1000 Hz. (Image courtesy of Tube DIY Asylum)

stage 12DW7 amplifier circuit that provides a 1200 Ω input impedance, and drives a recommended load impedance of >10 kΩ. Its gain is switchable in 3 dB steps from 33 dB to 66 dB, and it also has an input attenuator providing from 0 to -10 dB of adjustment, plus a -20 dB input pad. This preamp operates from 12 V to 48 V phantom power and passes phantom power through to the mic.

Photo 4 and **Photo 5** show a LaChappelle 583s and an Aphex Dual RPA 500 hollow-state mic preamp, respectively. These preamps also operate from phantom power, which they pass through to the mic. The schematic diagram of the Aphex unit is shown in **Figure 4**. Although it is a hybrid design, the op-amps cannot be overdriven, so any overdrive distortion produced in the preamp comes from the 12AT7 triode.

These preamps are designed to plug into "500-series lunchbox" racks, which can be obtained with 1, 2, 3, 4, 6, 8, 10, or 11 slots to accommodate different numbers of preamps. A populated 500-series rack can serve as a high-performance hollow-state preamp set for a home-brew mixer.

To complete the mixer, you'll need to add EQ channels as desired and what was called a "program amplifier" in the classic mixers.

EQ and Summing

If you want extensive EQ capabilities, you may want to use the 500-series preamp rack with a solid-state mixer to provide the EQ and actual mixing functions. But if you are happy with simpler EQ, you can build a Baxandall tone control stage as shown in **Figure 5**. The tube is a 6SL7 dual triode, but a 12AT7 could be used.

Figure 6 shows the program amplifier of the Altec 1567A mixer. More current terminology for "program amplifier" is "summing amplifier." The way this summing amplifier works is that the 330 kΩ isolation resistors R18, R19, R20, and R21 prevent the signal from one triode stage from being loaded down when the potentiometers of the other stages are set at minimum (zero Ω to ground). The signals from each triode—V1a and b and V2a and b—plus the balanced line input signal, are summed into the output according to the potentiometer settings.

With 500-series preamps providing about 66 dB of gain, and recording mics outputting about -47 dBV (Royer 121 ribbon mic) to -32 dBV (Neumann U67 capacitor mic) at 1 pascal (~94 dBSPL), the summing amplifier will have about a 19-34 dBV signal level at its inputs. This is typically enough signal for the output of a summing amp or mixer, meaning that the

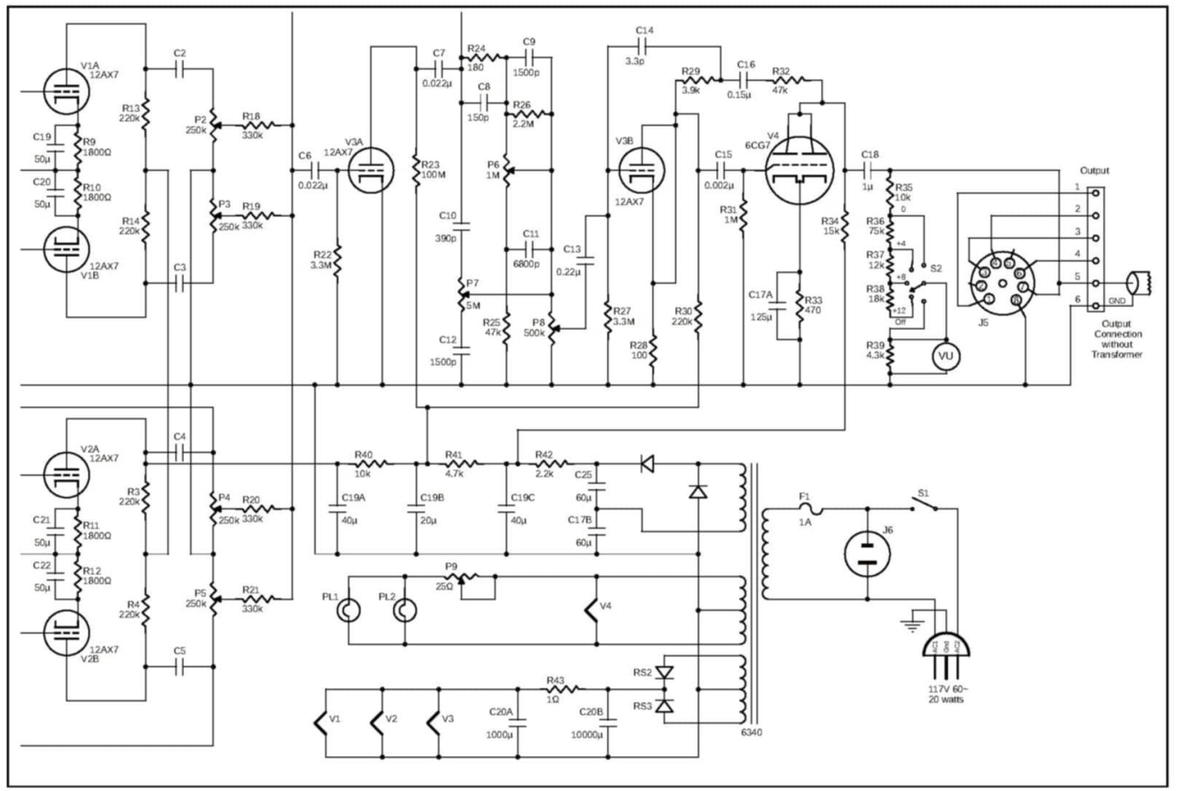


Figure 6: The Altec 1567A mixer's program amplifier has a simple summing circuit to mix the signals. The horizontal line between V1 and V2 is grounded. The junction of R18 and C6 is fed directly from the mixer's balanced line input terminals.

nominal gain of the summing amp could be unity.

For comparison, a Yamaha DM2000VCM digital mixer has a nominal output of +4 dBV and a maximum output before clipping of 18 dBV. A higher gain is a possibility, but we must avoid designing the mixer so that the summing stage is overdriven. The noise level at the output is almost completely determined by the noise level of the first preamp stage, unless there is enough loss in an EQ stage to reduce the level at its output below the level at the output of the first preamp stage.

A professional digital mixer may have an RCA output jack with an output impedance of 600 Ω, designed to drive a 10 kΩ unbalanced load, and a 75 Ω XLR output designed to drive a 600 Ω balanced load. The Altec 1567A is specified for a maximum output of +18 dBm. With an output transformer feeding a 600 Ω load, this translates to a voltage of about 15.8 dBV. Thus, the 6CG7 output stage of the 1567A will probably provide enough output level; there's no need to resort to a higher-power output stage such as the dual-6V6 stage used by the old Collins Mixers, which were designed to drive long telephone lines connecting the mixer to a remote radio transmitter.

Certainly dyed-in-the-wool DIYers might want to design their own hollow-state recording mixer from the ground up, including mic preamps, EQ stages, and summing amp, but assembling one from 500-series preamps and a home-brew summing amp—with or without EQ stages—is certainly a tried and proven way to achieve excellent results. Those audio enthusiasts wanting more engineering adventure might want to design the summing amp rather than copying a classic (e.g., an old Altec). Even in this case, beginning by analyzing one or more of the classic mixers' circuits can prove illuminating!

Article Update

Comparing IMD (and THD) Measurements

Following Michel Nieuwenhuizen's two-part article that was published in audioXpress*' October 2020 and November 2020 issues, we received a letter from Burkhard Vogel with a question that motivated a reply from the author.*

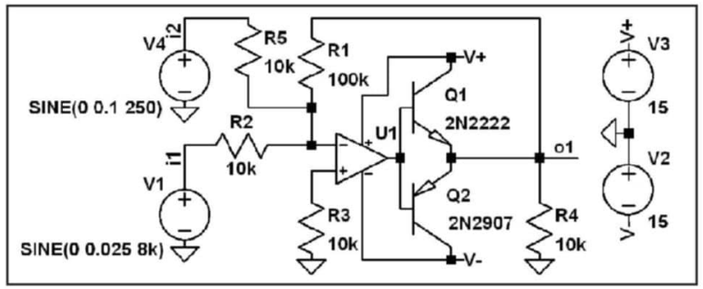


Figure 1: Example test amp for IMD purposes (A configured: 250 Hz/100mV + 8 kHz/25mV)

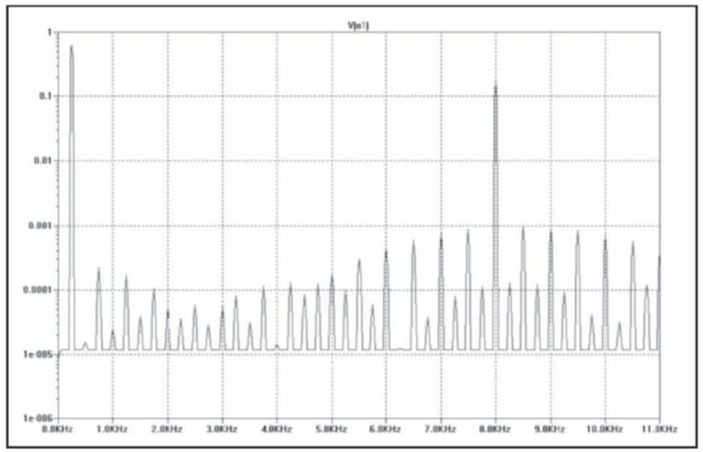


Figure 2: IMD according to the standard approach with A specs

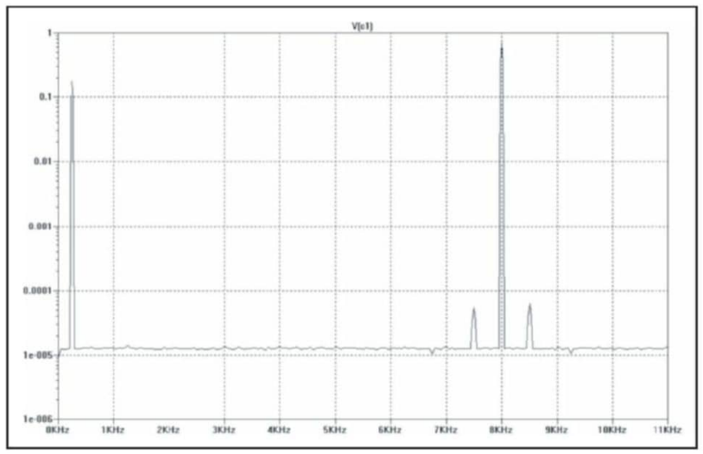


Figure 3: IMD according to the M. N. approach with B specs

Reader's Question:

I've read with great interest Mr. Michel Nieuwenhuizen's two articles in audioXpress October an November 2020 issues about his mic amp development. I guess it's a very clever design and a wonderful solution.

However, there is one point that does not match my knowledge about IMD measurements. It's his approach to measure it. In contrast to the standard "SMPTE" with two tones f1 & f2 and 4:1 amplitude relationship (f1 << f2), he has chosen the other way around:

f1 still << f2 but amplitude relationship 1:4 (see his Figure 12, *audioXpress* November 2020). This manuever automatically leads to a very much better looking IMD picture if we compare it with the result of the standard approach.

A short exercise should demonstrate what I mean. I performed two simulations. I took a simple and IMD producing amp [1] (**Figure 1**) and worked with two approaches:

- A specs with f1 = 250 Hz and f2 = 8 kHz (= last century's recommendation in DIN 60268-3), amplitude relationship 4:1 (other approaches work with 60 Hz/7 kHz, 4:1) [2], shown in **Figure 2**.
- B specs also with f1 = 250 Hz and f2 = 8 kHz and 1:4 amplitude relationship (Michel Nieuwenhuizen's approach as shown in **Figure 3**).

(For a better visibility, I've cut the frequency range in the figures down to linear 0 Hz ... 11 kHz.)

IMD (A) = 1.04 %, IMD (B) = 0.016 %

(Burkhard's Mathcad Worksheet can be found in the Supplementary Material on the audioXpress website).

Thus, my questions to the author:

Why did you take the B approach and how would Figures 10 and 11 look like with the standard-based A approach?

— Burkhard Vogel

Supplementary Material

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

References

[1] G. Brocard, *The LTSpice Simulator*, Swiridoff Verlag, 2013)

[2] B. Metzler, *Audio Measurement Handbook*, Audio Precision, 1993.

Author's Response:

Thank you for your letter. I read your comments with great interest. The choice of frequencies and amplitudes I used are indeed not standard. Somewhere during my experiments I settled for them. As far as I can remember, I got them from an article. One advantage of using the lower frequency of 1000Hz is that it gives the opportunity to see THD and IMD in one picture. Having the higher amplitude at 1kHz also helps to see low level THD components, which are otherwise obscured by the low-frequency noise of the simulation.

When I read your comments, I was surprised that the difference between the distortion components is so large when only the amplitudes of the two sinus components are interchanged. I would expect that for a decent linear Class A circuit it would in fact just swap some IMD and THD component amplitudes but little more. In order to see if I was right, I repeated some of my simulations using the reverse amplitudes (100 Hz 100 mV, 1000 Hz 25 mV), using Micro-Cap. First, I have **Figure 1**, the reference circuit (*audioXpress*, October 2020, Figure 1).

We see that in **Figure 2**, the harmonic distortion is approximately 20 dB lower at 3 kHz and at least 10 dB at 4 kHz but the harmonic distortion at 200 Hz and 300 Hz is 20 dB higher.

The intermodulation in Figure 2 is about 10 dB higher at 800 Hz and 1200 Hz but 10 dB lower at 1900 Hz and 2100 Hz. So yes there are differences, but the overall resulting THD and IMD are about the same. I also repeated the simulations with 250 Hz and 8 kHz, but with a bandwidth up to 30 kHz because we need to look at the higher harmonics of 8 kHz as well (see **Figure 3** and **Figure 4**). Here we see the same effect as in the previous figures.

In Figure 4, the distortion is higher at 250, 500, 7500, and 8500 Hz, but about the same amount lower around 16 kHz and 24 kHz.

As a further test I also repeated the simulations of my "extended circuit" (see Figure 13 in the second part of the article, *audioXpress*, November 2020). The results of the repeated simulations are shown in **Figures 5–8**. In Figure 6 probably the 200Hz component is higher but this is not visible.

In Conclusion

New simulations show that swapping the amplitudes and/or changing the frequencies has very little overall effect, in any case much less than predicted by your simulations. The circuit you simulated is very different from mine. It is a Class B amplifier, which has a completely different behavior with regard to distortion. In my Class A circuit, distortion will rise with higher input amplitude while a Class B amplifier, as simulated by you, has a distortion that becomes higher with lower amplitude because of crossover distortion.

I would like to thank you for bringing this issue to my attention, but I see no reason to doubt the simulations and measurements of my circuits.

—Michel Nieuwenhuizen

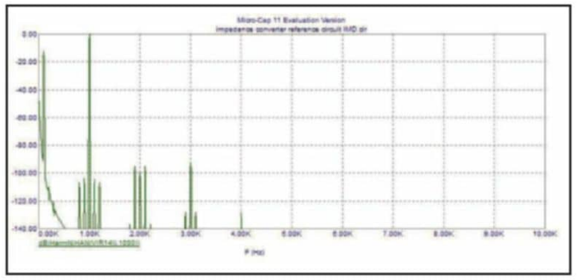


Figure 1: As in the article: 100Hz 25 mV, 1000Hz 100mV

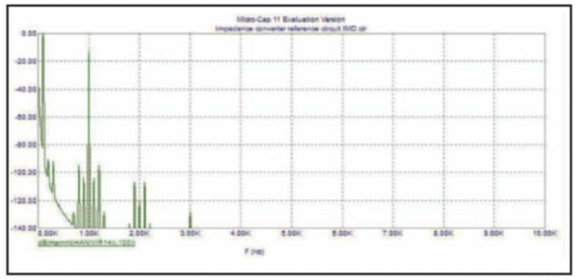


Figure 2: With reversed amplitudes 100Hz 100mV, 1000Hz 25mV

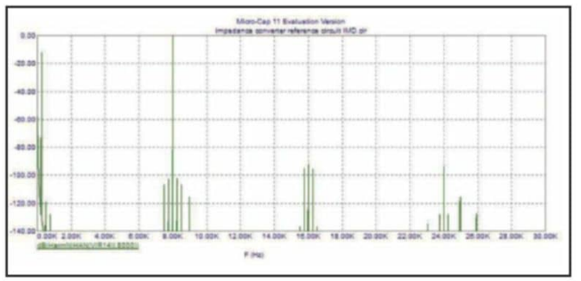


Figure 3: 250Hz 25mV and 8000 Hz 100mV

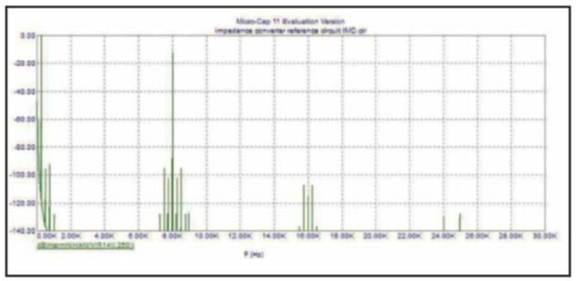


Figure 4: 250Hz 100mV and 8000 Hz 100mV

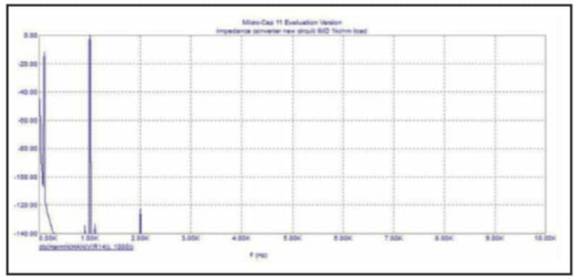


Figure 5: Extended circuit 100Hz 25 mV, 1000Hz 100mV

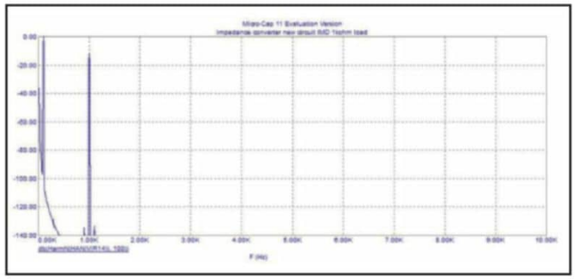


Figure 6: Extended circuit 100Hz 100mVm 1000Hz 25mV

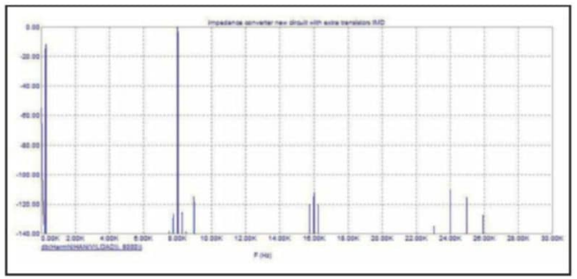


Figure 7: Extended circuit 250Hz 25mV, 8000Hz 100mV

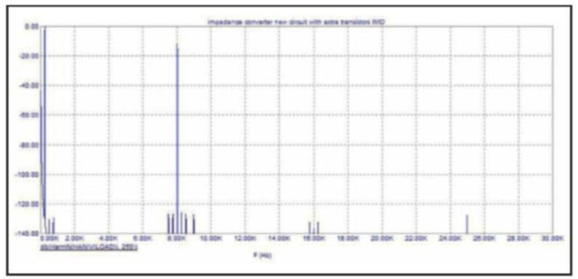
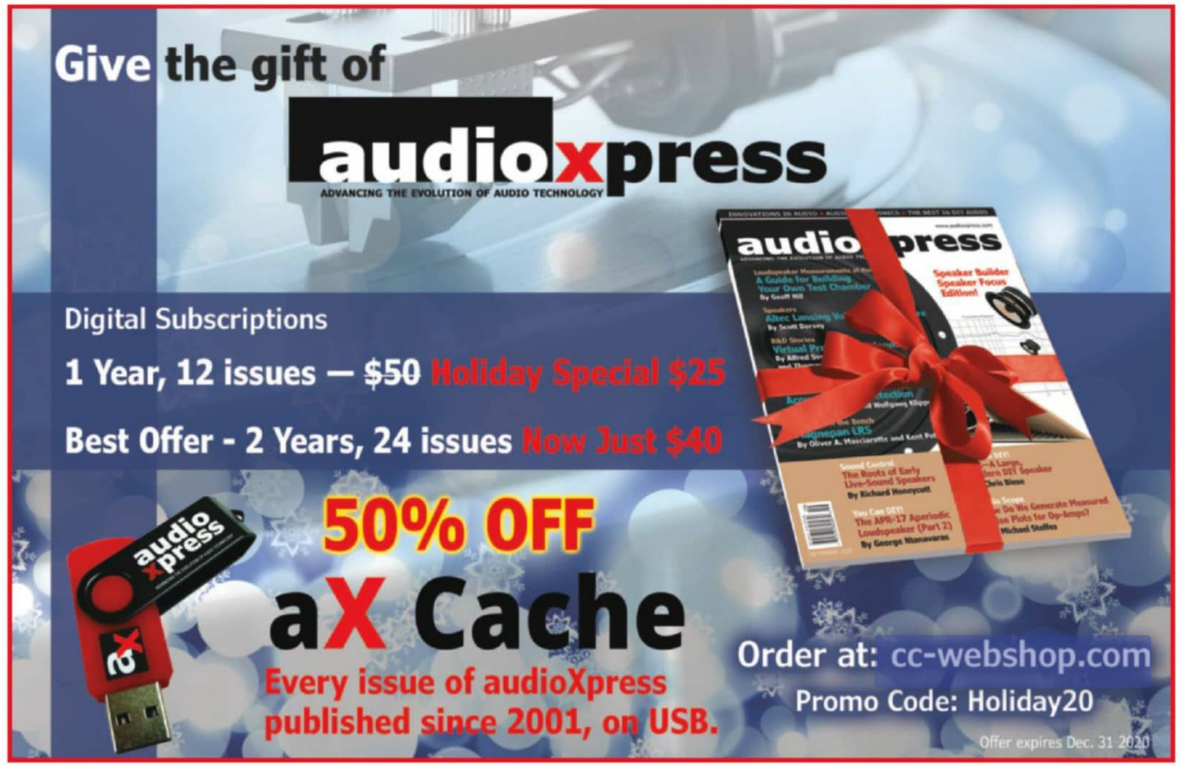


Figure 8: Extended circuit 250Hz 100mV, 8000 Hz 25mV



TRIAD

ELECTRONIC TRANSFORMERS

- Original Equipment
- Replacement
- Amateur

TRIAD MAGNETICS, INCORPORATED

460 HARLEY KNOX BLVD., PERRIS, CA 92571

www.TriadMagnetics.com

MODERN DESIGN
MEETS VINTAGE SOUND

Dayton Audio HTA100BT 100W Hybrid Stereo Tube Amplifier
with Bluetooth and Subwoofer Output

Debates may rage between whether solid-state or tube amplifiers produce a better listening experience; however, Dayton Audio solves that dispute! The HTA100BT Hybrid Stereo Tube Amplifier successfully employs contemporary amp technology that faithfully pays homage to vintage amp design. By harnessing the power and efficiency of solid-state class AB amp topology and a vacuum tube preamp section, the HTA100BT authentically reproduces tube amps' warmth and character with the flexibility the modern world expects.

- Authentic vacuum tube warmth and character with class AB amplifier efficiency produces an articulate and natural sonic experience
- Powerful 50 watts per channel of audiophile fidelity that is sure to please even the most discerning ear
- Convenient input connections including USB, optical, coaxial, stereo RCA jack, phono preamp, and Bluetooth V5
- Mono RCA subwoofer output gives you the ability to enjoy a 2.1 system when connected to an active sub
- Brushed aluminum housing with vintage front panel VU meters lends a clean, modern look with a nod to classic aesthetics
- Short circuit protection circuitry to prevent amp failure

Custom OEM inquiries welcome.

SAVE $12
off $100 order!
Code: AXMQ42020
Expires 12/31/2020

DAYTONAUDIO
Your path to great sound.™

Distributed By:
PARTS EXPRESS
AUDIO COMPONENTS & SOLUTIONS

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