

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Market Update

Audio Amplifier Technology and Design

By J. Martins

Professional Power Amplifiers A Market In Recovery

By James Kirby (Futuresource)

You Can DIY!

The DH-220C MOSFET Power Amplifier (Part 1)

By Bob Cordell and Rick Savas

The VSCS Amplifier

By Paul Marchese

Fresh From the Bench

NAD Electronics M33 Masters BluOS Streaming DAC Amplifier

By Stuart Yaniger

Hollow-State Electronics

Negative Feedback Amplifier Circuits

By Richard Honeycutt

Audio Electronics

Low Distortion Voltage- Controlled Floating Resistor

By Dimitri Danyuk

Fresh From the Bench

Nuheara IQbuds2 MAX Hearing-Assist Earphones

By Brent Butterworth

50
Years





The Original Western Electric 300B

Available in Matched Pairs, Matched Quads, and Singles
Order today at westernelectric.com • 5-Year Limited Warranty

The Western Electric 300B is now available for purchase in limited quantities. Tubes are made by hand in the USA at our latest manufacturing facility, the Rossville Works.

Orders are allocated for shipment first-in first-out. In some cases you may be redirected to an authorized dealer to better serve you in your region. Learn more about our effort to continue quality, American manufacturing at westernelectric.com.

Western Electric

MAKER OF ELECTRON TUBES & HIGH FIDELITY

Founded in 1869, we're an American manufacturer with an enduring relationship with quality. Today, we make stunning listening experiences through innovative audio equipment and legacy hi-fi like the 300B vacuum tube.



© 2021 Western Electric • info@westernelectric.com
410 Chickamauga Ave, Suite 300 • Rossville, GA 30741 • +1 (404) 352-2000



**HAMMOND
MANUFACTURING®**



MANUFACTURING & DESIGNING TRANSFORMERS & ENCLOSURES SINCE 1917

Chassis, Plates & Covers



Tube Output Transformers



Plate & Filament Transformers



Open & Enclosed Chokes



USA

Tel: (716) 630-7030

Fax: (716) 630-7042

Canada

Tel: (519) 822-2960

Fax: (519) 822-0715

UK

Tel: 01256 812812

Fax: 01256 332249

Australia

Tel: 8-8240-2244

Fax: 8-8240-2255

www.hammondmfg.com



July 2021

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. 2021

Printed in the US



Multiple Ways to Amplify

Have you noticed the important evolution that is taking place in professional audio applications, particularly with commercial installations? The last year and a half caused a business stagnation for many manufacturers of professional audio equipment, but those companies have continued to expand on many of the concepts, features, and technologies that were already under development in 2019, and further accelerated the research to improve their abilities to control, manage, supervise, and monitor vital systems remotely.

As in many other technology areas, the pandemic circumstances led to the accelerated adoption of web- and cloud-based tools that were merely being “considered” and suddenly became obviously essential.

Professional audio amplifiers are now sophisticated, multifunction systems that are able to deal with line and speaker levels, more demanding loads, and offering the highest levels of power per channel ever. In terms of design, professional audio amplifiers are also now evolved concepts, combining audio switching, signal processing, conversion, and multiple interfaces. More important, they are now fully connected systems, with embedded intelligence to support any type of speaker configuration that is connected. “Speaker controllers” are increasingly the amplifiers themselves, individually or stacked.

In this edition focused on audio amplifiers, we received more interesting announcements from the professional audio side, but also from the other extreme of the product segment—from companies that deal with very high-volume applications. The fact that those extremes are now offering the highest levels of sophistication says a lot about the impact of audio amplification technology. At the component level, we’ve seen a continuous flow of announcements for the past 18 months. As the demand for audio systems in all types of products continued to rise, audio amplification devices have also evolved at an astonishing pace—in performance, efficiency, and features.

When most of us think about audio amplifiers, we hardly ever think about the integrated devices that are today manufactured in the millions and used to power those tiny speakers in mobile phones, tablets, and laptops, and even featured in smart appliances. Those audio amplification chips are not only essential for such systems, but they have achieved a level of sophistication that will influence all other audio designs in other application segments. As an example, just before closing this edition, we wrote about a new revolutionary speaker design for home audio: the new Syng Cell Alpha wireless speaker. Coming from an experienced design team that has worked previously with some of the largest technology companies, this is a truly revolutionary concept in many aspects, and was made possible by the use of new, high efficiency gallium nitride (GaN) transistors. In this case, from GaN Systems, a company that is already powering products praised in high-end circles, and is now getting ready to make a large-scale appearance in a variety of applications. And significantly, some of those applications we don’t even associate with high-quality audio.

New power semiconductors are a clear example of what is in place for new high-fidelity sound systems. But so are new system-on-chips (SoCs) as we’ve seen recently from companies such as Qualcomm, which offers fully integrated platforms including audio codecs, DACs, and sophisticated DSP, integrated with the power amplifier stage. All engineered to work together to provide the best audio quality experiences possible from mobile devices, portable speakers and complete home entertainment systems.

For many audio manufacturers, these solutions available with complete development resources at attractive prices, are hard to resist. But the most important thing is that, when combined with well-tuned designs, they have the potential to offer incrementally better performance.

J. Martins

Editor-in-Chief

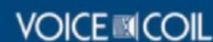
The Team

Publisher:	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, Dimitri Danyuk, 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, Michael Steffes, Stuart Yaniger**

OUR NETWORK



SUPPORTING COMPANIES

ACO Pacific, Inc.	55	Lundahl Transformers AB	19
Acoustic Sciences Corp.	65	Menlo Scientific, Ltd.	59
All Electronics Corp.	61	Parts Express International, Inc.	68
APEX Audio	47	Powersoft SpA	45
Celestion	15	Purifi Audio	23
COMSOL, Inc.	9	Redco Audio	65
Danville Signal Processing, Inc.	37	SB Acoustics	53
Hammond Manufacturing, Ltd.	3	Triad Magnetics	67
Hypex Electronics B.V.	29	Western Electric	2
KAB Electro-Acoustics	55	ZUMI Systems	49
Lenbrook International	25		

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.

The Audio Voice

Sign up now for our **FREE** weekly e-newsletter



Weekly Newsletter for the Audio Industry, Professional Designers and Audio Enthusiasts
Product Design | Audio Electronics | Acoustics | DIY | Innovations in Audio



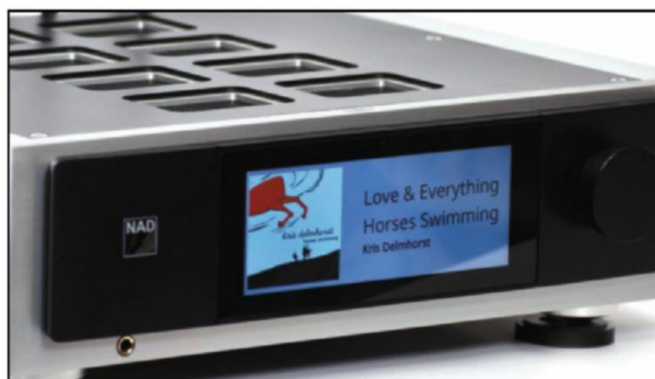
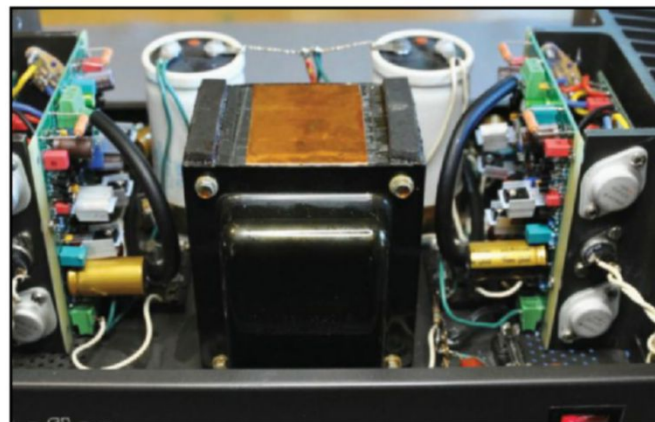


ax Cache

Every issue of audioXpress published since 2001, on USB.

Order at: cc-webshop.com





● Features

8 Nuheara IQbuds2 MAX Hearing-Assist Earphones Digitally Enhanced Hearing Devices

By Brent Butterworth

More than five years in development and after multiple iterations, Australian startup Nuheara achieved its goal with the Nuheara IQbuds2 MAX, a bridge toward truly smart hearables. The design uses digital technology to bring hearing aid functionality into true wireless earphones. Brent Butterworth gives them a listen.

16 NAD Electronics M33 Masters BluOS Streaming DAC Amplifier The Complete Package for a Wireless Life

By Stuart Yaniger

After reviewing the compact but mighty M10 Masters Series streaming amplifier from NAD Electronics, Stuart Yaniger simply had to try the new flagship NAD M33 Masters streaming DAC amplifier. This was the first mainstream product to implement the Purifi Audio Eigentakt (self-clocking) amplifier modules, but there is a whole lot more to make it a very complete package for a musical life. The article reviews the experience and performance measurements.

40 The DH-220C MOSFET Power Amplifier (Part 1) The Circuit

By Bob Cordell and Rick Savas

This reference DIY project describes an upgrade construction project for the legendary Hafler DH-220 lateral MOSFET power amplifier, but it can also be used to upgrade other Hafler models or even inspire new designs.

50 A New Type of Amplifier for Direct Driving Loudspeakers (Part 1) Theory Behind the Design

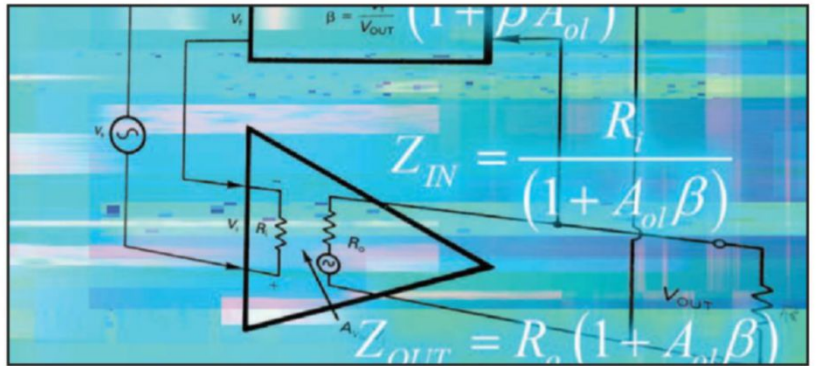
By Paul Marchese

This article describes an amplifier that is a voltage source at low frequencies, then acts as a current source at higher frequencies. It can also be configured to be any ratio of voltage source vs. current source and/or be frequency independent.

56 Low Distortion Voltage-Controlled Floating Resistor

By Dimitri Danyuk

Learn more about this voltage-controlled floating resistor project. The circuit consists of a JFET and an optocoupler with a photovoltaic output, and it can serve as an effective substitute for the H11Fxm Photo-FET series from ON Semiconductor.



● Audio Amplifier Technology and Design

MARKET UPDATE

24 The Multiple Faces of Amplification Evolution

By J. Martins

Addressing audio amplification trends and perspectives, this article highlights products from different application segments to illustrate the state of the industry.

38 Professional Power Amplifiers

A Market in Recovery

By James Kirby

Futuresource, a leading provider of professional audio market research, publishes market reports across a range of categories, including loudspeakers, amplifiers, microphones, mixers, headphones, and more. This latest article discusses the professional amplifier market and coincides with the release of its new Professional Power Amplifiers Market Assessment Report 2021.

● Columns

HOLLOW-STATE ELECTRONICS

62 Negative Feedback Amplifier Circuits

By Richard Honeycutt

● 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



audioexpresscommunity



linkedin.com/company/audioexpress

Nuheara IQbuds2 MAX Hearing-Assist Earphones

Digitally Enhanced Hearing Devices

By
Brent Butterworth



More than five years in development and after multiple iterations, Australian startup Nuheara achieved its goal with the Nuheara IQbuds2 MAX, a bridge toward truly smart hearables. The design uses digital technology to bring hearing aid functionality into true wireless earphones. Brent Butterworth gives them a listen.

Prescription hearing aids typically cost thousands of dollars, but relatively inexpensive true wireless earphones often incorporate similar technology and hardware. On at least a fundamental level, all it takes to add hearing-aid functions to these earphones is an app and some firmware modifications. So it's not surprising to see these functions appearing in new Apple earphones, or to see new companies emerge in an attempt to create a new class of products intended to augment hearing while providing great entertainment.

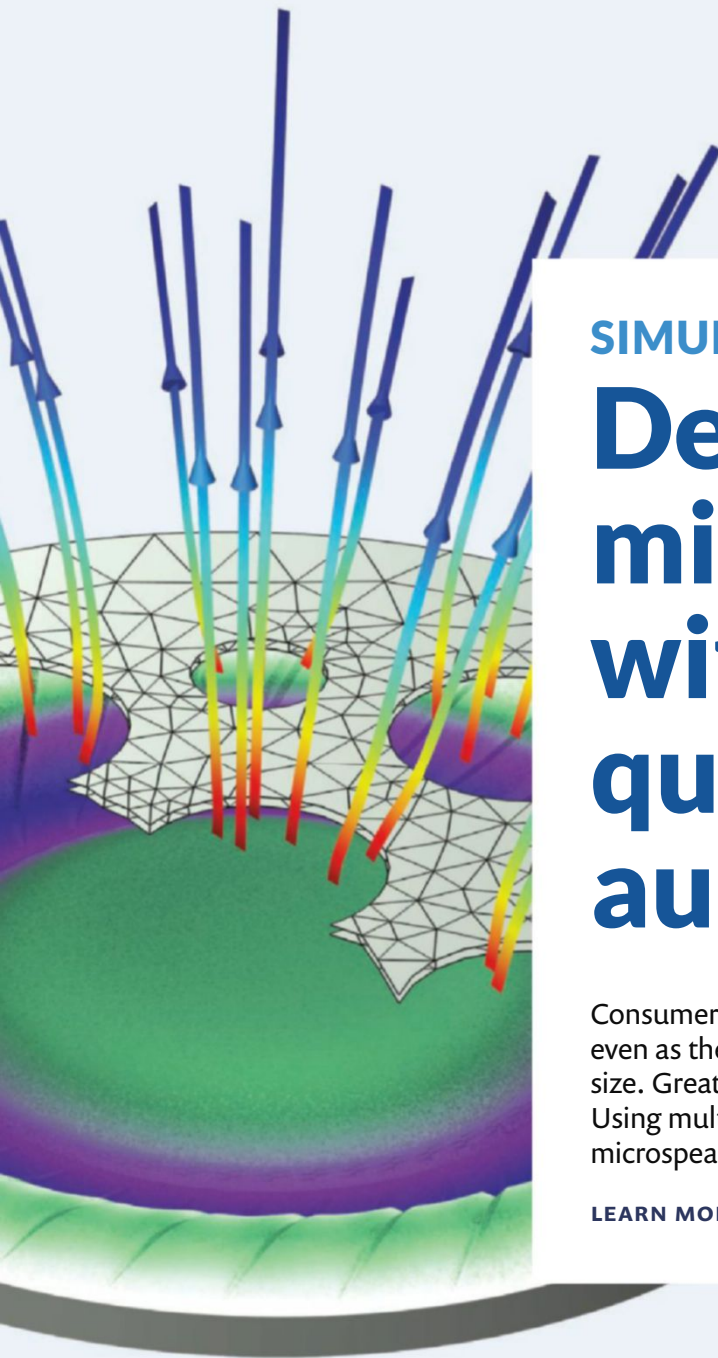
Nuheara is one of those companies, and the IQbuds2 MAX is its flagship product—a \$399 set of true wireless earphones that tests the wearer's hearing, applies a corrective response curve, and also adds such convenience features as active noise cancelling and IPX4 water resistance (**Photo 1**).

The IQbuds2 MAX webpage refers to the product as a “situational use hearing device,” and says that it's not designed for all-day use as hearing aids are. Presumably, a hearing-impaired person would wear the IQbuds2 MAX to parties and restaurants to help with speech comprehension, and also use them for music, podcast listening, and TV watching, if the TV has a Bluetooth transmitter or an outboard transmitter is connected.

The earphones include an app that administers a hearing evaluation that Nuheara calls EarID. It's based on the NAL-NL2 test, created by Australian Hearing's National Acoustic Laboratories and licensed by HEARWorks. NAL-NL2 focuses on speech intelligibility. HEARWorks says the test is licensed by 18 different hearing aid and audiology equipment companies, and is used for more hearing aid fittings than any other test in the world.



Photo 1: This technical rendering shows an exploded view of the IQbuds2 MAX.



SIMULATION CASE STUDY

Design microspeakers with great sound quality for any audio device

Consumers are always going to expect the best sound quality, even as the components of audio devices continue to decrease in size. Great sound quality requires well-designed microspeakers. Using multiphysics simulation, engineers can test and improve microspeaker designs, like this industry example.

LEARN MORE comsol.blog/microspeakers



The COMSOL Multiphysics® software is used for simulating designs, devices, and processes in all fields of engineering, manufacturing, and scientific research.

The Product

The IQbuds2 MAX look a little bulky compared with most current true wireless earphones; physically, they're more comparable to models that were sold three or four years ago (**Photo 2**). Each one contains a 9.2mm dynamic driver. The battery life is rated at 5 hours for streaming music and 8 hours for hearing assistance only. The included charging case provides three extra recharges.

In addition to the standard Bluetooth SBC codec, the IQbuds2 MAX includes aptX Low Latency, which could reduce lip-sync errors for TV viewing, depending on the latency of the display. To use the IQbuds2 MAX for TV viewing, you'll need to add a Bluetooth transmitter. Nuheara offers the \$99 IQstream TV transmitter, which is compatible with aptX Low Latency.

Each earphone incorporates three microphones. The multiple microphones facilitate several features, including active noise cancelling, focused listening, and phone conversations. The only

controls are a single touch-sensitive button on the side of each earpiece.

Most of the power of the IQbuds2 MAX is in the smartphone app (**Photo 3**). In addition to accessing Ear ID, it also offers numerous listening modes. These mix between noise cancelling and hear-through mode (allowing external sound in through the microphones) and employ different types of filtering. There are seven basic modes: workout, street, home, office, restaurant, driving, and airplane. Four of these can be saved as favorites so you can select among them by touching the right earpiece. All can be altered to your liking using adjustments within the app. The app also enables the user to assign different functions to the controls on the left and right earpieces.

The adjustments are on four screens, three of which employ circular sliders. The one shown in **Figure 1** turns noise cancelling on and off and activates the World mode, which allow a hear-through function. Another, called Speech In Noise Control (SINC),



Photo 2: Here is a side by side comparison of the Nuheara earbuds with the Jabra's Elite Sport 65t and respective charging cases.

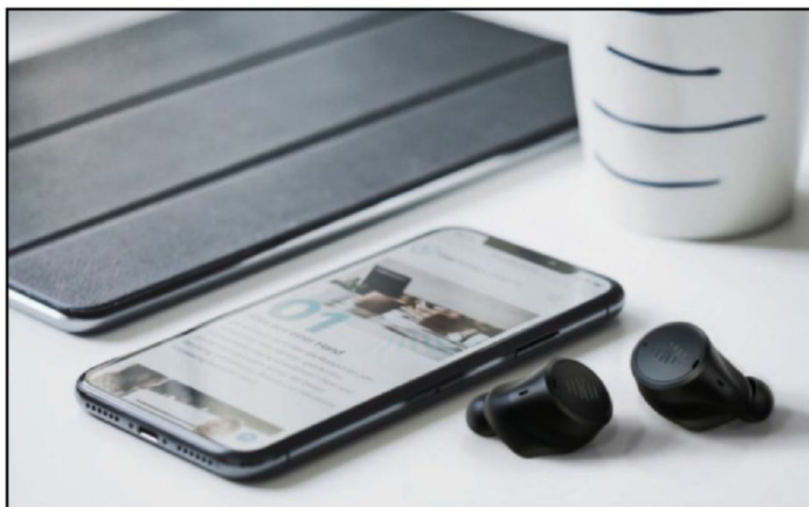


Photo 3: Most of the power of the IQbuds2 MAX is in the smartphone app.

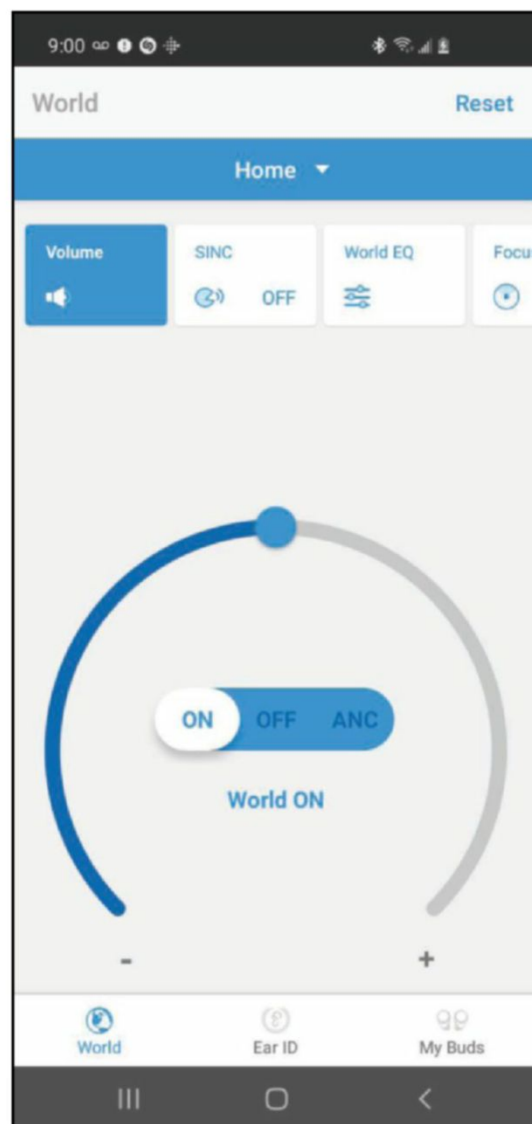


Figure 1: Noise-cancelling control screen from the IQbuds2 MAX control app

mixes between allowing all ambient sounds in and focusing solely on speech. A World EQ screen adjusts the center frequency of the filtering of the World mode, from bass to treble. A fourth screen, shown in **Figure 2**, turns the Focus mode on and off; this one is non-adjustable. Surprisingly, there are no adjustments affecting the sound of the earphones themselves, other than turning Ear ID on and off.

The Hearing Test

My hearing loss—last tested by an audiologist in spring 2020—is mild, and not something I notice on a day-to-day basis. In other words, about typical for a 59-year-old male. But it’s enough to show up on a hearing test, and I know the results well enough to know if results from an app-administered hearing test are in line with the results I’d get from an audiologist.

The test that Ear ID administers automatically is similar to an audiologist’s test in that it plays tones at six frequencies, from 500Hz to 6kHz, and

asks you to tap an onscreen button when you hear the tone. It gradually decreases the level of each tone to find your hearing threshold for that frequency in each ear.

First, it “listens” to find out if the ambient sound level is low enough to allow an accurate test. Once that test is passed, it plays tones to see if the seal of the earphones in each ear is adequate (**Figure 3**). Then the actual test begins, and it takes about 3 minutes per ear. Once it’s completed, the app creates an Ear ID for you. You can retake the test again as often as you like.

The chart the app provides is not as detailed or accurate as the test you’d get from an audiologist, but it was clear enough for me to get an idea of how it compares to the results from my audiologist’s evaluations. In the resulting chart, shown in **Figure 4**, the lighter the rings, the higher the frequency they represent. Ear ID correctly figured out that my right ear is better than my left. While there are no decibel markings on the chart, it

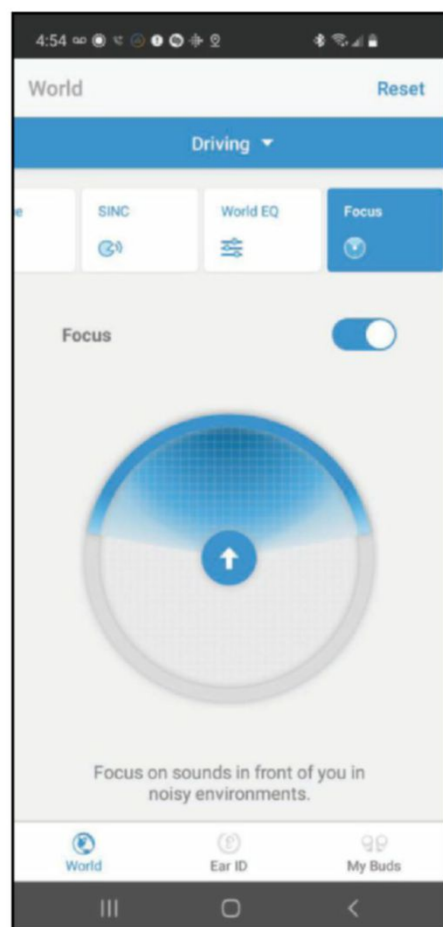


Figure 2: Focus control screen from the IQbuds2 MAX control app

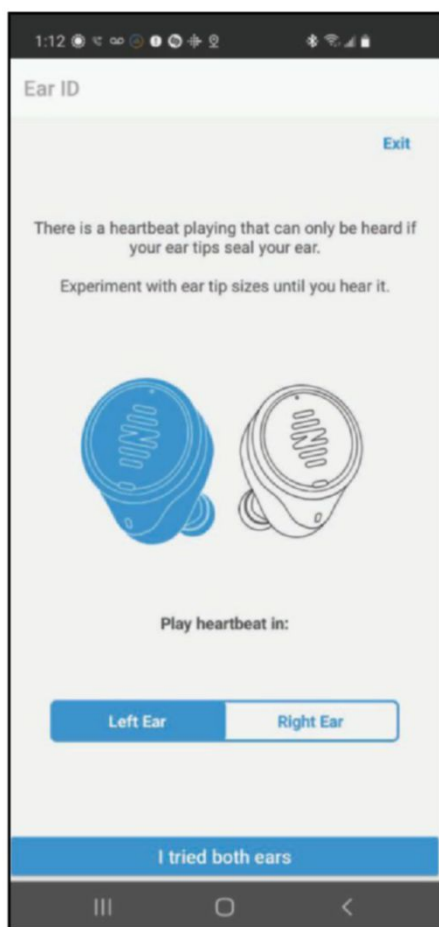


Figure 3: Seal-checking routine from the IQbuds2 MAX control app, which checks for a tight seal in the ear canal by testing the audibility of a low-frequency heartbeat tone

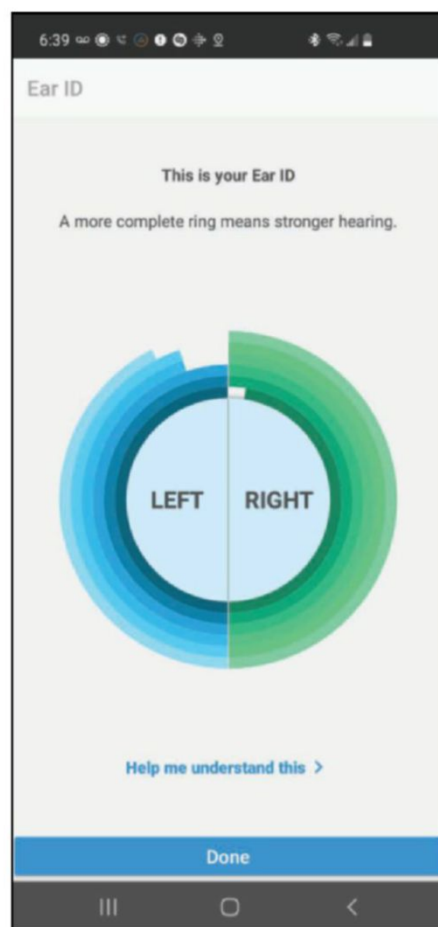


Figure 4: Ear ID hearing test results screen from the IQbuds2 MAX control app

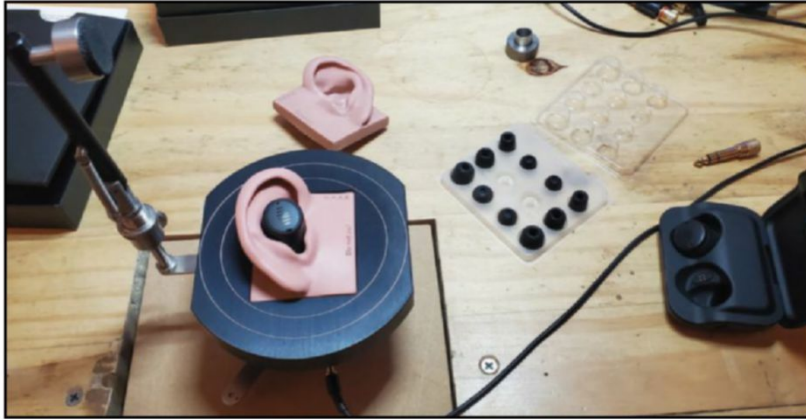


Photo 4: The IQbuds2 MAX are shown under measurement with the GRAS Ear and Cheek Simulator with anthropometric pinna.

seems to be saying that the loss in my left ear is mild at 3kHz and 4kHz, and a little worse in my right ear at 6kHz. It also says my right ear has no significant high-frequency hearing loss, but has a bit of low-frequency hearing loss in the 500Hz range. All of those results except the last one are consistent with my audiologist tests. I would guess that the earphone didn't seal as well in my right ear canal as it did in the left ear, resulting in a bit of leakage and a reduction in bass response that the app interpreted as hearing loss. Unfortunately, my right ear canal is especially large, and sometimes even the largest tips supplied with earphones don't seal completely.

Measurements

I performed measurements using GRAS Sound & Vibration test equipment: the Model 43AG, an RA0402 high-resolution ear simulator/coupler, and an M-Audio Mobile Pre USB interface with TrueAudio TrueRTA real-time analyzer software (**Photo 4**). Because the IQbuds2 MAX wouldn't pair with the

Mpow BH259A transmitter I use for measuring Bluetooth headphones, I couldn't use my usual Audiomatic CLIO 12 analyzer, and I had to rely on test signals generated by the AudioTool Android app in my Samsung Galaxy S10 phone.

Like many higher-end earphones, the IQbuds2 MAX mute themselves when they're removed from the ear, so they have to be "tricked" into working when placed in an ear simulator. Typically, earphones detect whether they're in the ear using a light sensor, which is easy to fool with a tiny wad of poster adhesive. But the IQbuds2 MAX also have another sensor, perhaps an orientation sensor or a way to sense the proximity of the other earpiece. The only way I could figure out to defeat the second sensor was to hold the ear simulator up next to my ear and transfer the earphone gently from my ear into the ear simulator. To say the least, this complicated my measurements even further.

I started with the most basic of measurements for earphones: frequency response. In this case, I measured the IQbuds2 MAX 1/12-octave-smoothed frequency response with Ear ID off, so I'd be measuring the "native" response of the driver and whatever DSP tuning Nuheara provides as a default. You can see in **Figure 5**, which compares the IQbuds2 MAX with two conventional true wireless earphones (the JBL Club Pro+ and the Status Audio Between Pro), that there's nothing unusual about the acoustical design of the IQbuds2 MAX. So that suggests these should be pretty good earphones for listening to music, with or without Ear ID.

To test the hearing-assistance functions, I used an array of six Dayton Audio B652 speakers mounted around my lab plus a Sunfire True Subwoofer Jr., all playing uncorrelated pink noise at 60dB. This is

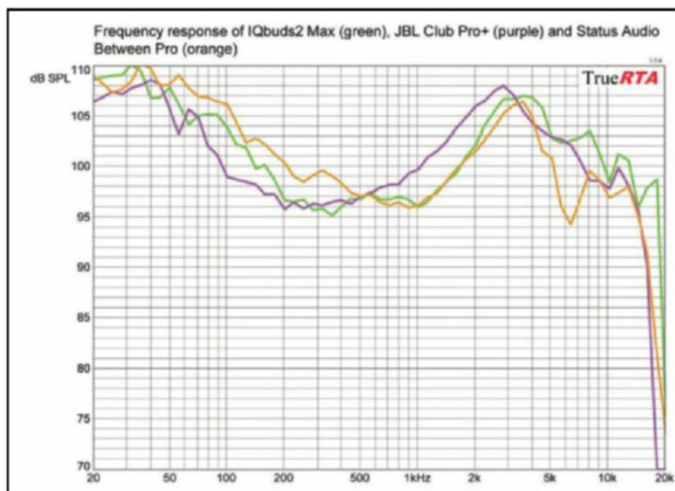


Figure 5: The 1/6th-octave-smoothed frequency response of IQbuds2 MAX vs. two conventional true wireless earphones, tested by playing pink noise through the devices under test

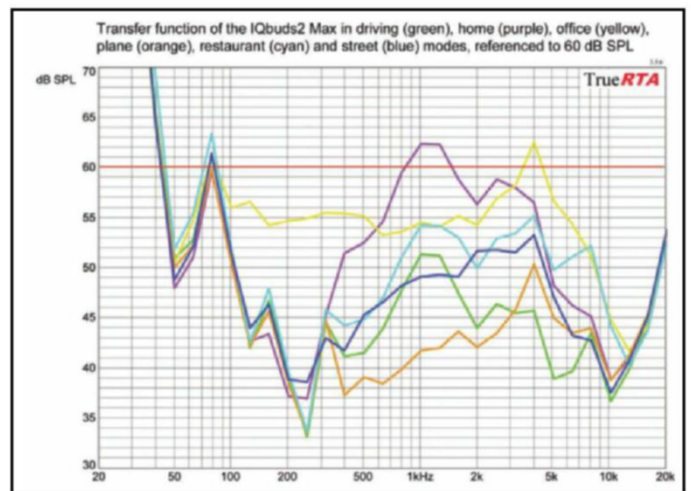


Figure 6: Effects of the IQbuds2 MAX's different situational listening modes, measured with pink noise

the same procedure I use to test noise cancelling in headphones and earphones, but at a noise level 25dB lower.

Figure 6 shows how the different hearing-assistance modes filter the sound. The 60dB baseline (corrected for the response of the speakers, room and ear/cheek simulator) is shown in red. (Note that TrueRTA displays traces in only six different colors, so I had to leave out the Workout mode.) The Office, Plane, and Street modes emphasize 4kHz (presumably to improve speech recognition), while the home and driving modes emphasize 1kHz (for reasons I can't guess). Restaurant mode is more of a broadband boost between 1kHz and 5kHz. Of course, most of these modes are just an estimation of what kind of emphasis you might need in these environments, because from an ambient noise standpoint, offices, restaurants, and streets, for example, can differ quite a bit. But at least the user has a variety of emphasis profiles easily available.

I also measured the effects of the noise cancelling feature, which you can see in **Figure 7**. I measured these curves the same way I did for the previously mentioned modes, but at my usual baseline of 85dB pink noise for noise-cancelling measurements. I used the switch on the screen from Figure 1, flicking among the ANC on, ANC off, and World modes (with World EQ centered). The noise cancelling is fairly effective, reducing low-frequency sounds (where most jet airliner cabin noise resides) by about -20dB. The World mode, with the EQ centered, admits a broad band of sound between about 700Hz and 9kHz, which should be plenty to permit good speech comprehension.

Measuring the effect of the Focus control was complicated for the reasons cited earlier. I employed much the same method I did when testing the directional function of the Widex Moment hearing aids, with the earphones positioned between one Audioengine HD3 speaker in front and another one in back. But I had to sit between the speakers and hold the ear simulator up to my ear. Regardless, it did seem to work. I ran measurements of the differences in response from front and rear with the Focus mode turned on and off, subtracted the rear response from the front response in both cases, and then subtracted the difference

with Focus on from the difference with Focus off. You can see the results in **Figure 8**. Focus mode does seem to reduce the response from the rear by -1dB to -5dB in a band between 1kHz and 4kHz, which is where the strongest directional cues occur. This mirrored my results when using the IQbuds2 MAX. I found that Focus mode had an audible but mild effect, probably enough to make a modest difference when conversing in a loud restaurant.

Subjective Testing

As I wrote this, the coronavirus pandemic was starting to wind down, but I wasn't yet traveling or interacting with people face-to-face on a regular basis. However, I was able to use the IQbuds2 MAX when shopping, walking the dog, and for music listening and TV watching. The app made it easy for me to try evaluating the sound in different environments and hear the effects on speech and ambient sounds.

Despite the bulky design, I found the IQbuds2 MAX earphones easy to fit into my ears, and comfortable to wear. It helps that they come with six different sets of eartips: three sizes each in foam and silicone (**Photo 5**).

With the Ear ID correction activated, I heard a slight high-frequency boost that, to my ears, improved the sound of music and podcasts just a bit. The earphones still sounded good without Ear ID, although I'd describe their native, non-Ear ID tuning as slightly on the soft side, with a somewhat mellow treble. I did most of my listening with Ear ID on, which is a good sign—I've tried some hearing test and augmentation apps that made the sound excessively shrill and unenjoyable. Overall, I'd say that with Ear ID on, the IQbuds2 MAX are competitive with typical good-quality earphones priced in the low three figures.

While I didn't have access to places that would be suitable for testing the office, restaurant, and plane sound enhancement modes, I was able to try the various modes in my wanderings around Los Angeles, CA—from going to the grocery store and the music store, to walking my dog, to chatting with socially distanced neighbors at the park. Being able to preset four modes

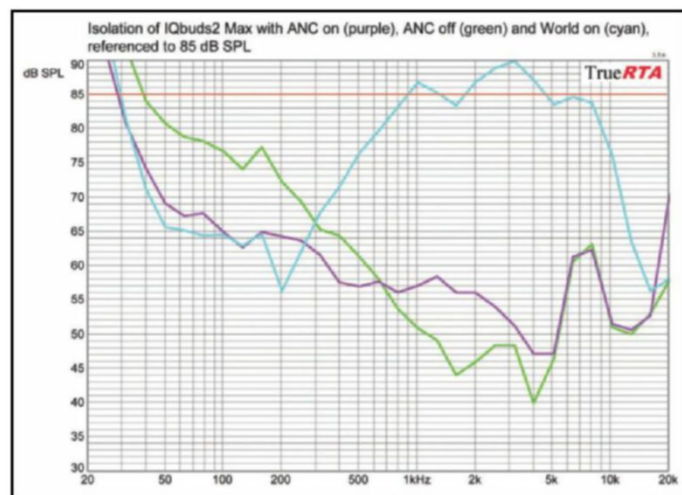


Figure 7: Effect of the noise cancelling and World modes of the IQbuds2 MAX, with a baseline of 85dB SPL. The lower the trace on the chart, the greater the isolation

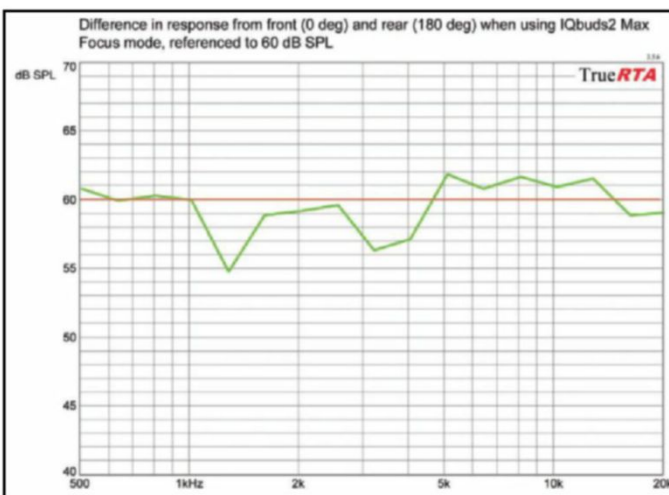


Figure 8: Difference in response from front to rear when using the Focus mode in the IQbuds2 MAX



Photo 5: The complete Nuheara package and its accessories

for quick access was a big help. I often started out my journeys fussing with the controls in the app, but lost interest because usually I could get the response I wanted simply by selecting the most suitable mode by tapping the right-side earpiece.

This is an important point—many of the people within the demographics who most often experience hearing loss are not inclined to open a smartphone app to do much of anything, much less fuss with a bunch of settings for their earphones. The way the IQbuds2 MAX setup works, a tech-savvy relative can help the wearer set up the earphones, and all the wearer really needs to do after that is touch the right-side earpiece to access the different modes.

I found that initially the enhancement modes could sound unnatural because they tended to amplify and exaggerate certain sounds—for example, walking through the parking lot to the grocery store, I was bothered by a “whooshing” sound I’d never noticed before, which I later realized was road noise from the nearby freeway that my brain had long ago learned to tune out from my non-augmented ears.

When I went into the store, the whooshing disappeared, but I had heightened awareness of all the little sounds around me, such as people crinkling plastic bags, grocery cart wheels squeaking, and all the little bleeps and bloops that the various machines in the store emit. But after about 10 minutes of shopping, I realized I’d completely forgotten I was wearing the earphones.

The IQbuds2 MAX did a great job of amplifying not only voices, but alert sounds. The best example I can cite is that when in Driving mode, the IQbuds2 MAX seemed to make the clicking sound of the turn signal about 6dB louder, to the point where there was no way I could forget the signal was on. If the IQbuds2 MAX can deter even just a few drivers from obviously leaving their blinker on as they drive down the road, Nuheara will have made the world a better place.

The only time the IQbuds2 MAX disappointed me was when I kept hearing a distorted, crackling sound in my left ear as I was driving. I thought perhaps a feedback or feed-forward circuit in the earphones’ active noise cancelling was being driven into clipping, as often happens in a jet airliner during takeoff when the engines are running at full rated power and cabin noise is at its max. But I later realized the problem was wind noise caused by the air current from my car’s ventilation system. If possible, it’d be nice to add a wind noise cancellation algorithm to these earphones, like the algorithm used in my Jabra Elite Sport 65t earphones.

Conclusion

According to the World Health Organization, hearing loss sufferers are expected to nearly double in number by 2050. Yet the average \$2,300 per ear that one has to spend on traditional hearing aids purchased through an audiologist is clearly out of reach for many people. While people with severe hearing loss obviously need to consult health professionals for solutions tailored to their problem, people who just have difficulty understanding conversation in noisy restaurants would benefit from a less costly solution like the IQbuds2 MAX.

From my experience with the IQbuds2 MAX, I think it has the power to provide a significant and welcome improvement in speech comprehension, and it’s a good-sounding set of noise-cancelling earphones to boot. People who have a mild but troublesome degree of hearing loss can pick these up (at a discount) from online merchants, spend half an hour figuring out how they work and taking the Ear ID test, and potentially get a significant quality-of-life improvement. Even if they don’t, they’ll still have a nice set of noise-cancelling earphones, at only a small price premium compared to what they’d pay for noise-cancelling earphones from the big consumer brands.

I’ve already started telling my friends who have hearing loss about the IQbuds2 MAX; they’re all intrigued, and I expect several will buy a set. Clearly, this is a product that fills an all-too-common need.

About the Author

Brent Butterworth has worked as an audio journalist and technical consultant since 1989. He currently writes for *Wirecutter*, *SoundStage*, and *JazzTimes*, and has published speaker, headphone, and amplifier measurements in *audioXpress*, *Sound & Vision*, *Home Theater*, *Home Theater Review*, and many other publications. He has also served as a consultant on measurement, design, and tuning of audio products, with a special focus on active speakers and DSP tuning.



DESIGNATED DRIVER

Leading Pro Audio brands choose Celestion

Innovation, performance and reliability make Celestion the drivers of choice for a growing number of big-name PA brands. So when you need superior quality compression drivers or professional loudspeakers, check out about the speakers the pros use.



celestion.com

CELESTION

NAD Electronics M33 Masters BluOS Streaming DAC Amplifier

The Complete Package for a Wireless Life



Photo 1: The NAD M33 streaming amplifier is a one-package solution integrating streaming, analog and digital inputs, DSP, and power electronics.

By
Stuart Yaniger

Photos by Cynthia Wenslow

After reviewing the compact but mighty M10 Masters Series streaming amplifier from NAD Electronics, Stuart Yaniger simply had to try the new flagship NAD M33 Masters streaming DAC amplifier. This was the first mainstream product to implement the Purifi Audio Eigentakt (self-clocking) amplifier modules, but there is a whole lot more to make it a very complete package for a musical life. The article reviews the experience and performance measurements.

NAD is one of the longtime companies that has kept a consistent identity over the five decades of its existence and three successive owners—the brand has an honorable history of no-frills gear offering high performance and value to the broad middle price segment of the audio market.

Interestingly, NAD's history actually traces back to the purchase of the storied loudspeaker company Acoustic Research (AR) by the large conglomerate Teledyne in the late 1960s. Teledyne was surprised that AR had been selling electronics as well as loudspeakers, and was only interested in the speakers. So New Acoustic Dimension, originally formed by its founder Dr. Martin Borish as a distribution company, purchased the designs, tooling, and supply chain that AR had been using, and brought in its principal electronics designer, Bjørn Erik Edvardsen. Clearly, the concept of high performance at moderate price, AR's claim to fame, was built into the company's DNA.

The NAD 3020 integrated amplifier, introduced in the late 1970s, sold millions of units and was the centerpiece of several of the sound systems I used back in those days. The secret to achieving this value vs. performance was a very forward-thinking business model, with local marketing and engineering and outsourced manufacturing. This is common practice today because of the vastly decreased capital expenditures and operating costs, but was a rarity 40 years ago, and a great challenge because getting reliable contract manufacturers was far more hit and miss than today.

After several ownership changes over the years, the contemporary NAD was acquired by Lenbrook, a family-owned Canadian company, which has accumulated a synergistic trio of audio companies: PSB loudspeakers, BlueSound audio software, and NAD. The combination of companies enables users to have a fully integrated Lenbrook system, with a rich feature set of control capabilities.

NAD's flagship products are called the Masters Series, and the latest offering is its top-of-the-line M33. Recently, I had the chance to spend a few months using this remarkable product, the NAD Masters Series M33 BluOS Streaming DAC Amplifier.

The Product Concept

The M33 is designed as an actual all-in-one device, combining the functionality of a high-powered (200W into 8Ω, 380W into 4Ω) high-performance integrated amplifier having analog and digital I/O with a network streaming system to accommodate both modern and legacy sources. It also contains an electronic crossover for biamping. The superb room acoustics correction software, Dirac LE, is built in. Only speakers are required to complete a versatile audio system.

BlueSound's proprietary operating system, BluOS, lies at the heart of the M33 system package. I described the features of this interface in my review of the NAD M10 (see Resources), but to summarize, it works through the home network, either via wired LAN or Wi-Fi, and allows control of multiple systems in a home from a single (or multiple, if desired) wireless control point. The BluOS app (available for Mac, PC, Android, and iOS) can be run from pads, phones, laptops, or even workstations. It's basically an audio "Swiss Army knife," enabling the user to select the source (including streaming and music on network drives), select the room, select the hardware, and direct the music to whichever device and room is chosen. A nifty feature is "Follow Me," the ability to move a playlist from one room and one device in the BlueOS ecosystem to the next as you move about the house, requiring only a tap on a virtual button.

Besides source and destination control, BluOS also enables control of basic audio functions. Although this can also be handled by dedicated remote controls (either the one provided with the M33 or programmable universal controls), many users, myself included, prefer using a smartphone for control since it's handy, easy to use, and integrates the audio settings with music selection.

In my review of the M33's smaller sibling (the M10, see Resources), I lamented that the term "lifestyle product" has developed a pejorative connotation. And in this case, an undeserved one—if you're a serious music listener, having a single package with a high level of versatility and performance coupled with ease of use is a major positive. This may be less important to someone for whom music is a necessary evil for evaluating sound systems, but perfect for the music-first listener who doesn't want to fool around with equipment



Photo 2: The M33's rear panel shows the array of input and output options—plus a headphone jack on the front panel. On the left side are two Modular Design Construction (MDC) slots that allow for further expansion. Recently, NAD introduced an MDC module that adds support for DSD256 music playback and streaming over USB.

but wants good sound. Besides working as a stand-alone app, BluOS integrates with home automation systems (e.g., Lutron, Crestron, Control 4, and several other popular systems).

Although it can be used as a conventional integrated amplifier, the M33 is fundamentally a network device. It can stream from Internet services (e.g., Amazon HD, Tidal, Qobuz, etc.) and from any NAS music libraries on the same network to which it's connected. And because it's a network device, just as with your computer, the M33 (and other BluOS devices) can get firmware updates automatically. NAD has been using this capability to fix bugs that weren't caught during development but caused issues for some users. For example, in my M10 review, I noted that the digital inputs had periodic glitching, making measurement difficult and occasionally being severe enough to be audible. A firmware update a few months ago fixed this issue without me having to do any work or having to send the unit to a service center. Keep in mind that many of the small issues I discovered during this review process may already have been fixed via the automatic updating.

Besides bug repair, the updates have also added new capabilities to the M33. The most recent rounds (as of the writing of this article) include the addition of Radio Paradise and Sirius XM to the streaming options. It is now also certified for Roon. For music played via a NAS drive, the M33 can handle just about all of the alphabet soup of formats: FLAC, AIFF, WAV, MQA, OGG, WMA, ALAC, OPUS, and probably a few more I missed. The M33 can also be controlled via voice commands using Alexa, Siri, and Google Assistant.

I mentioned Dirac room acoustics correction; the under-500 Hz version is included. It may be upgraded to the full-range package for an additional \$99. Many feel that the basic package is sufficient, but in my view, the small marginal cost for the full version is more than worth it. I have a much more detailed explanation of Dirac's operation and

details about setup in my review of the NAD M10 (see Resources). The basic version does support Dirac's Bass Control for driving multiple woofers independently.

Finally, it should be mentioned that the power amplification here is a pair of Class-D modules using Eigentakt technology (see Resources) licensed from Purifi. My previous experience with Purifi Eigentakt confirmed that its amps had high power, ultra-low distortion and noise, low source impedance, and stability into every load I threw at it.

The M33 in Execution

The M33 is a moderately large but relatively light (22lb/10kg) unit with a front panel featuring a single large control knob and a black Gorilla glass LCD touch screen showing the user's choice of input sources, album art, or a pair of large virtual power meters (**Photo 1**). The casework is an attractive and easy-to-maintain brushed metal, with vents on the top. The bottom has four hard metal cones as feet. If you're worried about damaging furniture, four magnetic pucks with depressions in the center are provided to attach to the points of the cones.

The rear panel is where all the action is. Where the M33 fundamentally distinguishes itself from the M10 is greater input versatility (**Photo 2**). Besides the usual line level analog inputs, the M33 has balanced analog I/O, HDMI, two optical and two coaxial SPDIF inputs, balanced AES/EBU, and in a nod to fashionable anachronism, phono inputs. All analog inputs are digitized before being fed to the M33's DSP and power amp sections.

There's a dedicated headphone output, a feature lacking in the M10; Bluetooth headphone capability is still provided, however. The user can also send music to the M33 via Bluetooth; the M33 is equipped with aptX-HD. Similarly, Apple's Airplay 2 can be used. There's also two expansion slots, dubbed Modular Design Construction (MDC) to add in any future type of I/O that might become popular. The star of the show, suggested by the Wi-Fi antennas and the Ethernet connector, is certainly the network streaming capability.

A set of unbalanced preamp outputs are also provided so that an external power amp can be used, just in case the power of the M33 is insufficient. Unfortunately, it lacks the preamp to power amp jumpers that were useful for the old 3020 integrated so that the power amps can also be used in isolation (e.g., for driving a subwoofer while an external amp drives the main speakers). I hope that in a fit of nostalgia, NAD brings back that useful feature.

The M33 is packed in a manner that is functional but not over the top. Besides the main padded box and a slick cloth bag protecting the M33 from scuffing, there are two sub-boxes containing the power cords, Wi-Fi antennas, the remote control, a measurement mic and adapter for Dirac, and a USB thumb drive with manuals and other documentation (**Photo 3**). The remote control is remarkably solid and heavy, so much so that it could be a pretty effective weapon if thrown; no cheap plastic casing here! Adding to the impression of luxury, the USB drive is brushed metal and snapped into a leather scabbard with an embossed NAD logo.

The moving magnet phono stage is bipolar op-amp based (NAD demurred on disclosure of the device used, but claims it is "known for its warm musical sound"). In moving magnet mode, it has 40dB of gain at 1kHz and a 47k Ω input impedance. In the context of the M33's overall design, I was surprised to find that the RIAA is not done digitally after a flat gain block, but rather using a classic feedback EQ.

According to Chief Technical Officer Greg Stidsen, this results in more headroom before overload. NAD says that it uses close tolerance, premium parts (polystyrene caps, 1% resistors) for the EQ. One interesting design twist is the use of a parallel feedback loop that is frequency selective to implement the infrasonic filter, with a 10Hz inflection point and 24dB/octave slope. This provides protection against infrasonic noise (mechanical/acoustic feedback), which can drive the amplifier into clipping and speakers into excursion distortion at fairly low musical playback levels, but has much less impact on deep bass from music than, say,



Photo 3: The supplied accessories include a heavy duty remote control, a measurement mic, and a fancy USB thumb drive with product documentation.

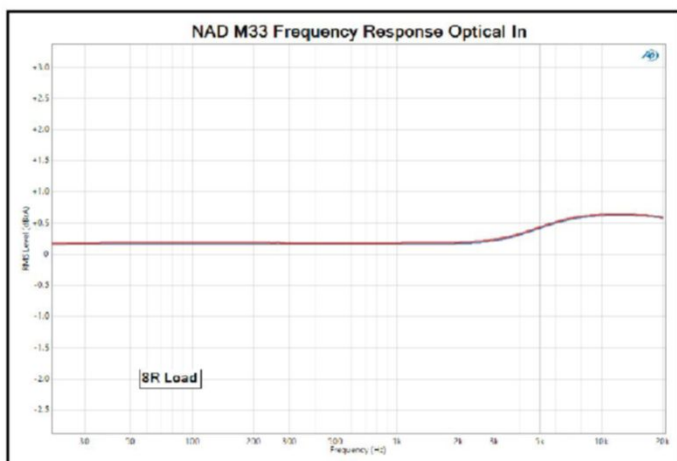


Figure 1: The initial measured frequency response showed a small treble lift. This traced to a tone control setting bug.

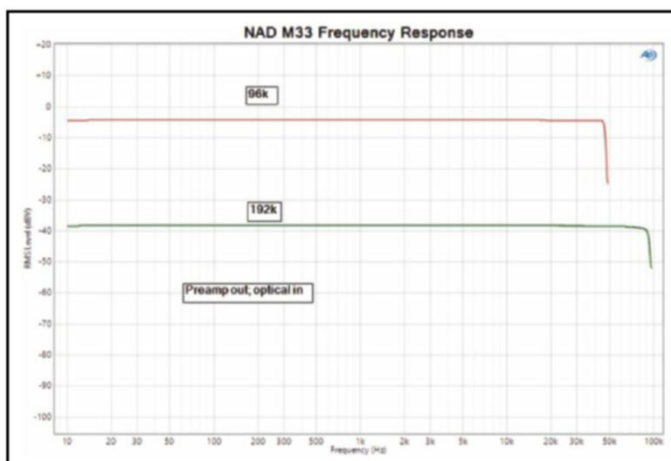


Figure 2: The frequency response at the preamp outputs is nearly perfectly flat.

using the IEC amendment in the RIAA equalization. When switched into moving coil mode, the input uses a discrete bipolar transistor flat gain stage with 20dB of gain and has a 100Ω input impedance.

The M33 in Use

The electrical setup of the M33 was simple, with balanced analog, unbalanced analog, and digital sources connected in seconds. The binding posts used for the speaker outputs are the same ones used across Lenbrook's top-of-the-line products, and were easy and convenient to use for hooking up my speakers. They accommodate pretty hefty wires and spade lugs, as well as the usual banana and pin plugs, and are easily tightened by hand using a molded tab on the thumb nut for leverage. I really like that aspect of the ergonomics.

With all my positive anticipation at getting this beauty fired up and making music, the part I dreaded had to come first: setting it up on our wireless network. When I originally received the M10, getting it working via wireless was a major adventure, so much so that I recommended (and still recommend) purchasing from a dealer who agrees to set it up and get it running.

I was very pleasantly surprised to have the M33 setup go almost automatically. Once I had it added to the BluOS app on my phone, it immediately connected to our Wi-Fi and could stream via our music services or our network drive. Slick and smooth, until...

When it comes to networking, nothing ever seems straightforward here. I always wonder if our setup should be used for worst-case testing. In any case, I moved the M10 to another room during the M33 testing with the idea of using the BluOS capability via Wi-Fi to control both. Unfortunately, adding the M33 caused the M10 to disappear from the network. This started me on a multi-week process with NAD's technical support, and the final answer involved doing factory resets (considerably

more work than a reboot) and setting the M10 up on an Ethernet connection before trying to connect it via Wi-Fi.

Once things were working, switching between players and rooms was very smooth and easy. That said, about once a week, the M33 would refuse to connect to the network. A relatively quick reboot would restore things, but still, this was a bit annoying. I will repeat my earlier advice: buy the M33 from a dealer willing to do setup and basic support.

We are a world leading manufacturer of high performance transformers for audio applications.
Let us help you improve your sound.

"IF YOU CAN HEAR IT,
IT'S NOT OURS"



LUNDAHL

— TRANSFORMERS —

www.lundahltransformers.com
office@lundahltransformers.com

I also felt that the sound seemed just a tiny bit forward and bright compared to the M10, but thought I'd check some measurements before trying to validate my impression with an ABX test. The first thing I checked was frequency response, and got a bit of a surprise (**Figure 1**); the top three octaves were boosted by about 0.5dB. I checked the tone controls in BlueOS and the button was set to off. Turning it on, the treble slider was up a notch or two, which I thought shouldn't matter if the tone controls are supposedly off. But just for fun, I set the treble control for flat before turning the tone controls off again. Lo and behold, the frequency response was now flat. A minor bug, but be careful that you don't repeat my mistake.

Since I already had a license for the full-frequency version of Dirac, I was able to get that up and running on the M33 in seconds. Unfortunately, it would not allow me to use the project files from my previous measurements, so the Dirac measurements had to be repeated. I've done them enough times and had left in place the floor and mic stand markings that I used for the M10 review so I was through the measurements in minutes. A very minor inconvenience.

Once everything was set up and Dirac was in play, all the minor annoyances faded away—the sound was absolutely gorgeous. I expected the M33 to sound pretty much like the M10, and that was indeed the case, as far as it went. By that last phrase, I mean that on particularly challenging material, I could squeeze

a bit more volume out of the M33 because of the higher available power from the Eigentakt modules.

For example, I have recently been bingeing on Dream Theater, and at the appropriate volume using the M10, I start hearing a bit of graininess set in from time to time; I was particularly getting this impression when listening to "The Dance of Eternity" from their superb *Metropolis Part 2: Scenes from a Memory* album (Elektra), streamed via Amazon HD. At subjectively similar levels, the M33 sounded seamless and completely in control.

On the more delicate end, Sarah Jarosz's "Maggie" from *World On The Ground* (Rounded) really showed off Dirac's ability to separate instruments in a mix. The downside is that it's too easy to hear the clumsy reverb on Jarosz's voice, but still, the overall purity of the mix is honored. This led me to Nickel Creek's "Elsie" from *A Dotted Line* (Nonesuch) with a beautifully recorded violin and mandolin lines dancing with the flat-picked guitar. And then... well, let me just say that streaming from a service like Amazon HD has destroyed hundreds of hours of productive time for me as I run down one musical rabbit hole after another and listen to music instead of being productive. Ah, well, life in 2021.

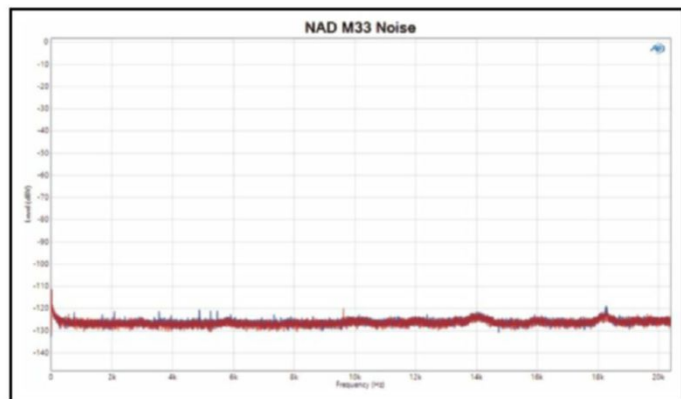


Figure 3: The noise spectrum taken at the power amp outputs confirms the impression of dead silence at the speakers.

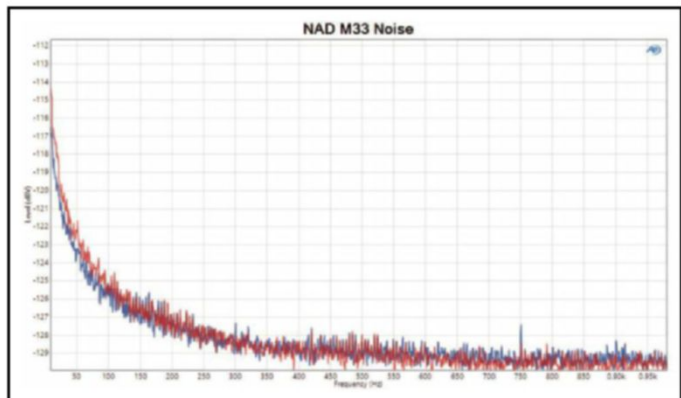


Figure 4: The noise floor has its corner frequency at about 100Hz to 150Hz.

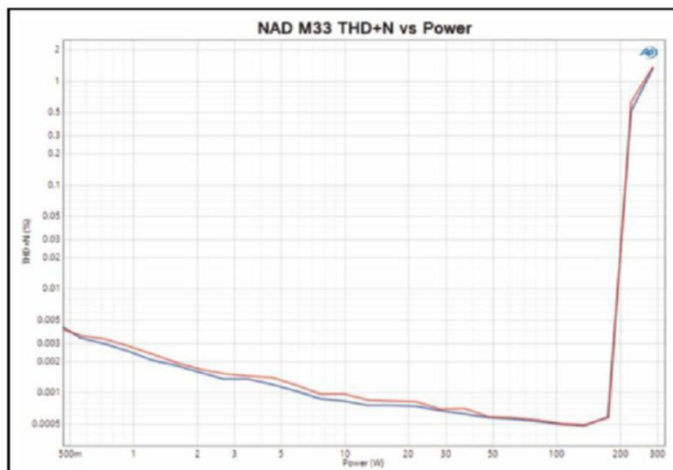


Figure 5: Measuring THD+N vs. power shows the M33 performing significantly better than specified ratings.

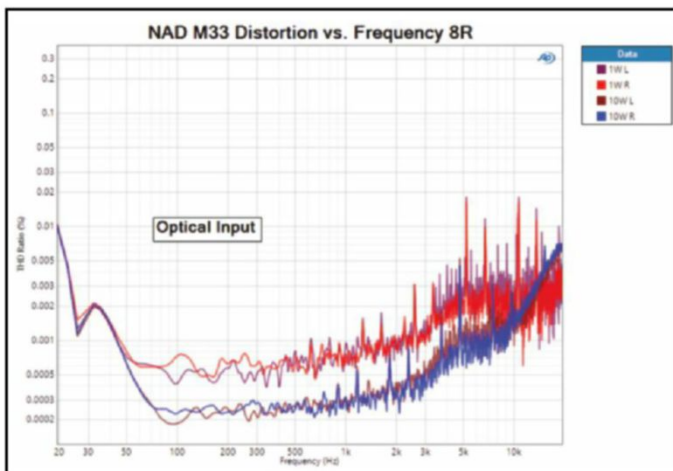


Figure 6: The THD+N vs. frequency curves at two different power levels appears to be dominated by noise.

Just for completeness, I double-checked that the MQA decoding functioned from Tidal. Well, the light went on, so I trust that there was actually decoding. And as non-auditory subjective comment, the LCD touch display was sharp and clear, and very responsive to touch. The large volume control had a smooth feel and was a tactile pleasure to use. The look and feel here are absolutely impeccable. Through my time with it, the M33 was quiet, powerful, clean, and responsive—an absolute pleasure to use every day.

The M33 on the Bench

It's always nice to have an engineered (as opposed to "designed") product on the test bench. And indeed, the M33 showed very few surprises. Because the M33s were in somewhat short supply during this review, my time with it was somewhat limited. So the measurements here focused on some basic stuff that I haven't seen elsewhere and a few things that might present an issue for users. As usual, my test setup comprises an Audio Precision APx525 audio analyzer and various homemade dummy loads for the power amplifier.

Figure 2 shows the frequency response measured at the preamp outputs at two different sampling rates. There's almost

no droop until the Nyquist cut-off. The source impedance of the amplifier was about 0.2Ω , so the frequency response will not be strongly affected by the varying impedance of loudspeaker loads.

Like its smaller sibling, the M33 was dead silent in my listening setup. **Figure 3** shows why—there's no appreciable noise spikes or hash. At lower frequencies, the noise rises slightly, as shown in **Figure 4**, with a noise corner frequency of roughly 100Hz to 150Hz.

The Eigentakt technology power amps do not disappoint. **Figure 5** shows THD+N as a function of power for a 1kHz signal driving an 8Ω load. Power exceeds 260W at the 1% distortion point, easily beating specification. **Figure 6** shows the variation of distortion plus noise with frequency at power levels of 1W and 10W into 8Ω ; it appears to be dominated by noise rather than harmonics. This is confirmed by looking at the distortion spectrum (**Figure 7**), which shows harmonic components barely larger than (and perhaps even below) the AP analyzer's residual distortion.

Increasing the power to 150W yielded the spectrum in **Figure 8**. Again, the distortion components are dominated by the AP analyzer's residuals. This is consistent with my previous measurements of Eigentakt amplification. These are remarkable works of engineering!

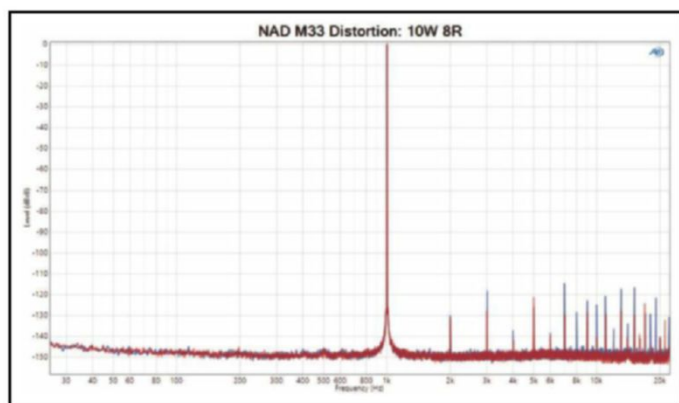


Figure 7: The distortion spectrum of the M33 driving an 8Ω load to 10W is dominated by the residual of the AP analyzer.

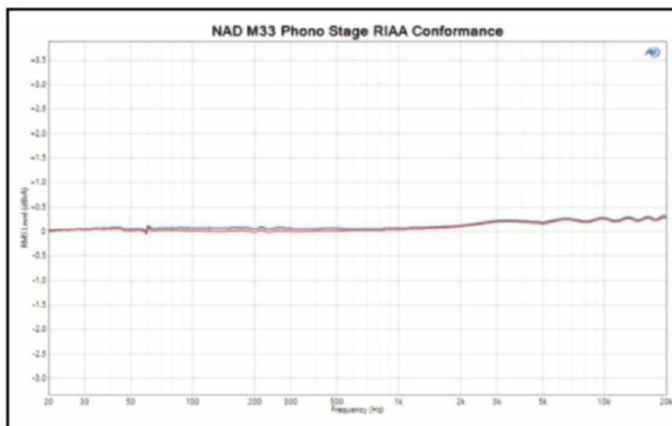


Figure 9: The RIAA conformance and channel matching in Phono mode is nearly perfect, other than a very minor rise in the top two octaves.

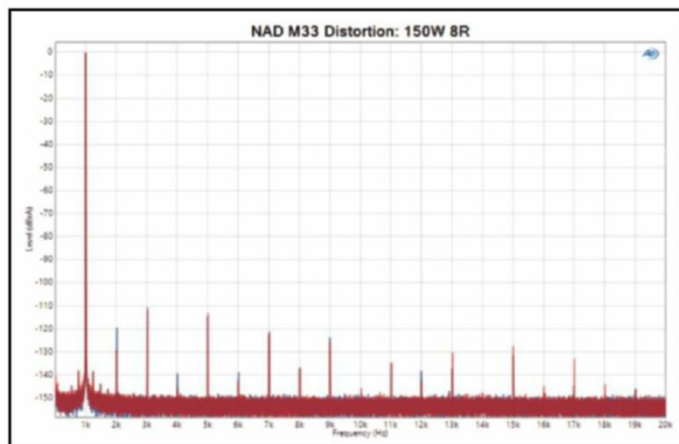


Figure 8: At 150W into 8Ω , the M33's distortion is still low enough to be difficult to measure.

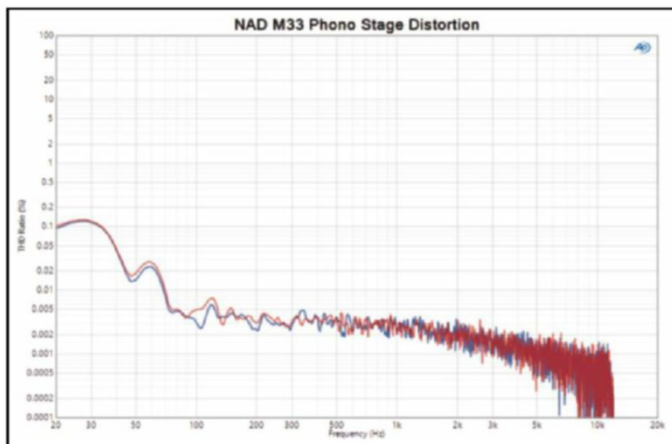


Figure 10: Phono stage distortion and noise at 500mV preamp out is extremely low and only rises a bit in the bottom octave.

In much modern equipment, phono stages are added in as afterthoughts—and the performance proves it. Is that the case here as well? **Figure 9** shows the frequency response of the phono stage when driven by an inverse-RIAA equalized signal. There's a slight lift in the top two octaves (0.2dB or so), which is negligible. Otherwise, the response is quite flat, and even more impressively, the channel-to-channel matching is nearly perfect. Likewise, the distortion of the phono stage at 500mV preamp output (**Figure 10**) is very low, rising a bit in the lowest frequencies where the gain is highest. This phono stage is definitely not an afterthought and shows excellent engineering and execution.

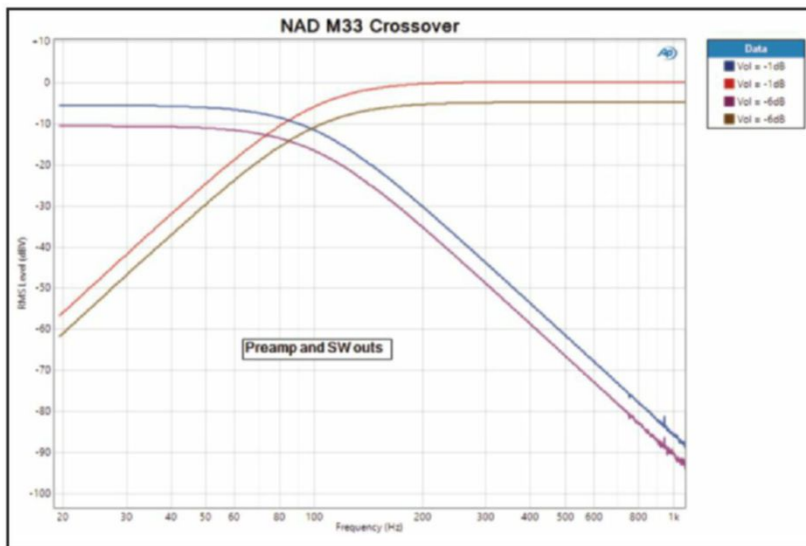
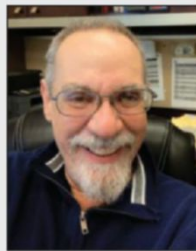


Figure 11: The built-in adjustable frequency electronic crossover has fourth-order Linkwitz-Reilly filter responses. Note the lower output of the bass section.

About the Author

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



Resources

NAD M33 Product Webpage | <https://nadelectronics.com/product/m33-bluos-streaming-dac-amplifier>

S. Yaniger, "NAD M10 Masters Series Streaming Amplifier," *audioXpress*, August 2020, <https://audioxpress.com/article/fresh-from-the-bench-nad-m10-masters-series-streaming-amplifier>

S. Yaniger, "A Tale of Two Class D Amplifiers," *audioXpress*, July 2020, <https://audioxpress.com/article/fresh-from-the-bench-a-tale-of-two-class-d-amplifiers-orchard-audio-bosc-and-purifi-audio-eigentakt-eval1>

Finally, the M33 has an adjustable frequency electronic crossover for use in systems having separate woofers or subwoofers. NAD does not specify filter responses, or if it does, it's deeply buried in its literature. **Figure 11** shows the response at subwoofer and main outputs with the crossover set at 110Hz. These conform closely to a fourth-order Linkwitz-Reilly response. As was the case for the M10, the bass output is inexplicably 6dB lower than the main output, and the relative levels cannot be adjusted. So having a separate gain control for the subwoofers is a must. The stereo bass outputs are independent of each other, which is a big advantage at higher crossover frequencies.

Conclusions

The M33 is not inexpensive at about \$5000, but the performance and feature set packed into it are quite significant! The user gets a single box that handles just about any conceivable input type, with full network and streaming capability, the muscle needed to drive nearly any speaker load; one of the top room acoustics corrections systems; the ability to play files in almost every imaginable format (including MQA); and integration into home automation systems. There's also the capability of expansion for future formats. Additionally, the user interface app is slick and intuitive, has multi-device capability, and receives regular updates, bug fixes, and new features for both the BluOS app and the M33 via the Internet.

There's still a few bugs to be squashed, but they're relatively minor and easy to work around; they could even be fixed by the time this review is in print. But overall, the M33 is the complete package—it does it all, the sort of product that ends the chase for better sound on the source and electronics end. It's had a lot of accolades and based on my experience with it in my listening room, I think it absolutely deserves them. If your main love is playing with equipment and swapping, it may take the fun out of the hobby. If your main love is music and sound, this is pretty much as good as it gets.

Great product, highly recommended.

Author Acknowledgements: I'd like to thank Jeff Touzeau of Hummingbird Media for arranging the review and diligently connecting me with the right people to answer my questions. Lenbrook's Chief Technology Officer, Greg Stidsen, was a fountain of useful information and historical trivia. And Bob Moran, NAD's stalwart tech support manager, patiently helped me through my various networking issues.

**PURE
SOUND**



PTT6.5M **NEW!**

6.5" Ultra Low Distortion Midrange Driver

- ⊙ Extreme detail retrieval
- ⊙ Increased efficiency

PTT6.5W & PTT6.5X

6.5" Ultra Low Distortion Woofer & Extended Woofer

- ⊙ Powerful, precise bass with lifelike and uncluttered mids
- ⊙ Get "beyond 3-way" performance from a simple 2-way system

PTT4.0X

4" Ultra Low Distortion Extended Woofer

- ⊙ Uncompromised performance from compact designs
- ⊙ Matching Passive Radiators w/ Purifi Neutral Surround

1ET400A & EVAL kits

Ultra High Performance Class D Amplifier

- ⊙ Unrivalled measured and sonic performance
- ⊙ Outperforms amplifiers of any operating class
- ⊙ Industrial robustness



The Multiple Faces of Amplification Evolution

By
J. Martins
(Editor-in-Chief)



The NAD electronics C 298 two-channel amplifier combined with the C 658 BluOS Streaming DAC

Addressing audio amplification trends and perspectives, this article highlights products from different application segments to illustrate the state of the industry. From new networked amplifiers to new high-performance technologies and high-volume applications, amplification technologies are at the core of every audio product and system design.

With the current audio amplification market dominated by Class-D products, it's important to have a perspective of what exactly the product segment represents when analyzed from that technology's perspective. According to the 2020 Class-D Amplifier Market report from market research firm Futuresource Consulting, revenues surpassed \$1.8 billion USD in 2019, hitting 2.9 billion units shipped, before the effects of the global pandemic were felt. Numbers that should be hardly matched in 2021, when growth is expected to resume. Important to note that the low power category (<5W) in audio amplification devices represents 45% of revenues and 81% of total volumes and is expected to grow the highest in the category, at 8.4% CAGR in value by 2024. Battery-operated personal technology drives this category including mobile, wearable, and hearable devices. Mid-power devices (5W to 50W) represent 17% of units shipped, and are used in home audio, TV, and automotive applications, being the smallest category by revenue. High power systems (50W to 150W) represented the third largest volume share with only 1.8% of units shipped but

representing the second largest revenue share with 21.1%, due to automotive applications. Finally, the ultra-high power segment (>150W) has a negligible share by volume but stands in third position as the largest share by revenue (16%) due to high value professional audio products and markets.

Overall, as Futuresource notes in its category report, the market is demanding lower power levels since vendors are now able to optimize speaker configurations beyond power output and volume, while in the highest segment per volume, power efficiency in battery-operated products is the main priority for manufacturers and consumers.

Estimated to represent an overall worldwide business totalling approximately \$800 million USD, also according to Futuresource, professional power amplifiers still use a combination of topologies, including high volumes of Class-AB amplifiers, even though Class-D is quickly becoming the dominant technology of choice allowing for smaller and lighter systems.

It is important to note that professional audio amplifiers today are also sophisticated, multifunction

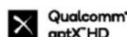


INTRODUCING THE NAD MASTERS M33

It's simply a matter of time.

The NAD Masters M33 is a combination of visionary ideas and decades of audio research, featuring the innovative Eigentakt™ amplification technology, which virtually eliminates audio distortion by correcting imprecise oscillation timings to deliver over 200 watts of rich, studio-quality sound. Add BluOSTM for pioneering hi-res multi-room functionality, a powerful NXP 1 GHz ARM Processor for bit-perfect audio processing, and a 32-bit ESS Sabre Hyperstream™ DAC for the highest accuracy reproduction possible, and it is clear that the NAD M33 is an uncompromising and defyingly simple “just add speakers” approach to high fidelity audio.

Discover more at: nadelectronics.com/m33





High-volume Class-D audio amplifier devices are available to the overall audio industry with increasingly high quality levels, at very affordable prices. Pictured is a Dayton Audio KAB Class D amplifier board, available in 60W, and 100W versions, powered by a highly efficient Texas Instrument Class-D chipset.

systems that have evolved from standard two-channel, single-function units, into multichannel and multi-configurable amplifiers able to easily deal with asymmetric loads, supporting high-voltage and high-current operation, and the most demanding bridge mono modes. At the same time, audio amplifiers evolved into much more complex systems, accommodating audio switching, signal processing, signal conversion, and multiple connectivity with remote control options, including real-time diagnostics and monitoring of the complete feature set. And following the introduction of audio networking integration and distributed signal processing, amplifiers have now evolved into fully connected systems, powered by remote web- and cloud-based tools.

Installed commercial applications have remained the largest value share of the market, followed by installed leisure, touring and

rental, and studio and broadcast applications. Together, installed commercial and installed leisure represent the highest growth areas for power amplifiers.

During 2020 and the first half of 2021, touring and live performances in large arenas and auditoriums were cancelled. For the audio industry, the need for ever more powerful racks of audio amplifiers was no longer a priority. And yet the leading amplification manufacturers in the segment had in fact released flagship products just prior to the global pandemic shutting down all large in-person gatherings. Many of the new designs targeted high-power, multichannel configurations, now required by an increasing number of immersive/spatial audio applications, which are starting to be reinstated as live music and events resume activity levels.

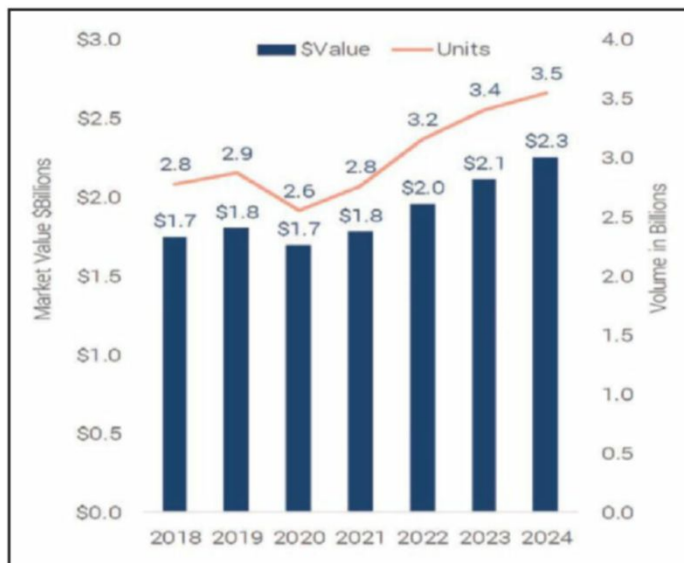
Time to Rebuild

As *audioXpress* noted in its Market Update on Audio Amplifiers published in 2019, pro audio companies had initiated a critical transition toward networked amplification solutions, and more flexible professional installation solutions with integrated digital signal processing (DSP), enabling completely new audio system designs. A transition that was critical as multiple application segments in the audio industry were seeing exponential growth in rental, portable sound, leisure, commercial, and even cinema sound applications, with the installation segment in general showing the biggest expansion in investments.

All that abruptly changed in 2020, but manufacturers didn't significantly alter their strategies or their technology road maps, knowing that the enforced pause would offer an extremely valuable opportunity for fully embracing the transition that was already taking place and increase the pace of technology updates. In audio electronics and specifically audio amplification, even as business volumes suffered, research and development never stopped and often even accelerated.

It is also important to note that the technology gap that was already specifically felt between the leading brands in pro audio amplification and the core volume manufacturers, namely from Asia, widened even more. When observing the offerings from QSC, Powersoft, Bose, and Harman's Crown, among many others, we see clear signs of a major transition. Particularly given that speaker manufacturers are increasingly building speakers with built-in amplifiers, which naturally are also part of a trend toward networked systems. In installed systems, speakers are now even being powered over the network, with standard 70V/100V systems evolving into Power-over-Ethernet (PoE) distributed systems that have the clear advantage of being monitored and adjusted in real time over the network.

The last 18 months also enabled manufacturers time to expand their network connectivity solutions. No doubt, the most visible effect is the massive number of new amplifiers with Dante connectivity, and an apparent pause in (publicly revealed) efforts in other audio networking technologies and protocols. Of course, we are certain that the investments in R&D from major players in the live performance and installation segments, from companies such as d&b audiotechnik, L-Acoustics, and Meyer Sound have continued, perfecting their own proprietary AVB-based systems. We just didn't



Worldwide Class-D amplifier market 2018-2024 (Futuresource Consulting)

see any visible results of those activities, including in the much propelled interoperability efforts with the Milan protocol, while Dante-based audio systems expanded their already massively dominant position as the de facto industry standard.

The global effects of the pandemic also offered an important validation for remote management systems and cloud-based tools, as a distributed workforce demonstrated the requirement to securely monitor and manage AV infrastructures remotely.

Strategies already in place on that front from companies such as QSC, Biamp, and Powersoft, have accelerated. At the same time, incumbent new player LEA Professional validated the company's vision for cloud-based, connected audio amplifiers. Since 2020, audio amplifiers have moved from the core of locally networked audio systems to fully distributed and remotely managed resources that can be accessed and configured over the Internet. LEA's products were designed to work like that from the company's recent inception. Others simply accelerated the networking concept and software tools to accommodate more remote management options.

What basically differentiates the approach from multiple manufacturers now is the combination between network interfaces and audio routing, processing modules, and amplification channels—some fully modular, some fully integrated in the power modules, some more firmware-based, others more software-oriented. Some manufacturers are centralizing all the functionalities within the amplifiers, others are placing audio amplification over the network and closer to the endpoints. In professional audio, amplification has now reached new levels of modularity and flexibility. The next predictable step will be adopting amplification topologies to match the new applications architectures, at the lowest and the highest power levels, with the highest levels of efficiency.

Power Stage Evolution

As professional audio applications continue to rely and expand on established Class-D platforms for the residential integration and home audio markets—where the focus on audio quality performance allows for a different set of requirements—there's been an important transition toward new better sounding Class-D audio systems, creating an opportunity for new players. *audioXpress* has previously illustrated those efforts from companies such as Purifi Audio (this edition shows a few examples of commercially available products based on Purifi's Eigentakt amplifier design,) and more recently we have also witnessed the release of the first commercially available product that uses technology from Axign, a new Dutch company founded in 2014 by former Philips and NXP employees who worked on the pioneering Class-D efforts (see Harman Kardon). Axign introduced the AX5689, a Class-D audio amplifier controller chip, implementing a digital control loop with feedback behind the output filter that suppresses all artifacts and provides dramatically improved audio quality. The Axign devices entered production in 2020 and they appeared for the first time in actual products from Harman International (see Harman Kardon). While Purifi's modules are being leveraged for high-end audio applications, the Axign AX5689 chips will enable new high sound quality mainstream designs in mid-range and lower-end systems, such as soundbars, connected streaming and smart speakers, and multichannel AV receivers and TVs.



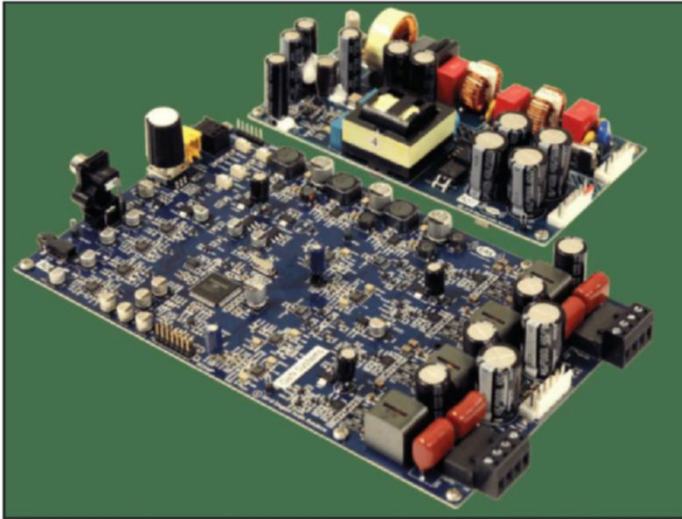
LEA Professional Connect Series Amplifiers offer remote control and monitoring of installed systems from anywhere using Amazon's AWS IoT Core as infrastructure to safely and securely transmit data to and from the amplifier.



Powersoft Mezzo, compact, Class-D amplifiers appear to be simple black boxes from the outside. Eight models offer different power and connectivity options, including Dante and AES67, and all are self-configuring, networkable systems, featuring automatic setup for audio parameters. All the units offer load monitoring, with the larger 600W models being able to deliver 4x 150W or 2x 300W, and can drive two 16Ω as well as 25V, 70V, or 100V.



GaN Systems is a specialized manufacturer of GaN (gallium nitride) power semiconductors already with a significant reach in Class-D Audio solutions, receiving very positive reactions, and ready to power new high-performance and affordable products.

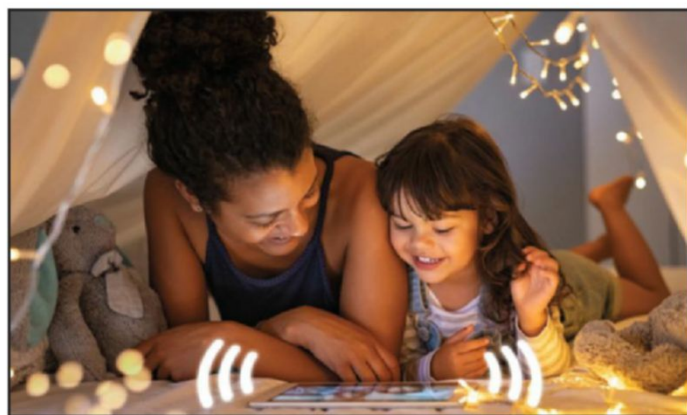


This GaN Systems amplifier evaluation kit includes a two-channel, 200W per channel at 8Ω (300W at 4Ω) Class-D audio amplifier and companion 400W, continuous power audio-grade switched-mode power supply (SMPS). The solution highlights an easy plug-and-play design with features such as multi-audio signal inputs, bridge-tied load output, and open-loop/closed-loop toggling. And, with GaN, the design becomes very efficient and operates without heatsinks.

With even more ambitious goals in terms of market reach, the latest trend in audio amplification sees the use of gallium nitride (GaN) technology in new Class-D audio amplifiers. Following pioneering efforts from Panasonic and Infineon, GaN is now appearing in a significant number of products from established and completely new brands with device solutions from Efficient Power Conversion (EPC), pSemi (Murata), Analog Devices, and GaN Systems.

Main volume uses for GaN transistors are currently in consumer fast chargers for mobile devices, and industrial power supplies, but there is a likely chance we will see GaN-based designs also take over the automotive audio market and high-volume mobile applications.

What is striking about the new GaN transistors is that they



Cirrus Logic's CS35L45 boosted amp offers a new level of smartphone audio performance for consumers, who are increasingly using their smartphone and tablet speakers.

are already taking high-end audio applications by storm, having become available in amplifier modules for audio enthusiasts and the DIY communities. And GaN-based Class-D audio solutions will enable a new generation of home entertainment and smart audio applications, starting with new Wi-Fi and Bluetooth-enabled speaker designs offering high-fidelity levels and support for high-resolution audio and spatial audio formats and sources.

GaN Systems is a leading manufacturer in this space and it offers a Class-D amplifier evaluation kit including a two-channel, 200W per channel (8Ω) board and companion audio-grade SMPS. As detailed by *audioXpress* in previous articles, and illustrated with some recent products highlighted in this market update (see Orchard Audio), GaN Systems stands out for offering some of the most efficient GaN transistors supporting a new generation of amplifier designs with improved audio quality and product performance. GaN transistors can operate at a much higher frequency that provides fast, clean switching transitions, enabling smaller and more efficient audio devices in Class-D audio systems. Using total harmonic distortion (THD) measurements as an accepted example of industry benchmark, the use of GaN power transistors instead of common silicon has demonstrated a significant improvement in THD due to GaN's faster switching capability in Class-D amplifiers. Faster switching results in smaller "dead time," which in turn enables lower crossover distortion and a lower THD.

According to GaN Systems, design engineers have greater flexibility in tuning the performance of both THD and noise floor as a result of GaN's significantly superior switching waveform shape. As is the case for any Class-D amplifier design, increasing the frequency response is a trade-off between efficiency and noise performance. The advantage of GaN is that the frequency response bandwidth can be extended higher before the efficiency drops below, and noise floor pushes above, that of other MOSFET or integrated power device solutions. This means more design choices for audio developers to improve audio quality performance across several metrics without compromise in increased energy efficiency, smaller form factors, and lower prices.

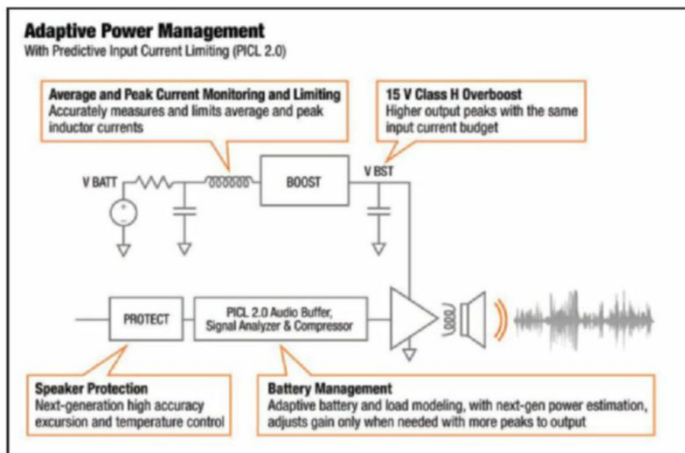
While not all GaN FET devices are equally applicable to Class-D audio amplifier applications and there are a lot of trade-offs required in the optimization of Class-D switching, the devices from GaN Systems that have been successfully implemented in actual amplifier designs show very low distortion, high efficiency, and low noise floor at the desired voltage and current levels for very audio quality audio reproduction.

As GaN Systems' CEO Jim Witham stated: "In 2020, companies who embraced GaN did so as early adopters. In 2021, the perceived risk factors will shift from not adopting GaN technology—to the risk of being left behind."

Major technology companies are already designing and shipping products with GaN today, and GaN Systems says they expect to see significantly increased demand across multiple markets in 2021.

The Mobile Extreme

At the lowest power levels—which represent the vast majority of audio amplification devices shipped—companies such as Cirrus Logic continue to "push the envelope" propelled by the extraordinary



The Cirrus Logic CS35L45 device features 7W output power capability and 15V overboost to enable higher peak power at all frequencies, resulting in more tonal balance at all volume levels, better dynamics and more spatial range and timbre.

volumes required in the mobile phone industry. As consumers increasingly use their phone's internal speakers to listen to music, podcasts, and gaming, Cirrus Logic has recently introduced its newest flagship CS35L45 boosted amplifier to enable an improved audio experience from smartphones, tablets, and mobile gaming devices. The company's new smart amplifier enables higher excursion for high-peak loudness, improved dynamic range and bass precision, with less noise and audible artifacts. It also helps to achieve improved tonal balance at all volume levels, enabling mobile device manufacturers to push audio performance to new industry benchmarks.

According to a recent user survey conducted by SAR Insight & Consulting for Cirrus Logic, 40% of consumers have increased their use of speaker mode over the previous 12 months. The global survey shows that as more people worked from home during the pandemic, they have increasingly used smartphone speakers to watch and share video via social media, participate in video calls and listen to podcasts or music while multitasking. These are all factors driving consumers' desire for better audio quality from their smartphones.

The new Cirrus Logic CS35L45 is a 15V smart boosted Class-D audio amplifier with DSP that can be easily integrated into a stereo or multi-speaker playback solution for next-generation mobile devices. By combining speaker protection and adaptive battery management features to drive higher output power in all battery conditions, the device helps to improve the performance in mid-tier designs without upgrading the device's speakers. These features result in up to 30% higher peak loudness compared to Cirrus Logic's previous generation boosted amplifier.

The CS35L45 amp's Y-bridge output driver and proprietary adaptive battery management technology helps to "look ahead" to sudden audio signal spikes in order to protect the battery, prevent pumping and clipping and maximize the audio performance at all volume levels, with no audible artifacts. 


hypexelectronics
www.hypex.nl



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


audio power conversion

© Hypex Electronics B.V. All Rights Reserved

Harman Kardon

New Citation Compact Amplifier Featuring Axign AX5689 Class-D Controller Solution

The new Harman Kardon Citation Amp offers a very interesting set of features, in a compact, Internet-connected Class-D design, extending even further the brand's Citation family of connected home audio products, including support for WiSA uncompressed wireless surround technology.

Harman Kardon, the original company founded in 1953 by Sidney Harman and engineer Bernard Kardon (which would later evolve to become Harman International), built its solid high-fidelity reputation on a series of key products. Those included early Citation tube amplifiers, followed later by Citation solid-state models, which have always embodied the brand's philosophy. More recently, when Harman decided to create its Citation family of products, the intent was to pave the way for a new generation of connected systems, starting in 2017 with the first Harman Kardon Citation smart speaker, quickly followed by the Citation Family of portable and multi-room home audio and entertainment systems, including a WiSA-enabled wireless surround sound range of speakers and soundbars.

For Harman Kardon to announce a new Citation amplifier as part of this family also meant going back to its origins, and making sure the brand would remain true to its spirit of pure, accurate sound quality.

The challenge for Harman, now part of Samsung, was to design a Class-D product that would make sense within the current Citation family of connected products, but also stand out in its technology implementation. And for that, Harman needed something new—leading its current audio amplifier design experts to approach Axign, a new company from The Netherlands, founded by former Philips and NXP employees who worked on the pioneering Class-D efforts. Axign was ready to deliver its

AX5689 audio amplifier controller chip, which uses a digital control loop with feedback that is able to improve dramatically the fidelity of existing Class-D solutions.

With a high-quality circuit design based on this technology, Harman introduced the new Citation amp, a two-channel concept that could also be easily compared to the successful Sonos Amp. Like the Sonos Amp, the new Harman Citation amp is a simple and compact unit with expanded connectivity to enable a diversity of use cases, including being adopted by custom residential integrators and installation companies.

This Class-D amplifier offers 125W of Class-D amplification per channel into 8Ω or 200W into 4Ω, while adding "streaming smarts." Connections include coaxial, optical, RCA, and HDMI ARC. Focusing on the wireless features, the Citation offers built-in Google Chromecast and Apple AirPlay, Wi-Fi (802.11ac) network, and Bluetooth 4.2 support. Support for Control4 and Crestron home automation is also possible, and the Citation Amp also offers "software smarts," which include adjustable equalization and presets to match different applications, while allowing Mono Summing and Variable Subwoofer Crossover.

More importantly, the Citation Amp also supports the WiSA wireless protocol—which uses a separate 5GHz frequency range—enabling the amplifier to connect wirelessly with the remaining Citation wireless (powered) speaker family and the growing number of WiSA-certified TVs and home entertainment sources. Considering that WiSA offers uncompressed wireless audio multichannel support, this is also a clear differentiator from Sonos and any other amp in this form-factor currently in the market, making it even more attractive for residential integration.

www.harmankardon.com | www.axign.nl



Carrying the long-standing tradition of the Harman Kardon Citation designation, the new Class-D Citation Amp evolves as a Digital Loop Amplifier controlled by the Axign AX5689 IC and audio converter. The Harman Kardon Citation Amp is the first commercially available implementation of the Axign technology, which offers a unique proposition for mainstream audio amplifier solutions, resulting in Class-A performance for Class-D efficiency, size, and cost.



The Harman Kardon Citation Amp can be used with a regular TV remote to control the volume and it actually ships with its own simple remote.

Cloud-Connected Audio Amplifiers

Comprehensively delivering on the promise of connected high-performance amplifiers, LEA Professional recently began global shipments of its 160W and 80W Connect Series amplifiers. The company's also launched the promised LEA Professional Cloud Platform, a perfect solution at a time of social distancing and increased demand for remote management.

The new pro audio amplifier company from South Bend, IN, is now shipping eight low-power models of connected amplifiers for professional-grade, system installations. The new models consist of the Connect Series CS84, CS88, CS164, and CS168 in both Network and Dante. All eight of the Connect Series low-power models are built with fully routable analog input/outputs, and with three network connection options: using the built-in WiFi Access Point, connect to an existing WiFi network, or using a Local Area Network (LAN) Ethernet connection. Once connected to a network, integrators have access to LEA's cloud platform for remote control and monitoring.

Designed and engineered to optimize configuration capabilities and control for integrators, the Connect Series amplifiers also feature DSP with built in crossovers, PEQ, limiters, and load monitoring. The models also feature LEA Professional's proprietary Smart Power Bridge, universal switch mode power supply (SMPS), and direct LoZ and HiZ selectable per channel in a one-rack-unit chassis.

Unique within the industry, all LEA Professional amplifiers leverage the built-in Amazon Web Services (AWS) IoT Core. This

cloud platform allows integrators to control and monitor critical operational data points that are integral to maintaining system health securely and remotely. The capability enhances preventative maintenance, simplifies troubleshooting, and cuts down or even eliminates on-site maintenance requirements.

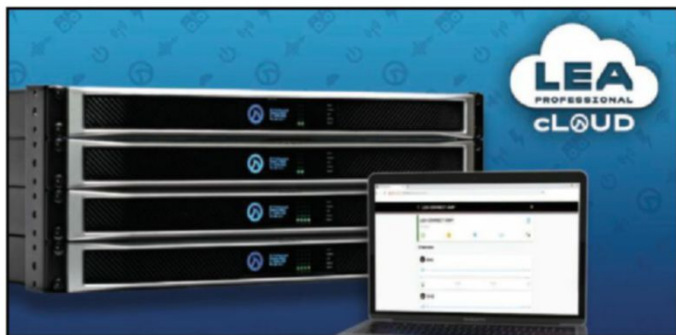
High-Power Connected Amplifiers

The most recent product introduction from LEA Professional, announced in earlier 2020, was four new Connect Series high-power amplifiers in a 2U chassis for large-format applications.

The range includes four-channel models available in configurations ranging from 1,500W to 3,000W (into 2Ω, 4Ω, and 8Ω and also 70Vrms and 100Vrms) with Dante and AES67 connectivity options and the same mention three different methods of network connection.

The amplifiers also feature advanced DSP with FIR Linear Phase and IIR Crossovers, and LEA Professional's proprietary Smart Power Bridge, universal SMPS with power factor correction and direct LoZ and HiZ selectable per channel. Like all the other Connect Series models, they are compatible with the LEA Professional Web UI and via the cloud from leaprofessional.com for control and monitoring. Neither interface requires a software download and provides control and monitoring capability from any mobile device, tablet or desktop computer.

www.leaprofessional.com



Professional audio amplifiers connected to the cloud was the original vision for LEA Professional when the company unveiled its first designs. The global pandemic totally validated the company's concept.



With the addition of the Connect Series CS84, CS88, CS164, and CS168 amplifiers, LEA now offers low-power models all with multiple network connection and remote management options.

Powersoft

Remote Management and Control for Networked Audio Amplifiers

Italian Class-D amplification and switched-mode technology pioneer Powersoft has kept busy during the pandemic period. As amplifiers are increasingly multifunctional systems that depend upon both internal features and external management and control software, the company focused on perfecting and evolving its already impressive family of solutions.

One of the most visible results was the launch of the ArmoniaPlus 2.0 software. This software update redefines what

Powersoft amplifiers can do, maintaining the same structure and features for Live Sound projects, while introducing completely new features for installations, focusing on the needs of system integrators for sources and zones management, and allowing the creation of fully customizable user interfaces and controls with the new Views Designer.

The updated software enables users to control a complete audio system from the company's range of DSP- and Dante-enabled

amplifiers. ArmoniaPlus 2.0 also now serves as the gateway into Powersoft's always expanding online resources. By registering a copy of the software, users will be able to gain access to the new MyPowersoft online platform through a single sign-in procedure.

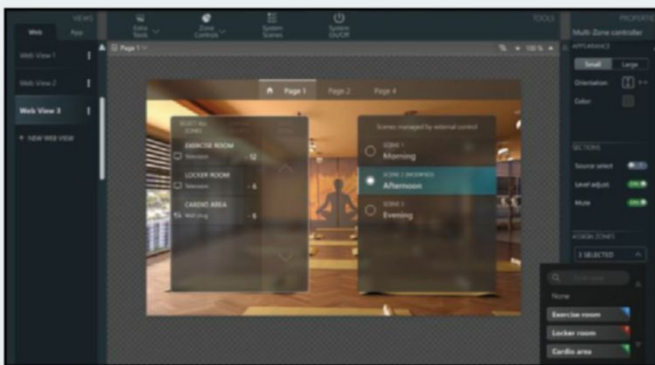
In fact, the Italian company also completely overhauled its website, improving not only the layout and navigation structure but mainly introducing MyPowersoft, an exclusive area allowing registered users to have a unique and seamless access to Powersoft ArmoniaPlus software and ProManagerPlus multi-purpose configuration tool, and gain access to an exclusive set of tools and services, including warranty status checks, E-Learning, certifications, technical support, and more.

Thanks to these improved features, Powersoft amplifiers are now capable of being remotely controlled for zone level control and system scene selection, from multiple mono, stereo, and multichannel sources and zones. This allows for decentralized system processing that eliminates the need for an external matrix in many applications, and can also reduce the number of cable runs, resulting in simpler and more effective installations.

Using the networking connectivity available in Powersoft's amplifiers, any analog or AES3 signal can be converted into Dante and shared over the network through the Dante Matrix, which is now natively integrated within ArmoniaPlus. And ArmoniaPlus 2.0 also allows sharing parameters over the network, meaning that a passive wall controller can be connected to one amplifier via GPI and control the parameters of other devices in the same network.



Powersoft continues to expand its amplifier range for the installation market offering multichannel platforms with built-in DSP, network connectivity, and the ability to feed speaker and line level systems, such as the compact Mezzo, and rack-mount Duecanali, Quattrocanali, and Ottocanali lines.



Powersoft introduced its latest Dynamic Music Distribution features, transforming installation amplifiers with innovative user interfaces that allow integrators to control entire systems and distribution applications.

The Views Designer makes it possible to create dozens of customizable user interfaces (Views) to control and monitor the functionalities of multiple sources and zones. These can be accessed from any mobile device with a web browser.

And Powersoft System Control is a dedicated app that enables users to control their system Views and to quickly load their ArmoniaPlus project designs to the amplifiers and external controls. The app configuration is also done in the new Views Designer.

Dynamic Music Distribution

The latest development from Powersoft expanded on the concept of complete remote management of audio amplifiers to offer System Integrators (SIs) the ability to scale inputs and zones conveniently and efficiently with the dynamic routing capabilities embedded on the company's fixed install amplifier platforms.

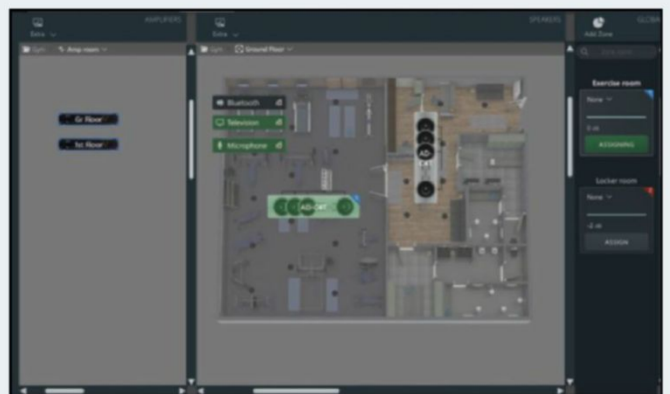
This ability makes it simple to share various music sources locally and/or between different zones in small- and medium-sized venues, independently from the source location and carry control signals in an automated fashion—all without the need of a centralized DSP. With dynamic routing, all inputs can be shared over the network as Dante signals.

Moreover, User Interfaces (UI) allow integrators to obtain control of the entirety of the system in background and foreground dynamic music distribution applications; an ability that is particularly useful in installed systems and verticals, such as bars and restaurants, hospitality, retail, and fitness establishments.

Powersoft is also complementing its amplifier remote management capabilities with a suite of hardware, software, and additional tools, such as the soon-to-be-launched Wall Mount Touch (WM Touch) controllers.

The Powersoft dynamic music distribution system can also be fully integrated into larger applications where input mixing, advanced processing, and controls are managed by third-party platforms.

www.powersoft.com



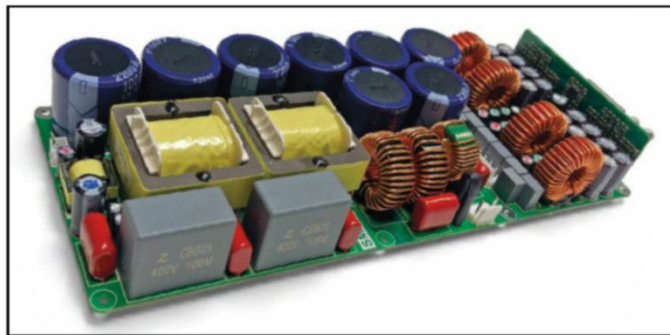
Dynamic routing capabilities over Dante networks, embedded on Powersoft's fixed install amplifier platforms, make it simple to share variable music sources locally and/or between different zones, independently from the source location, and carry control signals in an automated fashion—all without the need of a centralized DSP.

New SMA-2 Compact Four-Channel Amplifier Module

Based in Beringen, Belgium, APEX has earned a reputation for developing high-quality, innovative speaker processing and network system management products for the AV and pro audio markets, both under its own name and in support of partner brands. Now APEX has introduced a new four-channel OEM amplifier module, specifically designed for multi-way pro audio loudspeakers designs.

The new APEX SMA-2 is capable of delivering up to 3000W on a single 4 Ω channel and uses APEX's unique GlidePath direct drive architecture, which is a proprietary Class-D technology where the DSP system is an integral part of the amplifier system, with a fully DC-coupled audio signal path. GlidePath employs DC coupling throughout the signal path to reduce time offsets and distortions, and manage speaker cone excursion, prolonging speaker life by cutting mechanical wear.

Featuring onboard 64-bit DSP with a freely configurable signal path, the new APEX SMA-2 module provides an OEM solution for multi-way loudspeakers, subwoofers, and rackmount amplifiers in professional installation applications. The output power for each amplifier channel is assignable, creating possibilities ranging from high-power bridged submodules to four identical power modules or a combination of different power options. Possible configurations include 4x 750W, 2x 1500W, 1x 3000W, plus 1x 1500W and 2x 750W combined.



The SMA-2 incorporates a switch mode power supply (SMPS), two Class-D power amplifiers, and a powerful Analog Devices ADAU1462 DSP supported by high-performance ESS Technology analog-to-digital and digital-to-analog conversion stages, all on a single, compact PCB assembly. The module measures just 100mm x 250mm (3.9" x 9.8") and is ready to serve worldwide from 100-240VAC, with Active Power Factor Correction ensuring efficient use of mains power.

Like the existing SMA-1 two channel module, SMA-2 has been developed and built by APEX's in-house team in Belgium.

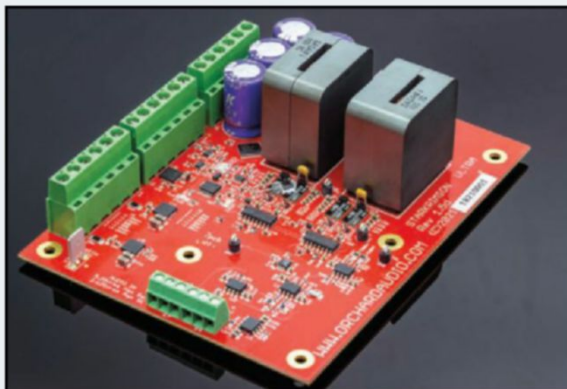
www.apex-audio.be/apex-oem

Orchard Audio

New Starkrimson 500W GaN Amplifier Modules

Orchard Audio, the innovative audio venture created by Leo Ayzenshtat, expanded its catalog of products for audio enthusiasts and integrators with the launch of Starkrimson Ultra Amp Modules. These high-performance audio power amplifier modules, utilizing gallium nitride (GaN) MOSFET transistors, expand the highly praised Starkrimson amplification line and are available as completed modules or unpopulated PCBs.

The new Starkrimson Ultra Amp modules from the New Jersey company are capable of 500W into 4 Ω and continue to explore the full potential of the latest GaN transistor technology to achieve extremely low noise and distortion for high-quality audio applications. The Starkrimson amplifier family was originally introduced by Orchard Audio as the BOSC hi-fi monoblock audio amplifier, exploring GaN semiconductor technology to achieve ultra-high performance, as reviewed by *audioXpress* in 2019. The concept, meanwhile renamed Starkrimson, allowed Orchard Audio to introduce the Starkrimson 150W Stereo Integrated Amplifier, combining two-channels of the existing modules and its popular PecanPi DAC into a single slim enclosure, with balanced analog, USB, or S/PDIF inputs.



Now, Orchard Audio has expanded the power level to 500WRMS from the previously available 150W with the new Starkrimson Ultra Amp Modules, which are also fully balanced from input to output, featuring differential and single-ended audio inputs, and capable to work at the 2 Ω level and with 20A output current. The new modules also offer protection against overcurrent, overvoltage, and thermal overheating.

Specifications provided by Leo Ayzenshtat detail 120dB (A-weighted) signal-to-noise ratio, and 250WRMS/500WRMS power output into 8 Ω /4 Ω , respectively, with a sensitivity of 5Vin for 250W into 8 Ω on the balanced input, and 2.5Vin for 250W into 8 Ω on the single-ended input.

Orchard Audio is now providing multiple options for those willing to experiment with the modules for system integration, building new designs or even consider volume production, from unpopulated PCBs (with BOM and details disclosed under NDA), or fully assembled and tested modules. There are also multiple options for add-on power supplies, including a Hypex SMPS1200A100 with Softstart solution to power two modules and support for 115 and 230VAC.

www.orchardaudio.com

NAD Electronics

Power Amplifiers with Purifi Eigentakt Technology

For audio enthusiasts considering their options for a new amplifier that uses Purifi Eigentakt amplification technology, NAD Electronics already offers the M33 Masters, a fully integrated system with DAC and streaming capabilities, reviewed in this edition of *audioXpress*.

More recently, the company has announced another high-performance, affordable and very attractive option. The C 298 (\$1,999 USD MSRP) is a two-channel design that is a perfect partner to high-quality preamplifiers or in any application where additional channels of high-end audio amplification are desired. This new NAD classic amp boasts 185W per channel into 8Ω and can be bridged to 620W × 1 into 8Ω to create a monoblock solution.

The new NAD basic amplifier is a compelling upgrade for virtually any advanced HiFi system and its robust power with upgrade flexibility will drive the most sophisticated and advanced loudspeakers on the market. The inclusion of both balanced and single-ended line inputs (switchable), make the C 298 an ideal choice for use with passive studio monitors.

For this design, NAD Electronics has moved away from the old-fashioned and very power-hungry linear power supplies. Instead, the company has developed even better performing circuits based on switch mode power supplies (SMPS) to complement Purifi's advanced Class-D output stages. These new designs are very linear over a wide bandwidth and provide consistent performance into all speaker loads, providing a dramatic advance over previous models.

The generously dimensioned power supply easily allows for 340W continuous into 4Ω and over 570W instantaneous power for short-term musical transients. Innovative Asymmetrical PowerDrive fully utilizes every last watt available with its vast reserves of dynamic power available to accurately reproduce musical transients without distortion or compression. This highly efficient supply also provides near-perfect regulation of voltage across a wide range

of conditions and provides a solid noise-free foundation for the amplifying stages.

The Eigentakt modules are manufactured by NAD under license from Purifi, allowing NAD to optimize these specifically in combination with the custom designed power supply and input stages of the C 298.

The selectable balanced inputs include a trim control useful for matching to other components. And a line out allows further addition of power on the same channel for additional speakers or subwoofers. Auto-sense with selectable threshold is perfect for automating complex systems or hiding the amp out of sight in a cabinet. NAD even included a ground lug that can be very useful for eliminating ground loops and noise in complex multi-unit systems. Added to all this connectivity is a bridge switch that turns the C 298 into an amazingly powerful monoblock amplifier (620W × 1 into 8Ω).

Masters M28 Seven-Channel Power Amplifier

NAD Electronics also expanded its Masters Series products with the M28, a seven-channel high performance amplifier that can serve as a powerful engine for a state-of-the-art multichannel system for music or home theatre.

First embraced in the NAD M33 Masters Series BluOS Streaming DAC Amplifier, the Masters M28 uses the Eigentakt Eigentakt (self-clocking) amplifier technology, manufactured by NAD under license from Purifi Audio. While there are many good sounding amplifiers on the market today, Purifi took it to a whole new level, and the measured result is less noise and lower distortion, combined with a discernible difference in the overall sound.

In common with previous generations of NAD HybridDigital technology, but unlike most Class-D amplifier designs, the M28 is not only able to easily handle long cable runs but is impervious to the loudspeaker load it is presented with, maintaining the same wide and even bandwidth response irrespective of the speaker's impedance and at the same time providing high current capability. On tap, the M28 offers 200W into 8Ω of continuous power for all channels driven at the same time, and 560W of dynamic power into 4Ω.

Eigentakt technology is very analog-like in behavior in the unlikely event the amplifier is driven into clipping. Whereas many Class-D designs can become unstable, Eigentakt behaves in a way (e.g., the benign way of clipping with instant recovery) of well executed traditional Class A/B amplifiers. And this is very important in a multichannel amplifier that needs to follow complex dynamic peaks in movie content.

For ease of installation in a variety of systems, the M28 accepts both single-ended RCA and balanced XLR inputs to provide the highest level of signal integrity with the lowest noise floor even with longer cable runs. The high-quality binding posts provide rock-solid speaker connections. The M28's styling is also designed to match the NAD Masters Series M17 V2i preamp/processor, to create a full seven-channel reference home theater system.

www.nadelectronics.com



The very simple rear panel of the NAD C 298 amplifier



The rear panel of the latest NAD M28 multichannel amplifier

Audio Amplifiers with Cloud Management and Processing Features

QSC was once globally recognized mainly as an audio amplifier company. It later expanded into manufacturing active speakers and eventually complete audio solutions. But anyone looking at the sophisticated range of networked audio amplification products from the Costa Mesa, CA, company, will be challenged to find the “audio amplifier” origins. In fact, the two latest series of audio amplification products remaining in the company’s catalog—CX-Q, and CXD-Q Series—are now named “Network Amplifiers.” And the company is now consolidating its offerings in the CX-Q range, all of which feature Q-SYS network connectivity and routable mic/line inputs.

Remaining in the more traditional line of products is the QSC DPA-Q Series of four- and eight-channel network amplifiers uniting the company’s legacy of robust Class-D hybrid power amplifiers, native network transport, and the control and monitoring capabilities of the Q-SYS Ecosystem.

The DPA-Q Series network amplifiers use a fifth-generation high-efficiency, Class-D hybrid powertrain design built upon the dependable PL380 PowerLight amplifier platform, but offering both high voltage and high current operation with excellent audio quality and thermal performance.

From its origins in musical instrument amplification, QSC is now globally recognized mainly for its Q-SYS software-based audio, video, and control ecosystem for professional installed, portable, production, corporate and cinema applications. Created from the ground-up, Q-SYS is a software-based platform built around an open IT-friendly ecosystem, created by QSC as an evolution from dedicated or closed-architecture DSP processors. Its usage of IT standard protocols makes the Q-SYS Ecosystem

highly extensible for future IT functions and platforms.

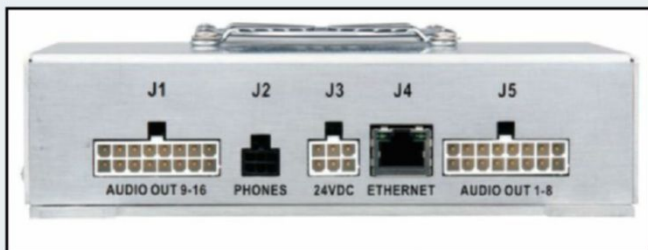
And the entire concept of networking blocks and interfaces was even more expanded in October 2019 with the acquisition of Attero Tech, a company with a highly innovative portfolio of networked AV endpoints and I/O peripherals. This acquisition boosted QSC’s development capabilities with Attero Tech’s Fort Wayne, IN-based engineering staff joining the multi-national QSC team.

Both QSC and Attero Tech were early adopters of network audio transport technology, including CobraNet, Dante, and AES67. And when QSC expanded the control capabilities of its Q-SYS Ecosystem, in 2017, it allowed Attero Tech to become the first manufacturer partner to independently develop Q-SYS Control integration plugins.

Since the integration with Attero Tech, QSC has released a significant number of new Network Audio Interfaces and I/O extenders, but also specific models of network amplifiers, now branded QSC. The latest Axon DTH1620 is a Dante/AES67 network amplifier specifically designed to support high-channel, low-power applications, such as theme park rides, attractions, and immersive exhibits. It offers 20W to each of its 16 channels in a ruggedized and compact form factor. It specializes in its ability to support both traditional passive transducers and haptic transducers for specialized audio applications.

All QSC Core processors use the same control engine and design software suite (Q-SYS Designer Software) as the rest of the Q-SYS Core processing portfolio. Recently, QSC launched the new Professional Tier feature set for Q-SYS Reflect Enterprise Manager remote monitoring and management service. This provides additional tools to truly centralize operations support for geographically disparate AV systems as well as reduce the need for technicians to be onsite in order to perform system updates. In addition, the Remote Q-SYS Designer feature enables users to troubleshoot, update, and redeploy design files to systems anytime from anywhere, without the need for complex network or remote desktop configurations. The Professional Tier joins the existing Basic and Standard Tiers for Enterprise Manager, offering end users and managed service providers a range of cloud-based service options for connected Q-SYS systems with remote management and monitoring.

www.qsc.com



The Attero Tech by QSC DTH1620 is a multichannel, low-power (16-channel x 20W) Dante/AES67 network amplifier for specialized audio applications.



Expanding on its range of networked solutions, QSC launched two new processors native to its Q-SYS Ecosystem—the smaller Q-SYS Core 8 Flex and Q-SYS Core Nano, offering a 64x64 networked I/O channel capacity with eight onboard FLEX audio channels and eight GPIOs. The Q-SYS Core Nano offers the same 64x64 networked audio I/O without the onboard analog I/O to support installations with smaller spaces with centralized processing and fully networked endpoints.

Bluesound Professional

Commercial Audio Solutions Powered by BluOS

Lenbrook International originally launched its range of audio solutions for residential integration applications with the creation of the Bluesound Professional brand. A successful combination of hardware and software solutions developed under the same roof for NAD Electronics, PSB Speakers, and Bluesound wireless multi-room systems—under the umbrella of the Lenbrook Group of companies—Bluesound Professional was created as an expansion toward commercial audio powered by BluOS, the multi-zone audio management software platform, now offering a complete streaming ecosystem for commercial spaces.

Reaching out to the commercial audio installation segment, the company has already introduced a comprehensive range of Bluesound Professional controllers, streamers, amplifiers, and speakers, all leveraging BluOS streaming. This complete solution also comes with the integration of multiple commercially licensed music streaming services, creating a totally turnkey solution.

Bluesound Professional's streamers and amplifiers fit in standard racking systems, while speakers and control panels include easy-to-install, wall mount brackets. All hardware offers a variety of standard inputs and outputs, including secure Phoenix connectors for balanced analog line outputs, balanced mic/line inputs, remote 12V trigger, IR, and system interrupt, together with hardwired Gigabit Ethernet (RJ 45) for reliable network connection at all times. Security to prevent unauthorized use of systems is built-in, while the BluOS software platform adds an easy-to-set-up solution, with network discovery, device self-discovery, and guided set-up and advanced customization options, including lock settings with password-protected accounts.

All BluOS Enabled products allow for a virtually unlimited number of zones of high-resolution grouped or independent audio environments, and interoperability among any BluOS Enabled devices within the BluOS ecosystem. The systems natively stream from music services licensed specifically for commercial use, as well as from other popular music streaming services.

Finally, the company offers numerous control options, including wall-mount control panel, IR remote, Windows or Macintosh computers, smart device app for iOS or Android, and drivers for use with popular control systems such as Crestron and KNX, among others.

www.bluesoundprofessional.com



Graeme Harrison, Vice-President and General Manager of Bluesound Professional



In a short period of time, Bluesound Professional has established a dedicated, global distribution network and brought to market a complete range of streamers, amplifiers, speakers, controls, and accessories, completing a highly anticipated suite of commercial audio products designed for the streaming age.

NAD Electronics

New High-Quality Audio Amplifiers for Installation

Using the established and reliable Hypex Ncore technology, NAD Electronics has launched multiple advanced audio amplifiers targeting installation and high-performance distributed audio systems. The NAD CI 8-120 DSP is a hybrid digital amplifier platform with high current capability, all in a slim 1U rack space. It uses a customized version of the proven Hypex UcD output stage and is capable of delivering 8x 120W per channel at 8Ω with extremely low distortion and noise in the audible range. It is also bridgeable to 4x 200W per channel at 8Ω. IHF Dynamic Power in bridged mode is 4x 350W per channel at 4Ω.



The NAD 16-60 DSP amplifier is a 16-channel commercial installation amplifier.

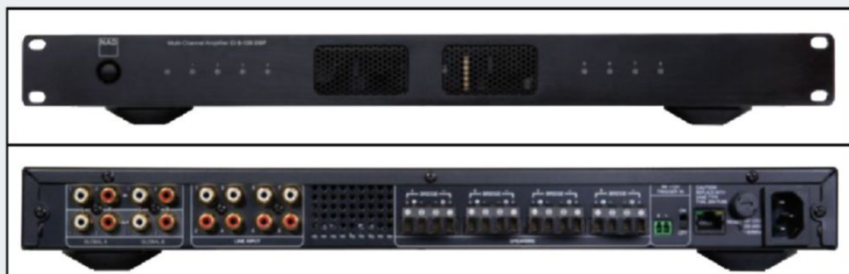
Designed to deal with the demands of the custom installation world, handling long cable runs and difficult speaker loads, the CI 8-120 DSP features the technological features demanded by today's custom integration market and allows installers to program and control a variety of functions which can be accessed through a web browser. The custom web UI can manage DSP calibration and IP control as well as offer several helpful diagnostic features. Once the installer has accessed the unit from within the network, the amplifier can be set up, calibrated, rebooted, reset, and updated.

Launched also in 2020, the CI 8-150 model is now NAD's flagship eight-channel distribution amplifier for multi-room audio systems. In addition to a dual bus system, the CI 8-150 offers multiple analog and digital inputs, including an additional two coax inputs and two optical inputs. This new amplifier

combines NAD's performance with lots of installer-friendly features, delivering a conservative 8x 150W per channel at 8Ω, it is also bridgeable to 4x 280W per channel at 8Ω with all channels driven. The Ncore amplifier platform delivers stable and efficient power with high current capability, all in a slim 1U rack space.

The latest product in the range is NAD's CI 16-60 DSP amplifier, an all-new advanced hybrid digital amplifier in a 2U rack space, also using a customized version of the proven Hypex UcD output stage. Targeted at high-performance distributed audio systems in professional installations, the CI 16-60 DSP delivers a conservative 60W per channel at 8Ω into all of its 16 channels. Each pair of channels is bridgeable to 140W per channel, if more power is desired.

The CI 16-60 DSP is also an IP-controlled amplifier, which enables the installer to configure and calibrate via a web-based user interface. The user interface offers temperature and power status monitoring, with access to multichannel digital signal processing (DSP), equalization control and a virtual patch bay. Rounding out the CI 16-60's impressive feature set are loop through jacks on all the inputs making it easy to daisy chain sources to multiple amplifiers for larger installations.



The NAD CI 8-120 DSP uses a customized version of the proven Hypex UcD output stage and delivers 8x 120W per channel.

www.nadelectronics.com
www.lenbrook.com

We wanted to show you all of the cool features of the dspNexus...

ADI SHARC DSP
8-Channel DSP Crossover

Serves as a DSP Crossover,
Music Server, & Control
Center all in one!

Integrated
Raspberry Pi

Built-in Headphone
Amplifier

Microphone input for measurements

State-of-the-art
Data Converters

Analog, USB UAC2,
S/PDIF I/O

audioweaver
Graphical Design Tools

Unfortunately, we ran out of space...

 **Danville Signal**
+1-507-263-5854
danvillesignal.com





Professional Power Amplifiers A Market in Recovery

By
James Kirby
(Futuresource Consulting)

Futuresource, a leading provider of professional audio market research, publishes regular market reports across a range of categories, including loudspeakers, amplifiers, microphones, mixers, headphones, and more. This latest article discusses the professional amplifier market and coincides with the release of its new Professional Power Amplifiers Market Assessment Report 2021.

The professional power amplifier market, and professional audio in general, has been an area of long-term market growth. Despite the rise of powered speakers, the amplifiers market has continued to see regular replacements and new opportunities over recent years, driven by innovation and demand at a macro level. However, moving into 2020, the entire professional audio market was severely affected by the outbreak of COVID-19. The professional amplifiers market itself experienced significant losses, with rates of decline well into double figures. Despite this decline, there remained some key areas of positivity and stability throughout the crisis and full market recovery to 2019 levels is expected over the coming years.

Live and Touring

The Live and Touring vertical was the most severely affected during the crisis, with the majority of amplifier revenues to this sector being “switched-off” overnight. The enforcement of social-distancing regulations and the cancellation of key events left rental providers with little requirement for new stock. From the beginning of the crisis, the touring vertical was phrased as “the first to go, and the last to come back” and moving into 2021 this held true, with regulations on large gatherings remaining in-place for many nations—and becoming stricter for some.

Over recent months, with a build-up starting at the end of 2020, improving confidence has begun to show across the touring

market. Mostly this has been driven by clear roadmaps and high vaccination rates in key nations, but also strength has been witnessed in financial indicators and the continued recovery of the installed live sound markets. Although the difficult times are far from over for touring manufacturers, government redevelopment funds and the phased return of live events is expected to provide significant recovery for the live events sector toward the end of 2021. 2022 is expected to be a record-breaking year for live events, and while this demand will take time to filter back to manufacturers, recovery is still expected within the five-year forecast period.

The market for installed live sound was impacted by many of the same trends as touring throughout the crisis. Nevertheless, throughout 2020 the market showed impressive resilience. It still experienced decline over the period, however also inspired confidence throughout the second half of the year, with many venue owners going ahead with projects and continuing to invest in audio equipment ready for the return of live events. Large projects in this category performed particularly well, driven by long-term investment plans and developments, while some smaller venues also witnessed positivity, particularly houses of worship in the US.

Amplifiers

Outside of live sound and touring, the amplifier market experienced much more stability over the past year. The Installed

Commercial segment has been an area of growth and innovation for some time, demonstrating key adoption in Class-D and audio networking technologies. Throughout the pandemic it also showed the most resilience for amplifiers, and some sub-vertical markets even experienced growth. These growth areas included AV conferencing and higher education, both driven forward by remote engagement trends necessitated by lockdowns and social distancing. Universities were forced into creating online lecture content overnight, and many chose to invest in both audio capture and sound reinforcement systems ready for the return of students.

For amplifiers, this created a new opportunity, as hybrid learning installations caused a chain-reaction of device upgrades. Meanwhile, many readers can relate to the challenges faced by corporations over the last year, with video conferencing quickly becoming a vital tool for home working. Audio also significantly benefitted from this trend, first in the home with IT products, but also later, driven by the return to physical offices. Many corporations continue to cement work-from-home and return-to-office strategies, and it's expected that over the course of this year even more demand will come for professional audio equipment in AV conferencing. A focus on quality and the use of AV conferencing and virtual events as a long-term strategy will be important in driving value for the amplifiers market.

Commercial Audio

PA/VA, life safety, and background music have consistently been strong areas of commercial audio, and throughout 2020 these segments also provided stability for the amplifier market. This was especially the case for PA/VA and life safety, driven by large projects and continued construction demand in the second half of the year. Although background music applications were some of the worst affected areas of commercial audio, they were still relatively strong compared to live sound. On average these applications depend much more on faster-paced, small business demand when compared with PA/VA, and therefore the segment was much more reactive to diminished sector revenues. Key applications include hospitality, restaurants, and retail, all of which were severely impacted by the pandemic. Despite severe declines, recovery is expected to be strong in these markets as economies begin to fully reopen worldwide.

The Future

Despite the challenging times faced by many



Powersoft's Quattrocanali 2404 amps offer the patented Smart Rails Management (SRM) technology, which allows users to maximize the efficiency of a system and drastically reduce power consumption in any load and usage condition.

amplifier vendors over the past year, confidence has begun to increase across all verticals. The installation segments have provided stability to the market, while underperforming sub-verticals are expected to rebound quickly. Although the touring and live sound segments have experienced record-levels of decline, optimism is growing and recovery will arrive within the five-year forecast period. The overall market still faces many challenges in the coming years, including fallout from the AKM factory fire, other component shortages and poor channel health. That said, these negatives will be overshadowed by continued growth in the globally vaccinated population, the reopening of the leisure economies, and the recovery of demand for key applications which depend on professional audio.

For more information about the report, visit: www.futuresource-consulting.com/reports/futuresource-professional-amplifiers. 



Here is the back view of a Stealth Acoustics SA2400, which is high-current two-channel amplifier with powerful built-in digital signal processing functions suitable as a low distortion power source.

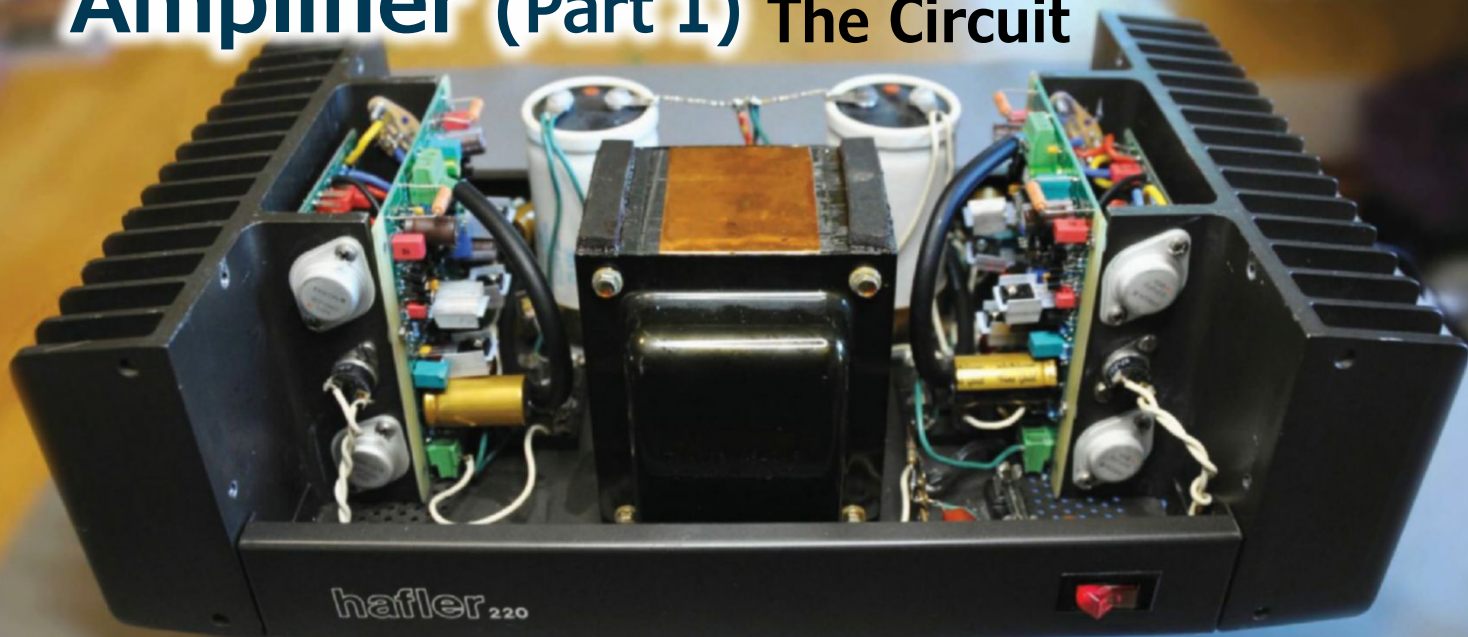
About the Author

James Kirby manages the Professional Audio team at Futuresource, reporting and consulting across product categories including professional loudspeakers, microphones, amplifiers, mixers, content creation, DSP, audio collaboration, and more. Harnessing his personal experiences as an end-user in these markets, James is driven by his enthusiasm and passion to bring high-quality research to the pro audio industry. This passion is reflected in his regular conversations with manufacturers and professionals across the supply chain. James holds a Bachelor of Science (BSc) in Business with Economics, with specialisms in Applied Strategy.



You Can DIY!

The DH-220C MOSFET Power Amplifier (Part 1) The Circuit



By
Bob Cordell and Rick Savas

Photo 1: This interior view showcases an upgrade construction project for the legendary Hafler DH-220 lateral MOSFET power amplifier.

This reference DIY project describes an upgrade construction project for the legendary Hafler DH-220 lateral MOSFET power amplifier, but it can also be used to upgrade other Hafler models or even inspire new designs. This first article describes the design and performance of the various circuit approaches, and the second article details building and testing instructions.

The Hafler MOSFET power amplifiers are legendary for their performance and price point. This amplifier was developed for David Hafler by Erno Borbely, a frequent contributor to *audioXpress* and its predecessor publications [1]. They were among the first to introduce the then-new Hitachi lateral MOSFETs to the audio power amplifier arena. These output devices were well-known for their smooth sound, reliability, and ruggedness, and as a bonus they are able to survive in the Hafler power amplifiers with no protection circuit other than a fuse. The amplifiers had good power supplies and were reasonably well-built. A huge number of them were sold and a good number of used ones are still available—both in working and non-working condition. That makes them a perfect choice for a re-build project.

This article describes a construction project for the Hafler DH-220 lateral MOSFET power amplifier (**Photo 1**). Here we describe the circuit (Part 1) and the assembly (Part 2) of the DH-220C upgrade [2, 3]. The design is implemented on two PCBs, one as a drop-in replacement for the original Hafler circuit board (analog front end, AFE) and the other as a new board (OPS) that consolidates all of the output stage wiring and components within the heatsink and under the main PCB (**Photo 2**).

This design can also be used to upgrade the Hafler DH-200, DH-500, P225, P230, and P500 amplifiers. A different output board is necessary for the DH-500, P230, and P500 because they employ three pairs of output transistors and have different heatsink profiles. The original DH-220 manual is available for reference [4]. There is also an excellent

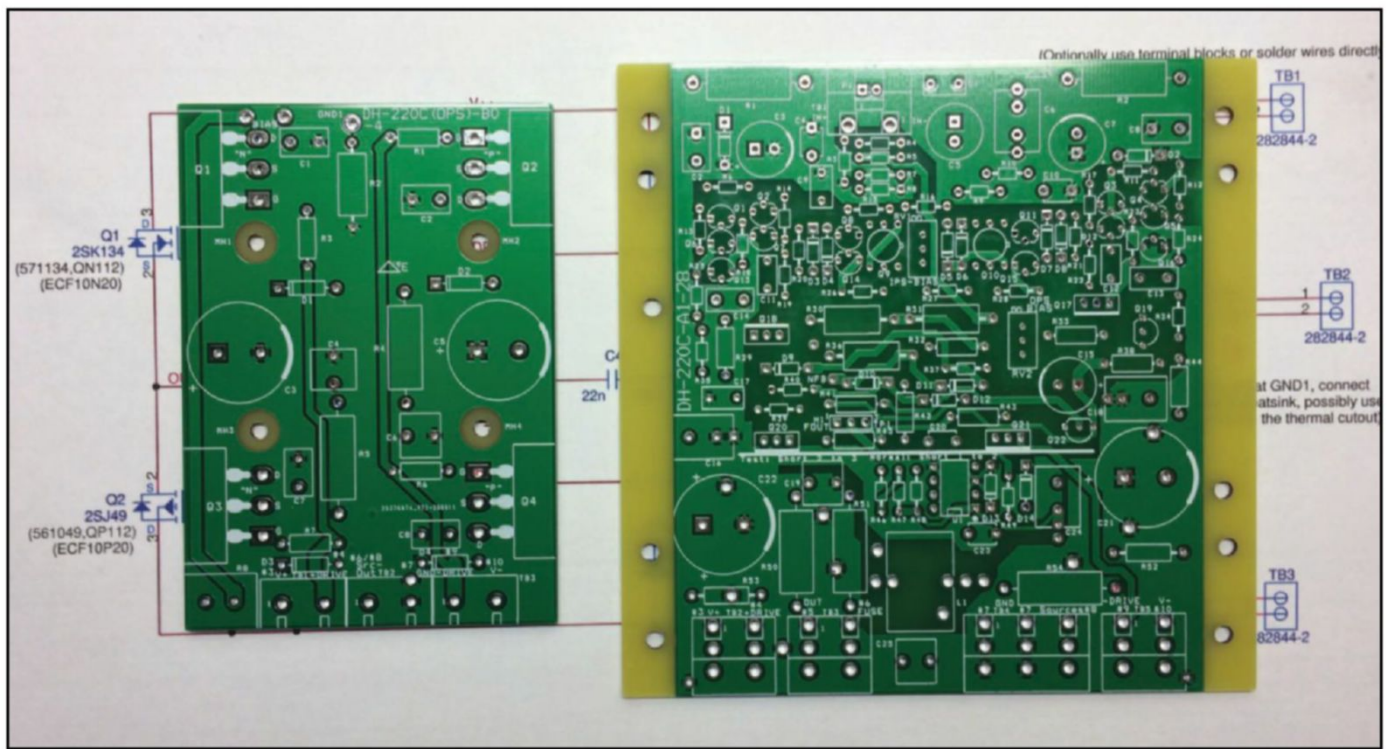


Photo 2: The new PCB set is ready for installation.

interview with Erno Borbely, designer of the DH-220 that was conducted by Jan Didden [1].

The design uses most of the original infrastructure of the DH-220, including the chassis, the power supply, heatsinks, and the original output stage power MOSFETs (**Photo 3**). The amplifier employs two output pairs and is capable of delivering more than 120W/8Ω using the original Hafler power supplies. Each output pair is biased at 200mA for low crossover distortion. The DH-220C incorporates a full-complementary JFET input stage and a push-pull VAS. The design also employs a DC servo for DC offset control [3].

The IPS/VAS

Figure 1 shows the IPS/VAS of the DH-220C. It is a full complementary JFET design using a floating tail [3]. All bipolar transistors in the design are NPN 2N5551 or PNP 2N5401 unless noted.

Dual monolithic low-noise N-channel LSK489 and P-channel LSJ689 JFETs are used for the input stage (IPS) [5]. Q9A and B, and Q10A and B form the floating full-complementary IPS, which operates at a tail current of 4mA as determined by trimmer RV1 and R16. RV1 is required to trim the tail current to its design value because of the variability of JFET threshold voltages. Source degeneration resistors R9 and R27 decrease the transconductance of the P-channel long-tailed pair (LTP) to equal that of the N-channel LTP. The JFET LTPs are cascoded to allow for higher rail voltages than the JFETs would otherwise support.

Each LTP is loaded by a current mirror, which includes an emitter follower (EF) “helper” transistor to supply the base current for the mirror transistors. The V_{be} voltage drop of the helper causes the differential collector voltages of the cascode transistors to be at the same DC potential because the subsequent VAS also includes an emitter

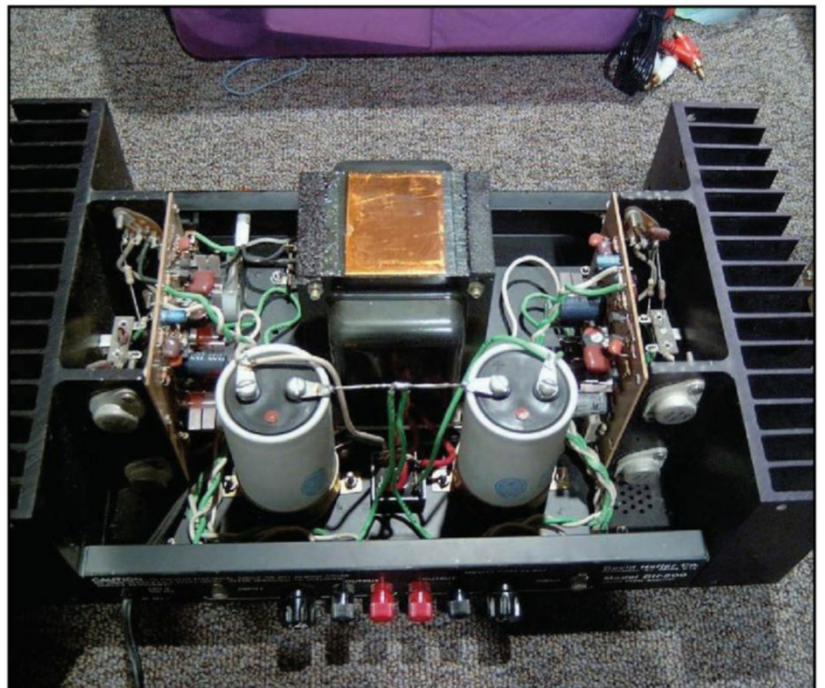


Photo 3: The design uses most of the original infrastructure of the DH-220, including the chassis, the power supply, heatsinks, and the original output stage power MOSFETs.

follower. This allows the use of the anti-parallel clamp diodes to limit differential voltage excursions to 1 diode drop. This provides cleaner clipping and also helps prevent over-current in the VAS emitter follower transistor during clipping. The 1N4149 clamp diodes are used because of their smaller capacitance as compared to the popular 1N4148.

Differential loading resistors R20 and R21 help stabilize the VAS bias current. Capacitors C13 and C14 reduce high-frequency interaction between the current mirror and the Miller-compensated 2T VAS by reducing the f_T of the helper transistors. Current mirror emitter resistors R12 and R24 are increased from 180Ω to 220Ω, due to slight V_{be} differences between the top and bottom VAS circuits.

Input and Feedback Networks

The input network provides AC coupling and two sections of low-pass filtering (LPF) for good EMI

immunity. The first LPF section employs R4 and C1. With a 100Ω preamplifier source impedance, it will be down 3dB at about 7MHz. R4 and C1 also serve to form a Zobel network that terminates the interconnect cable at the amplifier end in 68Ω at high frequencies. This is a compromise characteristic impedance for the interconnect cable. R7 and C9 form a second, more conventional input low-pass filter that is down 3dB at about 600kHz.

The feedback network consisting of R31, R30 and R36 sets the closed-loop gain at 28 with fairly low feedback network impedance to reduce noise. The 2W metal film resistors R30 and R36 minimize feedback resistor thermal distortion at low frequencies [3]. The IPS/VAS is intended for use with a DC servo, and the offset correction signal is injected by R32. Input offsets of ±150 mV can be corrected by the servo.

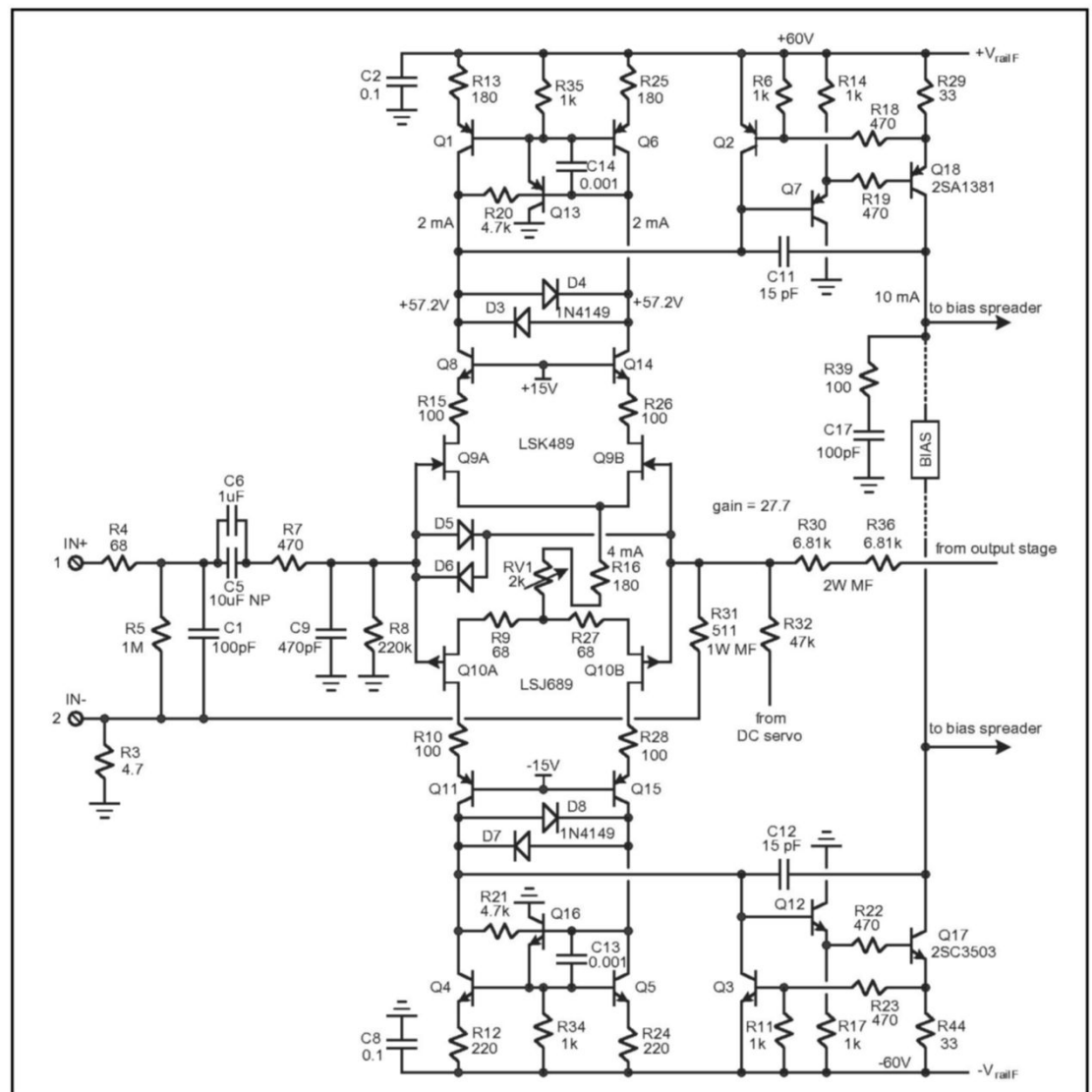


Figure 1: The schematic for the DH-220C's IPS/VAS

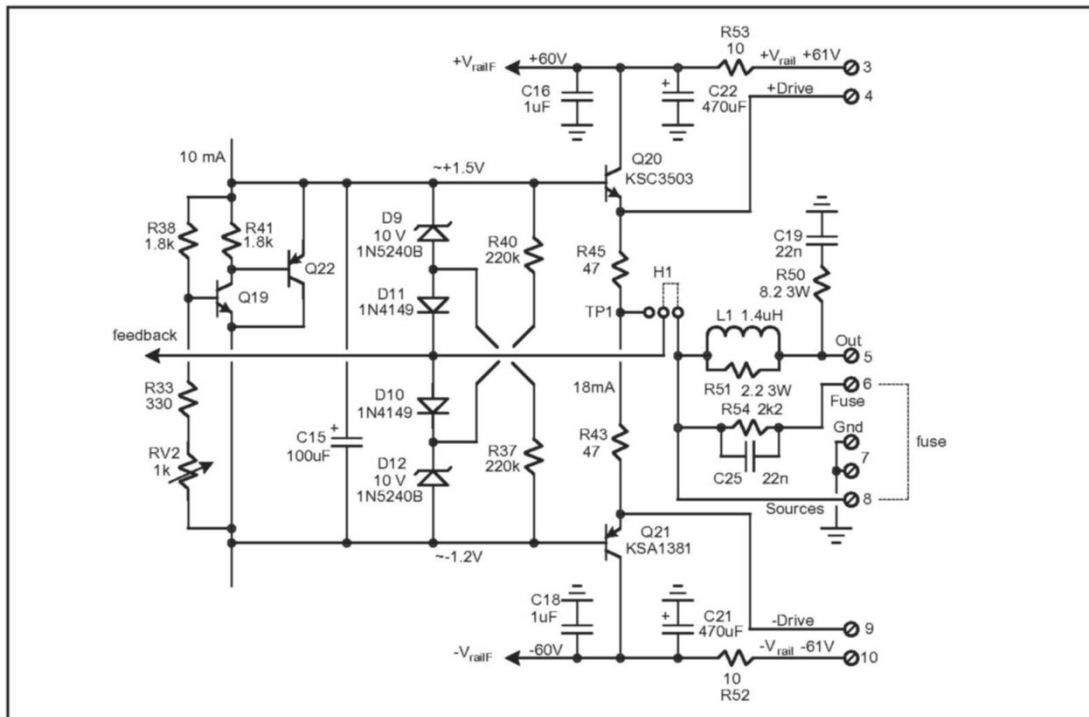


Figure 2: The schematic for the DH-220C's bias spreader, drivers, and output network

Push-Pull VAS, Bias Spreader, and Drivers

The push-pull VAS comprises a common-emitter stage preceded by an emitter follower that provides high current gain and isolation from C_{cb} of the CE stage. Q7 and Q18 on the positive side and Q12 and Q17 on the negative side implement the VAS. Quiescent current is set to about 10mA by the voltage drops across the current mirror emitter resistors R13/R25 and R12/R24. These voltage drops in turn depend on the 4mA tail current in the input stage. Q2 and Q3 provide current limiting at about 25mA. In combination with the limited voltage swing from the IPS clamp diodes, VAS transistor base resistors R19 and R22 help limit the current in emitter followers Q7 and Q12 during

clipping. C11 and C12 provide Miller compensation to set bandwidth at about 1MHz. Zobel network R39/C17 stabilizes VAS output impedance at high frequencies and suppresses peaking in the open-loop gain response [3].

Figure 2 shows the bias spreader and driver stage. The bias spreader is a CFP design that typically supplies a spread of about 2.7V to bias the lateral MOSFETs and their BJT emitter follower drivers. The CFP bias spreader provides a stiff, low-impedance voltage drop that is advantageous for full complementary designs like this, wherein the VAS bias current can be somewhat more variable than in single-ended designs where the VAS bias current is set by a fixed current source. Trimmer RV2 is used to set the quiescent output stage bias to

About the Authors

Bob Cordell is an electrical engineer who has been deeply involved in audio since his adventures with vacuum tube designs in his teen years. He is an equal-opportunity designer to this day, having built amplifiers with vacuum tubes, bipolar transistors, and MOSFETs. Bob is also a prolific designer of audio test equipment, including a high-performance THD analyzer and many purpose-built pieces of audio gear. He has published numerous articles and papers on power amplifier design and distortion measurement in the popular press and in the *Journal of the Audio Engineering Society*. In 1983 he published a power amplifier design combining vertical power MOSFETs with error correction, achieving unprecedented distortion levels of less than 0.001% at 20 kHz. He also consults in the audio and semiconductor industries. Bob authored the book *Designing Audio Power Amplifiers*, published by Focal/Taylor & Francis.



Bob worked at Bell Laboratories, Bellcore, David Sarnoff Research Center and Subcom, where his work has included design of integrated circuits and fiber optic communications systems. Bob maintains an audiophile website at www.cordellaudio.com where diverse material on audio electronics, loudspeakers, and instrumentation can be found.

Rick Savas has been involved with audio electronics from early on, building Popular Electronics kits and home brew. He attended Centennial College AT&T and earned a diploma in Electronics Engineering Technology 1980. He worked his way up to designing high-density PCBs as used in Gbit/s fiber optic networking systems. He then got back into audio design during his retirement, hooked up with Bob Cordell through DIYAudio, to make some of the designs in his book available for others to construct. Now he is onto the fun stuff—retirement, designing what he likes, and working with talents like Bob Cordell.



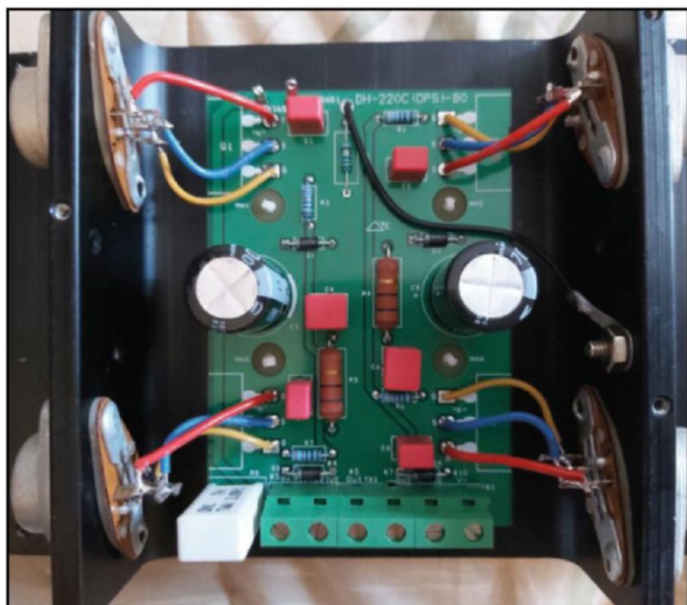


Photo 4: The new output PCB is getting mounted.

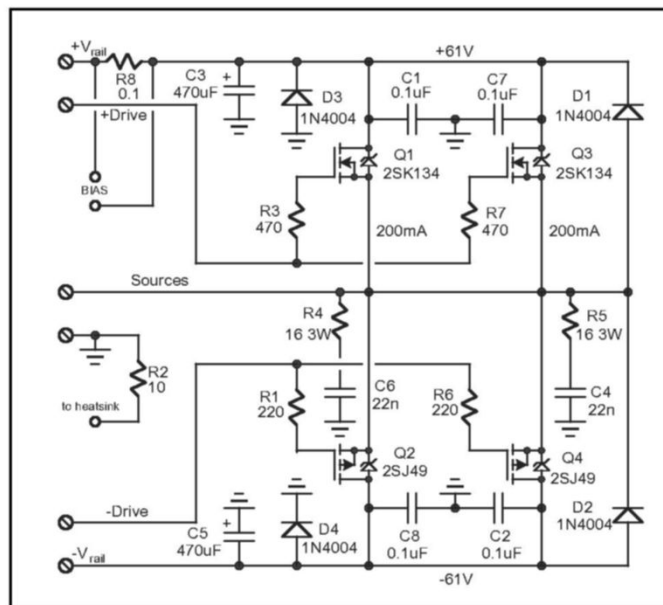


Figure 3: The schematic for the DH-220C's output stage

about 200mA per output pair. The 10-V Zener diodes D9 and D12 provide gate protection and current limiting for the output MOSFETs by limiting their gate voltage with respect to the output rail to about ± 10 V. R37 and R40 reverse-bias the Zener diodes by about 2.7V under normal signal conditions to reduce their capacitance.

The output transistors are driven by emitter followers that are biased at a healthy 18mA to be able to drive the capacitances of the output MOSFETs with adequate turn-off current even under conditions of fast turn-off. The driver bias

resistors provide a center-tap test point TP1 that allows measurement of output stage distortion alone or closure of the feedback loop without the output stage for testing purposes. This feature is implemented by header H1 and its jumper plug. Negative feedback is taken from either TP1 or the MOSFET sources in the output stage.

Output Stage

The output stage is on a small PCB that lies underneath the main circuit board within the heatsink area (**Photo 4**). It replaces the discrete fly-wiring that is used in the original design. It is shown in **Figure 3**. Gate stopper resistors are included with the same values as used in the original Hafler design. Two Zobel networks effectively in parallel (R4/C6 and R5/C4) allow the stabilizing action to be physically close to each pair of MOSFET sources, minimizing inductance. This also avoids the need for a non-inductive wire-wound resistor, using two 3-W metal oxide film (MOF) resistors [3].

R8 is a 0.1Ω resistor in series with the positive rail to the output MOSFETs. This allows monitoring of the output stage bias current for adjustment. Diodes D3 and D4 protect the amplifier from rail voltage reversal if one rail fails due to a blown fuse. D1 and D2 provide protection against inductive loads causing the output to snap outside the rail voltages. Quiescent current in each of the output transistors is set to about 200mA. This healthy bias extends the "Class-A" region to about 2.5W into 8Ω and helps reduce crossover distortion. The 10Ω resistor R2 connects the circuit ground to the

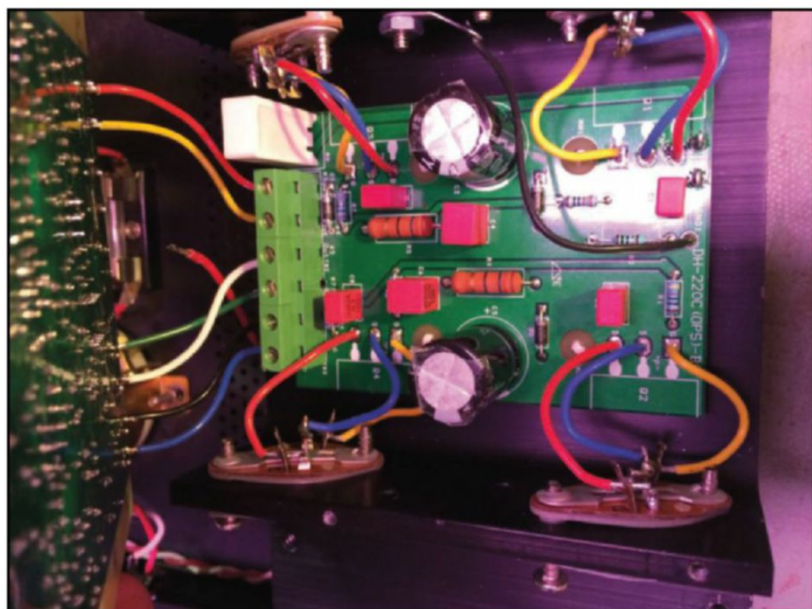


Photo 5: The new output devices have been connected.



DYNAMIC MUSIC DISTRIBUTION

High-quality music distribution in multi-zone / source applications

Powersoft offers System Integrators the ability to scale inputs and zones conveniently and efficiently with dynamic routing capabilities embedded in its amplifier platforms. This makes it simpler to share music between different zones independently from the source location, and to carry control signals in an automated fashion, and without the need of a centralized DSP.

Multiple UI options allow easy control of the whole system in dynamic music distribution applications.



SOURCES



ZONES



DYNAMIC
ROUTING



DYNAMIC
MATRIX



DYNAMIC DANTE
PATCHING



DSP



heatsink. This grounds the heatsink when the output module is being tested by itself. R2 prevents the formation of a ground loop when the output module is assembled to the amplifier (**Photo 5**).

DC Servo

The DH-220C uses a DC servo as shown in **Figure 4**. The servo can correct up to 150mV of

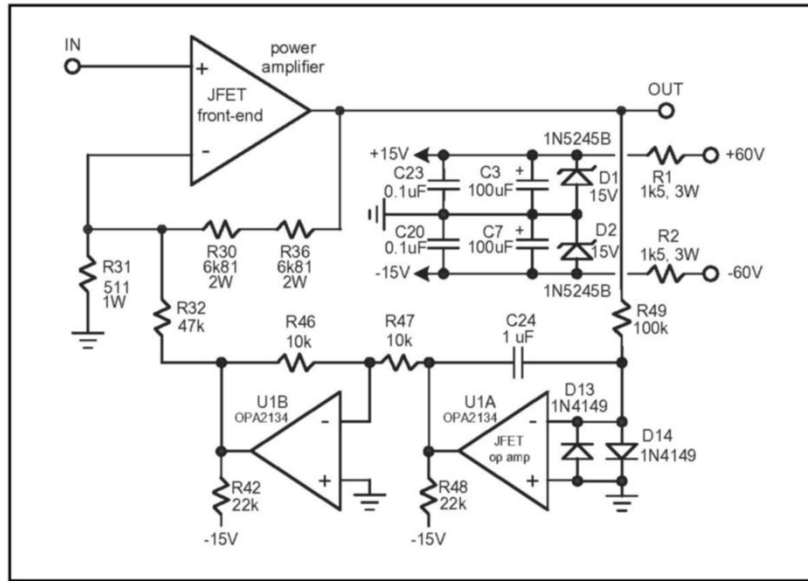


Figure 4: The DH-220C' DC servo employs an inverting integrator.

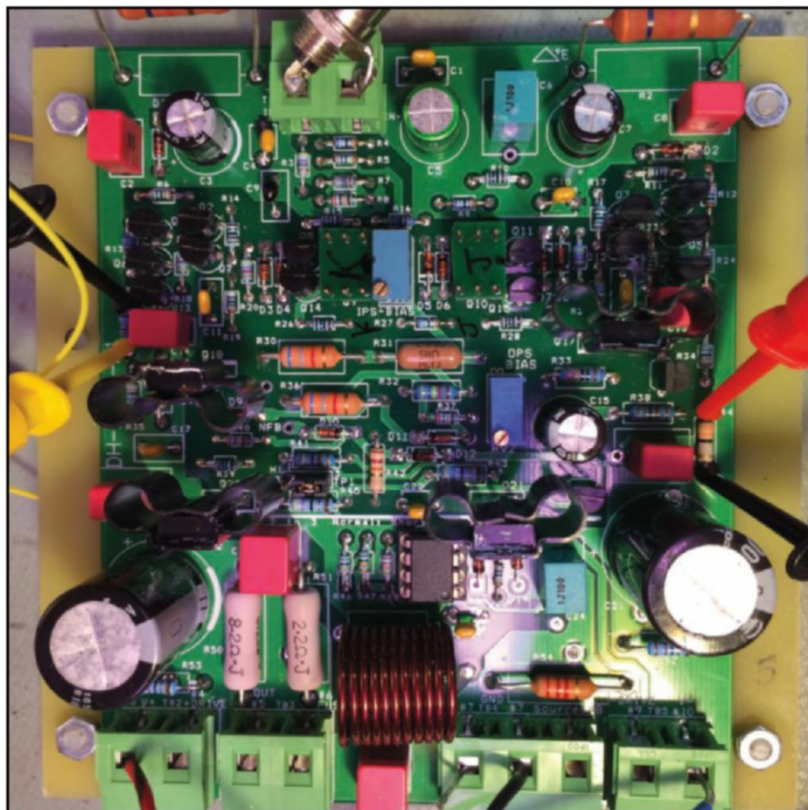


Photo 6: The completed board is ready to be tested.

input stage offset. DC servos function to control DC output offset in otherwise DC-coupled power amplifiers. A key advantage of DC servos is the elimination of the negative feedback electrolytic decoupling capacitor.

The concept of a DC servo is quite simple. The average DC level at the output is extracted by a low-pass filter, amplified, and fed back to the feedback side of the input stage. This drives the output DC value to zero or a very small value. This permits the use of a low-impedance DC-coupled feedback network while retaining high amplifier input impedance. In practice, an integrator is almost always used to provide both the low-pass filtering function and the gain. This is illustrated in Figure 4 where the input pair of the amplifier is implemented with JFETs. Without the DC servo, output offset would be 280mV with a JFET offset of 10mV.

The amplifier output is applied to a conventional inverting integrator U1A followed by a unity-gain inverter U1B to provide the proper feedback polarity. The integrator is implemented with an audio-quality OPA2134 JFET op-amp to avoid integrator offsets created by input bias current. The servo output from the inverter is applied to the feedback input of the amplifier input stage through a 47kΩ resistor. The DC offset at the output of the amplifier will be approximately the same as the offset of the op-amp, which should be well less than ±10 mV. The low-frequency cut-off created by the servo is at about 0.5Hz.

An audio system signal path often contains more coupling capacitors than necessary. There will usually be one at the output of the preamplifier, and yet another one at the input of the power amplifier. If the power amplifier has a DC servo, the input coupling capacitor of the amplifier can be eliminated, knowing that the DC output of the preamp is probably at or very close to zero. In this case, a further improvement in low-frequency transient response may result by the use of a DC servo. Coupling capacitors C5 and C6 can thus be optionally bypassed. Much more on the design of DC servos can be found in author Cordell's book, *Designing Audio Power Amplifiers* [3].

DH-220C Performance

Photo 6 shows the completed board, ready to be tested. The list below summarizes the measured performance of the DH-220C amplifier:

- Power output: 150W @ 8Ω; 260W @ 4Ω; 220W @ 2Ω (< 5 seconds)
- 1kHz THD+N: 0.002% at 120W, 8Ω; 0.0007% at 1W, 8Ω

- 20kHz THD+N: 0.02% at 120W, 8Ω; 0.006% at 1W, 8Ω
- Slew rate: >50V/μs
- Frequency response: 1Hz to 250kHz, +0, -1dB
- Damping factor: >500 from 20Hz to 1kHz; 70 at 20kHz
- 100kHz square wave response: 200ns rise time, no overshoot, into 8Ω
- Capacitive load: stable with all values of capacitance up to 1μF
- Noise with respect to 2.83V output: 104dB A-weighted; 100dB 20kHz un-weighted; 6nV/√Hz.

Figure 5 shows measured 1kHz THD+N vs. power output into 8Ω, 4Ω, 2Ω, and no-load with one channel driven. Using the DH-220 power supply that delivered 56V under load, the amplifier clipped at over 150W and delivered 120W into 8Ω with

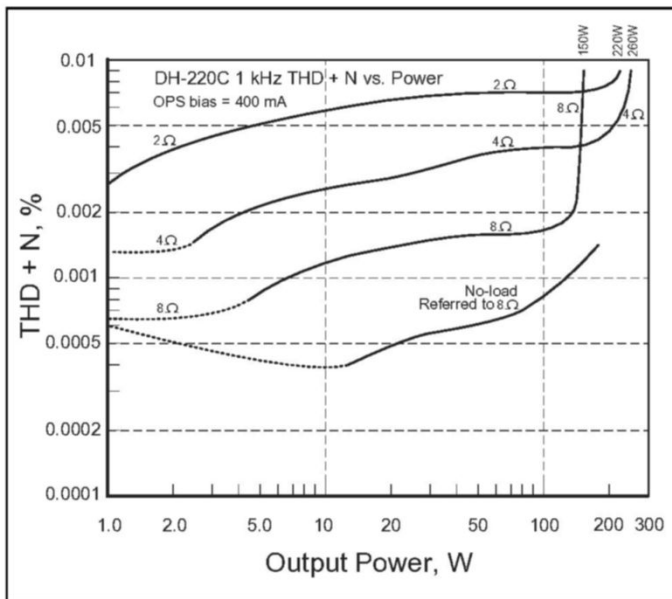


Figure 5: The graph shows 1kHz THD+N vs. Power.

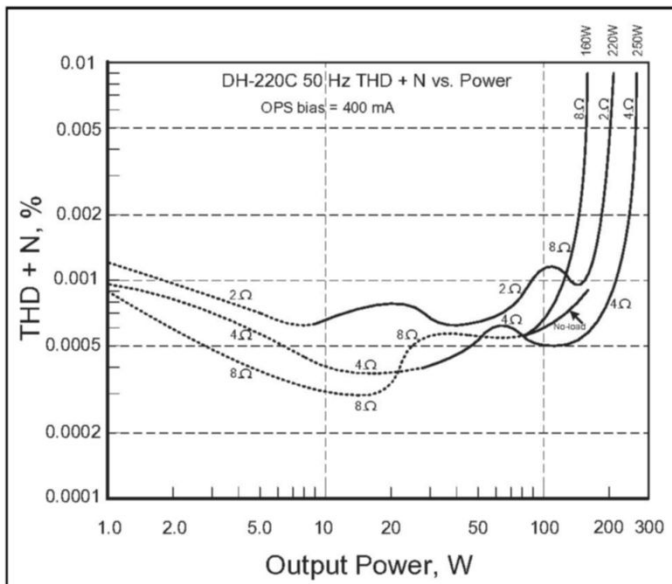


Figure 6: This graph shows 50Hz THD+N vs. Power.

THD+N of 0.002% at 1kHz. There was no rise of distortion at low power levels that would be indicative of crossover distortion. Dashed lines indicate noise. THD-1 was below 0.0011% in the critical power range between 1W and 10W. THD+N into 4Ω at 200W is only 0.005%, with clipping at 260W. THD+N at 2Ω is still only 0.008% at 200W with clipping at 220W. These measurements were taken with the load connected for less than 5 seconds to obtain the reading; long enough for the power supply to sag and short enough for the transistor junctions not to get too hot.

Figure 6 shows measured 50Hz THD+N vs. power. THD+N remains below 0.001% up to 120W into 8Ω, below 0.001% up to 200W into 4Ω, and below 0.002% up to 150W into 2Ω. At 10W, 50Hz THD+N is below 0.0007% for any of these load impedances. Very low distortion at 50Hz, even into a 2Ω load, indicates that this amplifier is capable of high output current and will provide articulate and authoritative bass reproduction. Note that 220W into 2Ω corresponds to peak output current of 14.8A. Thus, the amplifier is adept at driving difficult loudspeaker loads.

Figure 7 shows measured 20kHz THD+N vs. power. THD+N remains below 0.02% at 20kHz up to 120W into 8Ω. In the critical region below 2W it is only 0.006%. Clipping occurs at 150W with THD+N of 0.2%. Even at 200W into 2Ω, THD-20 is only 0.12%. THD increases from 1kHz to 20kHz as feedback loop gain decreases, as expected.

Powering Your Vision





Our uniquely flexible OEM ecosystem lets professional audio and AV brands bring exceptional new speaker and amplifier products to market faster and with less risk.

SMA-1 2-Channel Class-D amplifier module with integrated DSP • 1 x 750W/8ohm + 1 x 300W/4ohm • Super-compact design • Built-in active PFC	SMA-2 4-Channel Class-D amplifier module with integrated DSP • Up to 3000W on a 4 ohm channel • Versatile input configuration • Built-in active PFC
--	--

oem.apex-audio.be

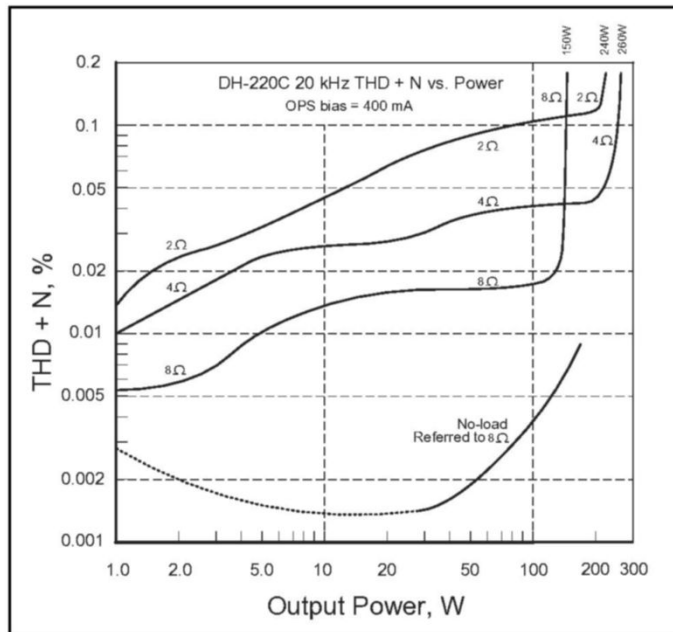



Figure 7: Here we measure 20kHz THD+N vs. Power.

Photo 7 shows the PCBs in place, ready for the final test. **Figure 8** shows measured 20kHz THD+N vs. power for the output stage alone. This shows that output stage distortion dominates, and comparison with Figure 7 allows one to see the reduction due to negative feedback around the output stage. THD+N is about 0.3% into 8Ω at 100W, about 0.5% into 4Ω at 200W and 0.8% into 2Ω at 180W. THD+N for the output stage varies little with frequency.

Summary

The DH-220C provides an upgrade path for the original Hafler DH-220 that includes a full-complementary JFET input stage

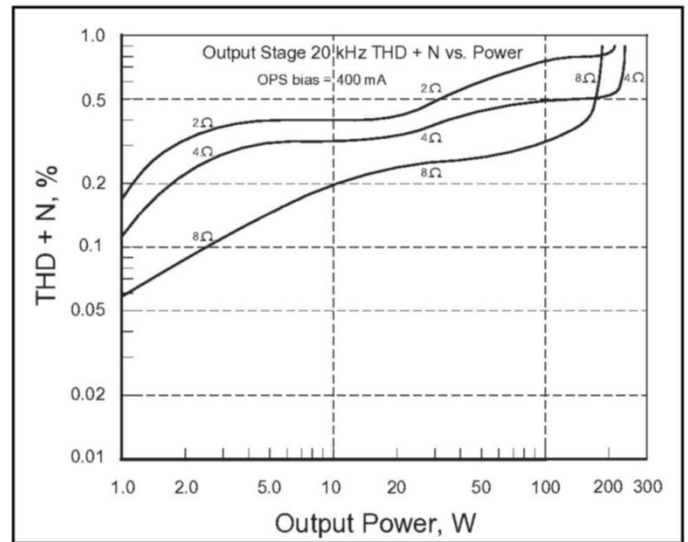


Figure 8: This graph measures output stage 20kHz THD+N vs. Power.

and a DC servo, among other circuit improvements. Reduced EMI susceptibility has also been implemented in the design. A separate output board consolidates and replaces the discrete wiring employed in the original Hafler design and places decoupling capacitors and the Zobel network closer to the output transistors, improving stability and reducing radiated nonlinear magnetic fields that can introduce distortion into the input stage.

Next Month

In Part 2, detailed assembly and testing will be described. Printed wiring boards are available for this project, as are pre-soldered LSK489 and LSJ689 dual JFETs in SOT-23 packages. The PCBs can be found on eBay with or without the JFETs. More information is available for purchasers of the PCBs. Additional information can also be found at <http://cordellaudio.com> [6].

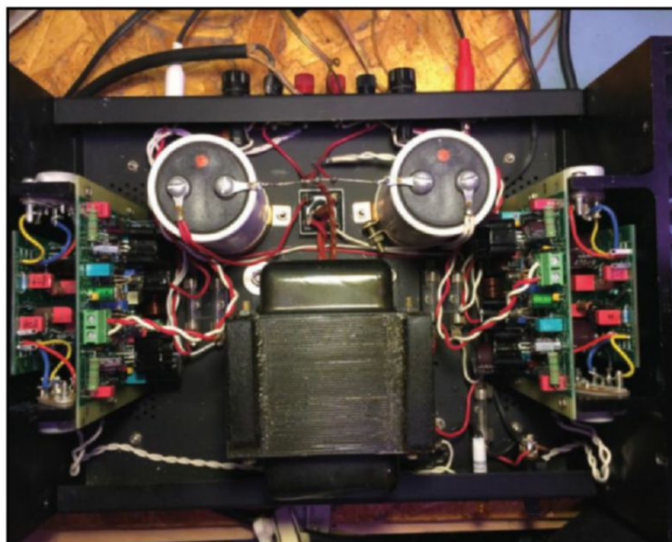


Photo 7: The DH-220C is ready for the final test.

References

- [1] J. Didden, "The Borbely Annals," *Multi Media Manufacturer*, September/October 2008, <https://audioxpress.com/article/The-Borbely-Annals-An-Interview-with-Erno-Borbely-By-Jan-Didden>
- [2] B. Cordell, "The DH-220C Audio Amplifier Design," Presented at Burning Amp 2016, www.youtube.com/watch?v=V7-27fDgqco
- [3] B. Cordell, *Designing Audio Power Amplifiers*, Second Edition, Routledge/Focal Press, 2019.
- [4] Hafler DH-220 user manual, https://hafler.com/pdf/archive/DH-220_amp_man.pdf.
- [5] Linear Integrated Systems LSK489 and LSJ689 datasheets, www.linearsystems.com.
- [6] Cordell Audio website, <http://cordellaudio.com>

ZUMI CS-A86

"The Swiss Army Knife For Audio Engineers"

ZUMI SYSTEMS



The ZUMI CS-A86 is a single board audio spectrum measurement instrument. The CS-A86 operates autonomously to execute advanced measurement sequences, apply pass-fail criteria and log all data. The device is aimed at implementation in automated test equipment for production and development. The primary function is to obtain frequency response function measurements for devices under test (DUT) that require testing of signal paths especially involving filter structures or other signal conditioning elements. The built-in capability to configure DUT's via serial interfaces allows a comprehensive set of quality control tests to be completed without user input.

Features

- Standalone audio spectrum measurement instrument
- 6 input channels
- 2 x 2ch analogue output
- Bluetooth A2DP source, HFP-AG, SPP
- Digital interfacing via USART, UART, SPI, I2C, SPP, GPIO

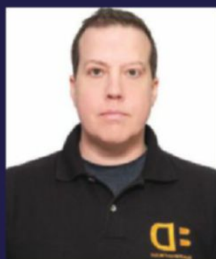
Applications

- Custom test fixtures for drivers and mics
- Automated production measurement for audio related PCBA's
- ANC filter testing
- Quality control systems with remote data reporting

THMD

*A weekly discussion on audio and headphone technologies
and the people who bring the technology to market*

THE Total
Harmonic
Discussion
PODCAST



Dave Lindberg
CEO of dB Enterprises Hong Kong



Simon Weston
CEO of ZUMI Systems Japan

New episodes every Monday at 9:15 AM EST

www.totalharmonicdiscussion.com

www.youtube.com/c/THDPodcast



For More Information contact www.zumisystems.com - www.totalharmonicdiscussion.com
For Sales Inquiries Contact dB Enterprises www.db-ent.com
Scan QR code and check out the THD Podcast (every Monday at 9:15 AM EST on YouTube)

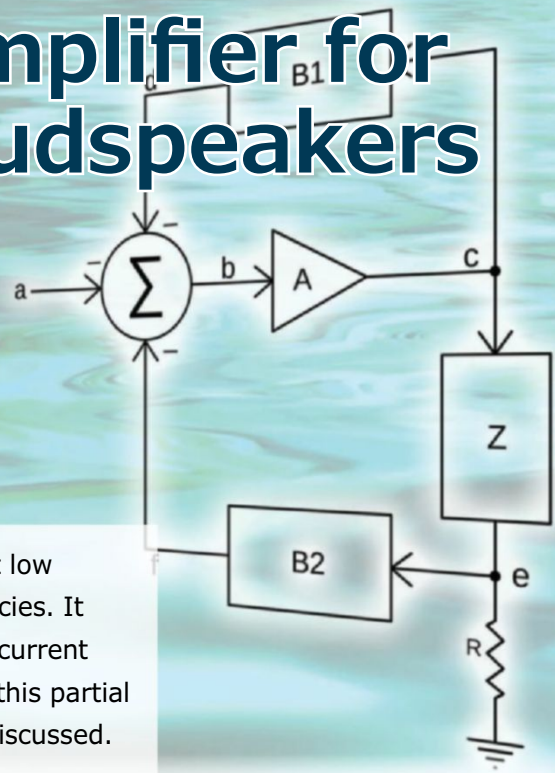
THMD 

A New Type of Amplifier for Direct Driving Loudspeakers (Part 1)

Theory Behind the Design

By
Paul Marchese

This article describes an amplifier that is a voltage source at low frequencies, then acts as a current source at higher frequencies. It can also be configured to be any ratio of voltage source vs. current source and/or be frequency independent. The properties of this partial voltage source/partial current source amplifier will also be discussed.



Above the mechanical resonance of a loudspeaker, it has been shown that a current source driving a loudspeaker may exhibit lower distortions than a voltage source [1]. Near resonance, a voltage source appears to be the better method in order to keep the damping factor high.

The Loudspeaker

A single speaker is a complicated system that can be analyzed and modeled by electrical (including moving magnetics), mechanical, and acoustical interacting subsystems. When more than one speaker is used (e.g., two-way or three-way speakers), electronic filtering (the crossover) is used to channel frequencies to drive their respective speaker. The loudspeaker is also the weakest link of the audio playback chain. Ever since magnetic wire players and 78 rpm turntables with crystal cartridges were replaced some 60 or 70-plus years ago, the loudspeaker has been by far the weakest link in the audio playback chain. To oversimplify, the 0.0001% to

0.1% distortions of amplifiers and other electronics are swamped by the 1% to 20% distortions of the loudspeaker.

The Passive Crossover

A common way to drive a multi-way loudspeaker is to have a large voltage amplifier drive a passive crossover and then drive the individual speakers (**Figure 1**). This is a common practice today even in very expensive loudspeakers. The main advantage of this method is that only two wires are needed to power the loudspeaker and these are usually at safe voltage and current levels. Another advantage is that for low component count filters, the crossover can be relatively inexpensive to make. This cost savings disappears when complex filters are designed using esoteric components.

The drawbacks of passive crossovers are many and significant [2,3,4]. Some of the drawbacks include:

- The amplifier is not driving the speaker directly but has to go through a complex impedance network before it gets to the speaker, which in itself is a complex load. So now a complex impedance source drives a complex impedance loudspeaker. If driven from a voltage source (which is usually the case), the current (force that drives the loudspeaker) is the voltage at the amplifier divided by the sum of both of these complex impedances.

Definitions to Clarify the Discussion

- A loudspeaker may be known as a driver, but not here.
- Driver = Electrical amplifier/Circuit
- Speaker = Single transducer
- Loudspeaker = Single transducer or two-way, three-way, etc. system in a box or fixture.

- The passive crossovers have serious limitations (e.g., filter complexity, component values, and non-ideal component effects). Magnetic coupling can also be a problem.

And not exclusive to passive networks, but an additional problem is:

The force driving the loudspeaker is calculated by $iL \times B$: (i) the current in the voice coil, multiplied times (L), the length of wire in the voice coil, (x) the cross product to (B), the magnetic Flux density. Nowhere is voltage mentioned in that equation.

Let's take a closer look at these three.

Crossovers

Crossover networks are derived from the passive ladder low-pass filter (LPF) networks. The high-pass filters (HPF) are transformed from the LPF by swapping capacitors and inductors. Bandpass filters (BPF) replace inductors and capacitors in the LPF with series or parallel inductor capacitor networks. If the bandwidth of a BPF is very wide, as it typically is in the midrange speaker, it is common practice not to use a BPF, but rather a HPF on the low-frequency end and a LPF on the high-frequency end. This keeps the component values from getting extreme—too large or too small. On a side note, the electrical sum of the LPF, BPF, and HPF in a three-way system will not add up to the original input signal in the time domain, even before it drives the speakers! But this is a subject for another article.

These normalized low-pass ladder networks are designed at a frequency of one radian/sec ($\omega=1$) and to be terminated into a resistive, 1Ω load. They are transformed into HPFs or BPFs, if necessary. Then they are frequency and impedance scaled appropriately

As an example, let's take the popular fourth-order Linkwitz Riley filter with a high-pass corner frequency of 200Hz, to drive a 7Ω woofer/midrange. Although the Linkwitz Riley filter is made by cascading the transfer function of two identical second-order Butterworth filters, you cannot cascade two passive filters together due to the interaction between components. Fortunately a two inductor, two capacitor ladder circuit does exist. **Figure 2** shows a Linkwitz Riley filter that was transformed to high pass, frequency scaled to 200Hz, and impedance scaled to 7Ω . There is also a two section, series RL, parallel RLC approximation of a loudspeaker electrical load. This loudspeaker model maximum impedance is 70Ω at 35.5Hz and has a Q of 4.22.

As can be seen in the HPF, the component values of both the inductors and the capacitors are fairly

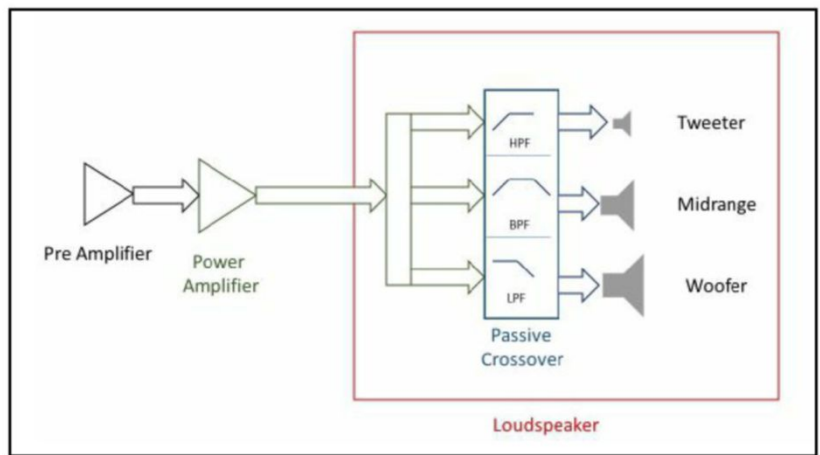


Figure 1: This is a block diagram of a typical three-way loudspeaker.

high. Avoiding electrolytic capacitors is difficult and avoiding ferrous inductors is going to be harder. This is where the sound quality can suffer. There has been a lot of discussion and analysis done on the sonic qualities of capacitors—di-electric absorption and voltage coefficients to name two—but it's the inductors that potentially can have more non-ideal problems, such as hysteresis, eddy currents, and saturation if a ferrous material is used, along with self-resonance, skin effect, and magnetic coupling. So the design of the inductors is critical. And, this is going to be an expensive filter if film capacitors are used.

Figure 3 shows the magnitude vs. frequency of this fourth-order HPF that was designed to be terminated into a 7Ω resistive load and then loaded with a two-section model of the loudspeaker. As can be seen, the voltage at the simulated loudspeaker load deviates from the resistive load. This is one part of the "art" of designing crossovers—tweaking component values in the network for a more ideal response with the actual load. Another way to compensate is to make the loudspeaker look more like a constant resistive load, using Zobel networks. The downside to this method is that it adds additional parts.

The Active Crossover

Direct driving the speakers usually uses an active crossover before the amplifiers, although a passive crossover can be used. The passive circuit here

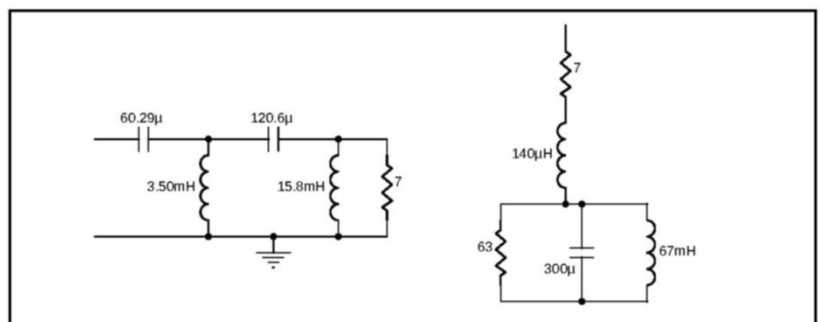


Figure 2: This Linkwitz Riley filter was transformed to high pass, frequency scaled to 200Hz, and impedance scaled to 7Ω ; N=4 passive HPF and series RC, parallel RLC loudspeaker simulator.

can be made from smaller “non-power,” less costly components since the signal is from the preamplifier, not power amplifier. **Figure 4** shows a block diagram of a typical three-way active loudspeaker.

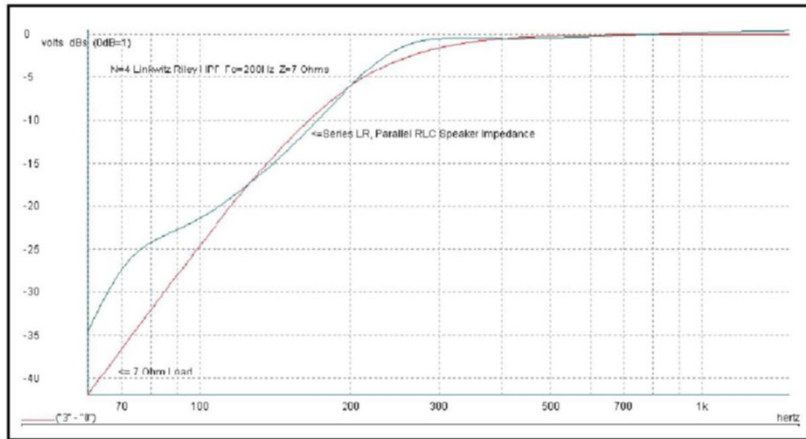


Figure 3: The graph shows the magnitude vs. frequency vs. load terminated in a 70Ω resistance vs. a 70Ω loudspeaker model.

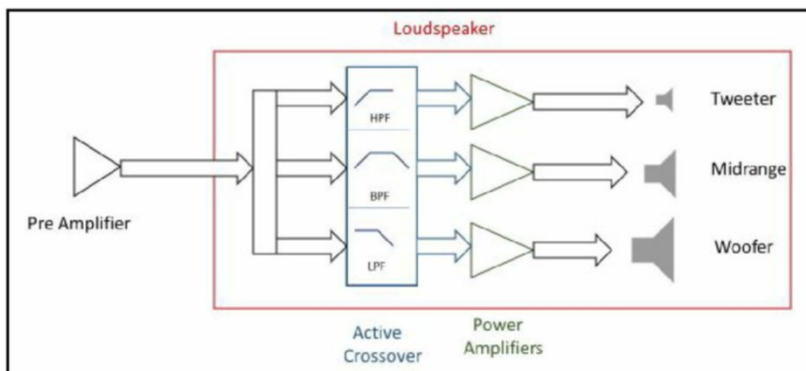


Figure 4: Here is a block diagram of a three-way active loudspeaker.

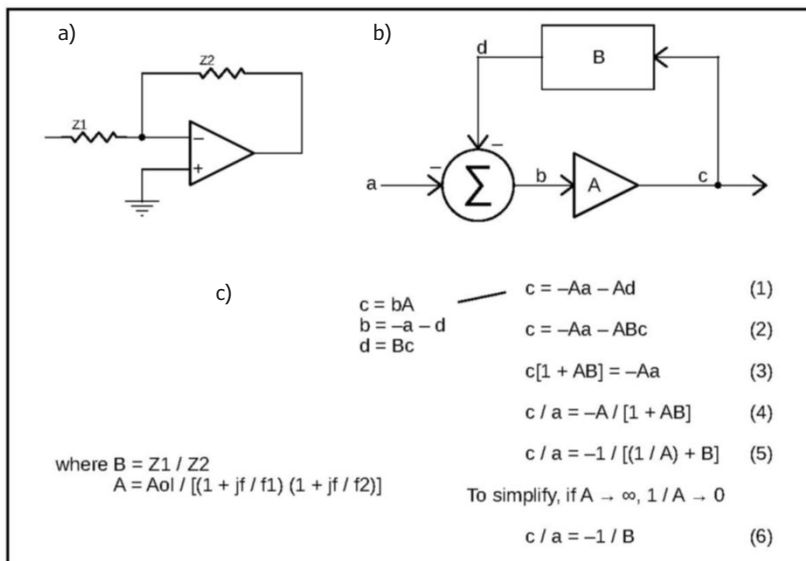


Figure 5: Here is an inverting amplifier (a); its block diagram (b); and the derivation (c) where Aol = the open loop amplifier gain, and f1 and f2 are the break frequencies.

Voltage Sources and Current Sources

The ideal voltage source (VS) has an output impedance of 0Ω. An easy load to drive is high impedance since it only has to supply little current (I) and therefore the power ($V \times I$) is low. Or from $P = V^2 / R$, the power is inversely proportional to resistance. The ideal current source (CS) has infinite impedance. An easy load for it to drive is a low impedance since it doesn't have to supply much voltage. Or from $P = I^2 R$, the power is proportional to resistance. Now an amplifier that changes from VS to CS will be half VS and half CS at the crossover point. Also this voltage source current source (VSCS) amplifier can be designed to be frequency independent with a constant VS/CS ratio of: (x)VS, and (1-x)CS, where x is between 0 and 1 ($0 \leq x \leq 1$). For example, if $x = 1/3$ then the amplifier is one-third VS and two-thirds CS. Being a partial voltage, partial current source seems contradictory on two fronts—the output impedances and power vs. load. Let's look at a frequency independent model.

Inverting vs. Non-Inverting

The inverting amplifier configuration is going to be used. One of its characteristics is that it usually has better linearity than the non-inverting circuit because the voltages at the inputs are not flying around [5]. Another is that passive networks can be designed closer to ideal since one termination is essentially ground. On the downside, the inverting amplifier does not have the high input impedance like that of the non-inverting configuration, which is mandatory (e.g., when dealing with high impedance transducers as in the sonar world). That is not the case here. The block diagram for the non-inverting amplifier is straightforward and correlates nicely with the actual circuit. The feedback factor (β) is readily calculated by a voltage divider and if resistive, its value is between 0 and 1, $[R1 / (R1 + Rfb)]$. The inverting block diagram vs. the circuit doesn't correlate as easily so two changes were made: (1) at the summer, the input signal (a) feeds an inverting input; and (2) the feedback factor (β) is changed to the ratio between the input impedance and the feedback impedance, $[Zin / Zfb]$ because it's a current summing node, not a voltage resistor divider network. This means that β can be greater than 1. **Figure 5** shows the inverting amplifier circuit, its block diagram, and the derivation of the output vs. the input.

Equation (4) from Figure 5 is commonly used in control theory. As can be seen, if $A\beta = -1$, a problem arises. I like to use Equation (5) from Figure 5 since there is only one “A” in the expression and if it is made large enough, $1/A$ is insignificant, so for simplicity Equation (6) can be used. Substituting

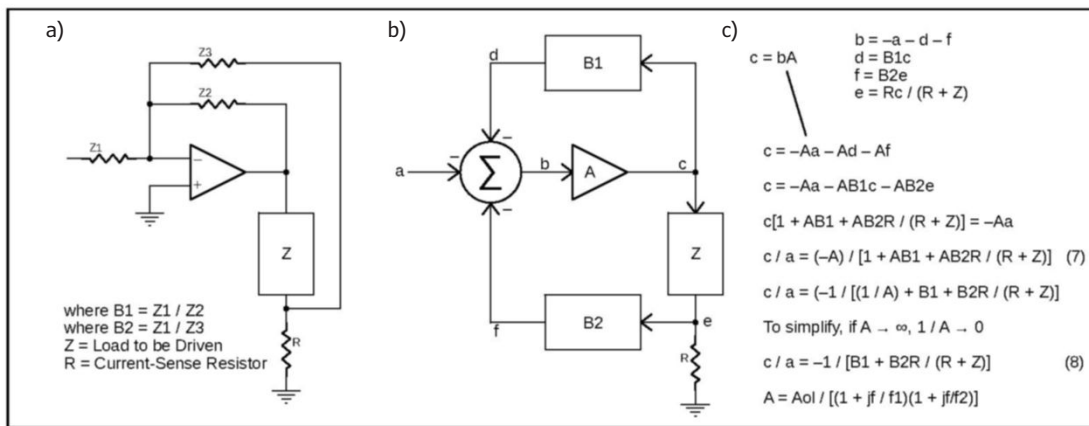


Figure 6: The VSCS amplifier circuit (a) and the block diagram (b) are shown along with the derivation (c).

$Z1/Z2$ for β in Equation (6) from Figure 5, results in the classic inverting gain equation:

$$V_{\text{output}}/V_{\text{input}} = -Z2/Z1$$

The VSCS Amplifier

Figure 6 shows the voltage source/current source (VSCS) amplifier circuit (left) and block diagram (right), and derivation. $Z2$ can be looked at as the voltage feedback path and $Z3$ as the current feedback path [(d) and (f), respectively, in the block diagram].

Like the inverting amplifier, Equation (7) from Figure 6 can be used to show instability, but this time when $A\beta1 + [A\beta2 R/(R+Z)]$ approaches -1 , the circuit is unstable. Note that now not only are the Amplifier

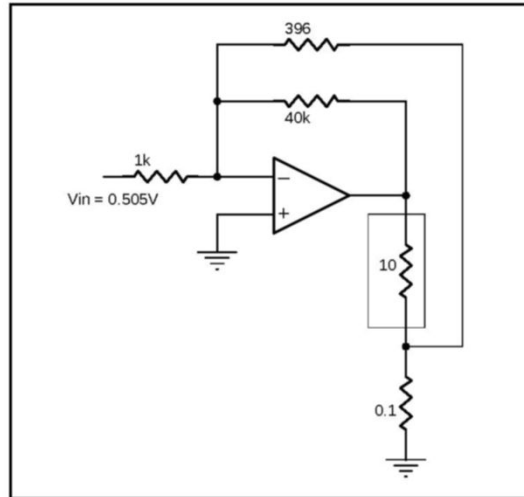


Figure 7: Here is a circuit with half VS, half CS, and a gain of 20V/V amplifier with a 10Ω load and 10W.

ACOUSTICS
 building your sound



6½" SATORI MW16TX-4/-8
 TeXtreme cone midwoofer

Read this product review in
 Voice Coil Magazine-June 2021

Find us on:

SB Acoustics

sbacoustics

SB Acoustics

sbacoustics.com

(A) and Feedback ($\beta_1 + \beta_2$) functions relevant, but also the Load impedance (Z) and the current sensing resistor (R). Equation (8) shows the simplified c/a function when 1/A is insignificant.

Equation (8):

$$c/a = V_{\text{output}}/V_{\text{input}} = -1/[\beta_1 + \beta_2 R/(R+Z)]$$

$$\text{or } -1/[\beta_1 + \beta_2/(1+Z/R)]$$

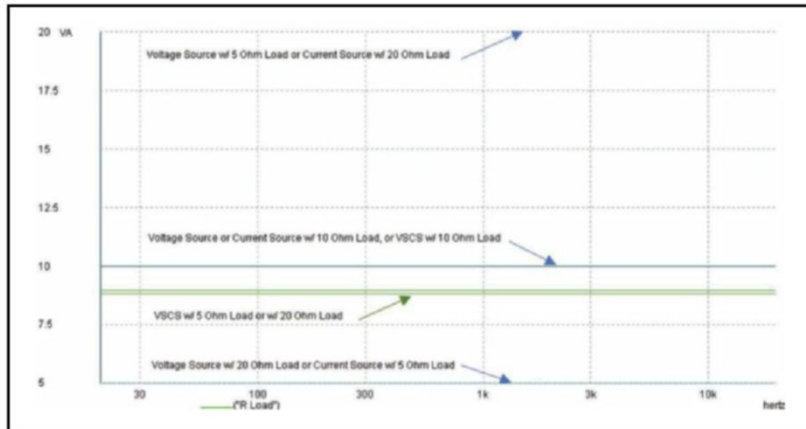


Figure 8: This simulation shows several scenarios: power into 5Ω/10Ω/20Ω loads with a voltage source, a current source, and a half VS/half CS amplifier.

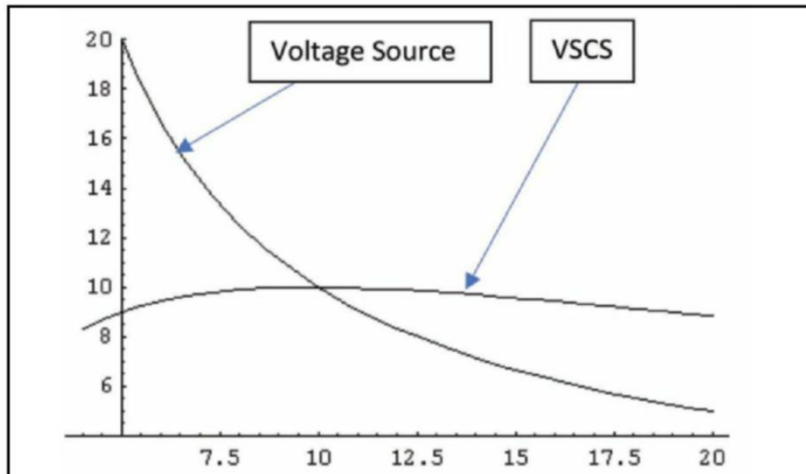


Figure 9: This graph shows power (W, the vertical axis) vs. load (Ω, the horizontal axis) of a voltage source and half VS/half CS amplifier.

About the Author

Paul Marchese served as an electrical technician in the service after high school and then got his BSEE degree from New Jersey Institute of Technology at the end of 1980. At Westinghouse, he participated in its Advanced Science Program. His entire 37 year engineering career was in analog design and included precision circuits, switched mode power supplies, low noise preamps and active and passive analog filtering from Hertz to Megahertz. His father was an electrical technician in the vacuum tube days. Recently retired, he infrequently does some contract work, but mostly passes his time trying to understand the physics of loudspeakers, climb and cut trees, (still) doing ups (push, pull, and sit), and juggling. As far as electrical history: Paul is a big fan of Oliver Heaviside and the contributions of Bell Labs (e.g., Zobel, Bode, Black, Darlington, Shockley, Shannon, Nyquist, Norton, Foster, Smith, Hamming, Karnaugh and Pindak to name a few).

VSCS Example

For a simple example let's design a VSCS circuit that is frequency independent, half a voltage source and half a current source into a 10Ω load, 10W into this load with 1/A being insignificant. The gain [6] is to be -20V/V (26dB), the current sense resistor equal to 0.1Ω and the input resistor Z1 equal to 1kΩ.

From Equation (8), if half of the gain is from voltage feedback then $\beta_1 = 1/40$. Since $\beta_1 = Z1/Z2$, then $R2 = R1/\beta_1 \Rightarrow 1k\Omega/(1/40) = 40k\Omega$.

And if half of the gain is from current feedback, then $[\beta_2/(1+Z/R)] = 1/40$.

Given Z and R, solving for β_2 , $\beta_2 = 101/40$. $R3 = Z1/\beta_2 \Rightarrow 1k\Omega/101/40 = \sim 396\Omega$

The voltage at (c) must be 10.1V to power the 10Ω load (with 0.1Ω sensing resistor) to 10W, therefore the input voltage should be: $10.1V/20V/V = 0.505V$. **Figure 7** shows such a circuit.

With a 10Ω load a VC, a CS, and the half VS/half CS amplifier will power the resistor with 10W. Now let's see what happens if the load is changed an octave in value in both directions, higher and lower. We know for a voltage source that $P=V^2/R$ and from a current source, $P=I^2R$. If the 10Ω load is increased to 20Ω, the power from a voltage source will decrease to 5W, while the power from a current source will increase to 20W. Conversely, if the load is decreased to 5Ω, the power from a voltage source will increase to 20W while the power from a current source will decrease to 5W. But what will happen using the half VS/half CS amplifier? **Figure 8** is a simulation that shows all the above scenarios.

As calculated, the power into the 5Ω and 20Ω load varies between 5W and 20W depending on which source is driving them, but the power from the half VS/half CS amplifier only decreases about 1.1W, to 8.9W with either load. It's acting somewhat like a constant power source. In this example, its power regulation with respect to the load is Δ2.2W vs. Δ15W, or almost 7 times less change in power than the purely voltage or current sources. Looking at the gain equation:

$$V_{\text{output}}/V_{\text{input}} = -1/[\beta_1 + \beta_2/(1+Z/R)]$$

all terms on the right side of the equation are constant except Z. Since R is small, $1+Z/R \approx Z/R$. So if Z is doubled, V_{out} is increased by 4/3. From $P=V^2/R$, $(4/3)^2/2 = 8/9$, which is in agreement with the simulation. Another way to look at this is since a voltage sources output power is inversely proportional to the load ($P=V^2/R$) and a current sources output power is proportional to the load ($P=I^2R$), then logically a half VC/half CS power source should be independent of the load and deliver constant power.

Figure 9 shows the power vs. load resistance of a voltage source and the half VC/half CS amplifier with the load varying from 5Ω to 20Ω . The slopes of the power vs. load curves for the voltage source and VSCS at 10Ω are about $1W/\Omega$ vs. $40mW/\Omega$, respectively, or a decrease of about 25 times.

Next Time

In Part 2 of this article, we will briefly look at magnetics—EMFs and questionable modeling—and come up with a practical feedback network on the voltage feedback side. Then, we examine the power delivered to a fairly accurate loudspeaker load and the non-ideal effects of a real-world power amplifier as well as variations of the VSCS design. 

Author's Note: At first, I thought this VSCS amplifier was an original idea, but later found out there was prior art. In the latest patent (US 7053705), the amplifier is a current source at low frequencies and changes into a voltage source at higher frequencies—just the opposite of this VSCS amplifier. Also the feedback networks are different. The derivations of the networks of the ones used in this amplifier are all shown.

References and Author Comments

- [1] E. Merilainen, *Current-Driving of Loudspeakers: Eliminating Major Distortion and Interference Effects by the Physically Correct Operation Method*, CreateSpace Independent Publishing Platform; First Edition February 2010. (Not the first person to bring to light that $F = iL \times B$ doesn't contain a voltage expression, but recently has written a book on the subject.)
- [2] V. Dickason, *The Loudspeaker Design Cookbook*, Sixth Edition, Section 7.90, Audio Amateur Press, 2000. (Vance Dickason lists eight advantages of active crossovers and direct drive vs. passive ones, including five references.)
- [3] B. Cordell, "Active Loudspeakers Power Point Presentation," Cordell Audio, www.cordellaudio.com. (Bob Cordell lists six advantages and "Too many to list.")
- [4] S. Linkwitz on passive networks: "They are a waste of time, if accuracy of reproduction is the goal," Linkwitz Lab, <http://linkwitzlab.com/crossovers.htm>
- [5] W. Jung, *Audio IC Op-Amp Applications*, Third Edition, p. 74, Sams, January 1986.
- [6] Author's Comment: "Gain" isn't easily defined since it's a combination of V/V on the voltage feedback side and A/V on the current feedback side. In this case the current gain is $1A/0.505V = \sim 1.98A/V$

CELEBRATING OVER 40 YEARS!

ACOUSTICS BEGINS WITH ACO™



**MEASUREMENT
MICROPHONES**
Polarized
&
Electret

Windscreens
3 to 18 inch &
Custom
Weather/UV
Resistant



Community/Industrial
Noise Monitors/Alarms
Indoors and Out





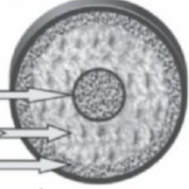
Microphone Systems
Standalone
IEPE Powered
Phantom Power
Custom

ACO Pacific, Inc Tel: +1-650-595-8588
www.acopacific.com sales@acopacific.com
Serving End-Users and OEMs Worldwide



Coax
Is Cool
Again!

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



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

KAB

Preserving The Sounds
Of A Lifetime

www.kabusa.com



Low Distortion Voltage-Controlled Floating Resistor

By
Dimitri Danyuk

This article proposes a voltage-controlled floating resistor. The circuit consists of a junction field-effect transistor (JFET) and an optocoupler with a photovoltaic output, and it can serve as an effective substitute for the H11FxM Photo-FET series from ON Semiconductor.

Voltage-controlled resistors (VCRs) are widely used in analog circuits, such as variable attenuators, variable gain amplifiers, filters with a tunable frequency response, controlled oscillators, and more. A VCR is often implemented with a JFET or a MOSFET with the drain-source resistance controlled by the gate-source voltage. In many programmable analog circuits, it is desirable that neither terminal of the VCR has connection to the control network.

The Photo-FET

This unique part (the H11FxM, **Figure 1**) uses a gallium-arsenide infrared emitter coupled to a symmetrically bilateral silicon photo detector. The detector is electrically isolated from the input and

performs like a floating field-effect transistor (FET)-based variable resistance designed for control of low-level AC and DC analog signals as described in a datasheet from ON Semiconductors [1]. The variable resistance value can be controlled within a 1:10 range or wider.

According to ON Semiconductor's datasheet [1], the voltage across the output terminals is limited to 100mV or so. My first intention was to evaluate distortion, generated by this VCR. For this purpose, I measured the distortion in a simple L-pad attenuator (Figure 1b) with series resistor $R1=600\Omega$ and a H11FxM VCR as the load. The distortion profile is shown in **Figure 2**. Second- and third-order distortion levels are about 50dB below fundamental (0.3%) at 10mVrms across the H11FxM VCR. This distortion level looks reasonably good and my next step was to compare the H11FxM's performance to a JFET-based attenuator with a photovoltaic photodiode output optocoupler.

The Photovoltaic Photodiode Output Optocoupler and a JFET

There are many optocouplers with photovoltaic

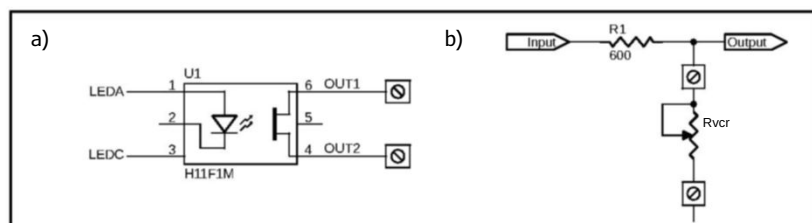


Figure 1: H11FxM symbol (a) and L-pad attenuator (b)

photodiode output (photovoltaic isolators) on the market. They can be single- and dual- channel, with outputs that can be used for direct drive of the gates of discrete power MOSFETs and/or IGBTs. The output of the optocoupler is controlled by radiation from a LED, which is optically isolated from the photovoltaic generator (a series array of photodiodes to generate 7V to 16V output). While these optocouplers were designed to create custom solid-state relays, they can be exploited in analog circuits as well.

JFETs operating in the ohmic (triode) region as variable resistors have been used in different circuits for a long time as referenced in Siliconix's Application Note AN105 [2] and in Paul Horowitz and Winfield Hill's *Art of Electronics* [3].

The drain current of the n-channel JFET in the ohmic (triode) region can be written as:

$$I_d = 2k(V_{gs} - V_{gsoff} - V_{ds}/2)V_{ds}$$

for $V_{gs} > V_{gsoff}$; $V_{ds} < V_{gs} - V_{gsoff}$

where k is a scale factor (depends on geometry of the FET, carrier mobility, etc.), V_{gsoff} is the gate to source pinch-off voltage, V_{ds} is the drain to source voltage, and V_{gs} is the gate to source voltage.

The equivalent resistance R_{ds} is:

$$R_{ds} = 2k / (V_{gs} - V_{gsoff} - V_{ds}/2)$$

It is obvious that R_{ds} is nonlinear as being a function of the voltage across channel V_{ds} . The term $V_{ds}/2$ can be canceled with addition of $V_{ds}/2$ to the control voltage (as detailed in Masanobu Chibana's US Patent, "Variable Resistor Device Using a Field Transistor" [4]) with a pair of resistors, called a "trick" [3].

Then R_{ds} becomes:

$$R_{ds} = 2k / (V_{gs} - V_{gsoff}), \text{ assuming } R_{ds0} = -2k / V_{gsoff}.$$

R_{ds} can be written as $R_{ds} = R_{ds0} / (1 - V_{gs}/V_{gsoff})$, where $R_{ds0} = R_{ds}$ for $V_{gs} = 0$;

$$R_{ds0} = 1/G_{fs0},$$

where G_{fs0} is a full conduction transconductance.

The schematic of a dual JFET controlled by a photovoltaic photodiode output optocoupler is shown in **Figure 3a**. The photovoltaic output optocoupler is loaded with external resistor R_1 . With this load, the optocoupler output voltage is reduced (compared to the open circuit) but becomes an almost linear function of the variable LED current. The measured resistance between VCR output terminals is shown in **Figure 4**. As expected, R_{vcr} is inversely proportional to LED current. There is no change in R_{vcr} for LED currents below 0.6mA.

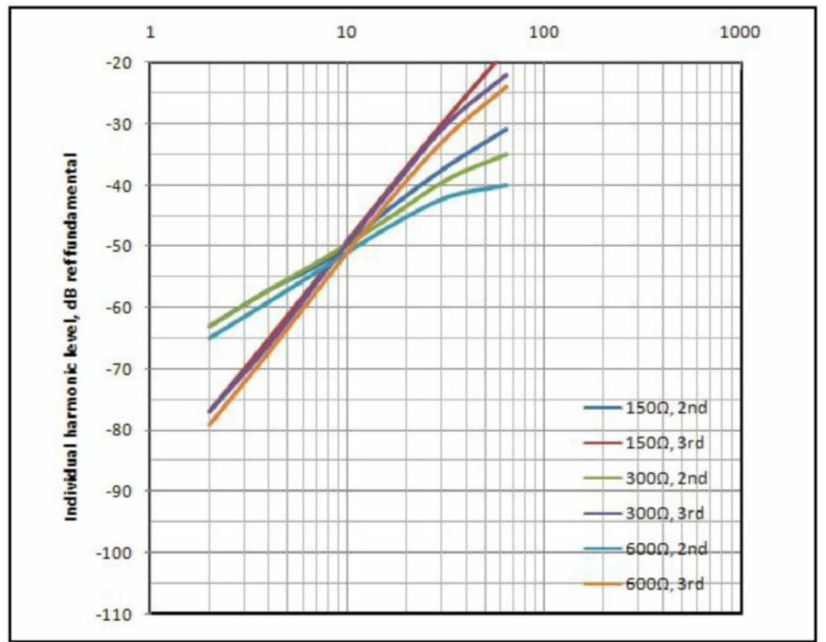


Figure 2: Levels of the second- and third-order harmonic distortion on the output of the attenuator (Figure 1) as a function of the voltage across the H11Fxm output terminals for the different VCR values (R_{vcr})

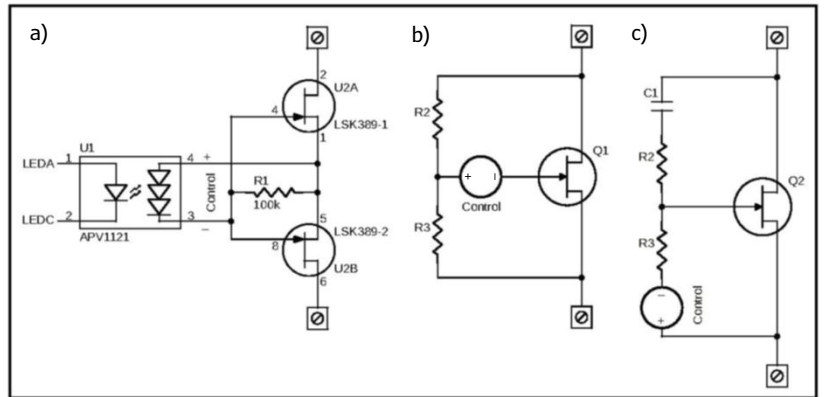


Figure 3: (a) Photovoltaic photodiode output optocoupler U1 and dual JFET U2; (b, c) VCRs with reduced nonlinearity based on a JFET in the ohmic (triode) region according to [2-4], resistors R_2, R_3 provide distortion reduction (capacitor C_1 in (c) circuit makes the gate control voltage V_{gs} equal to gate control voltage in circuit (b))

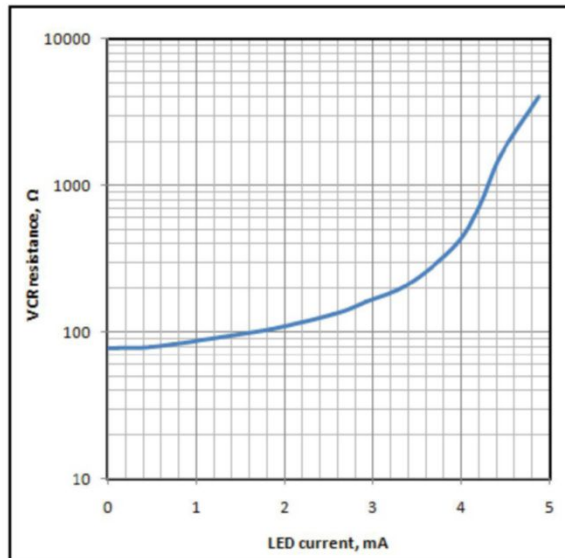


Figure 4: Typical measured R_{vcr} for the circuit shown in Figure 3a

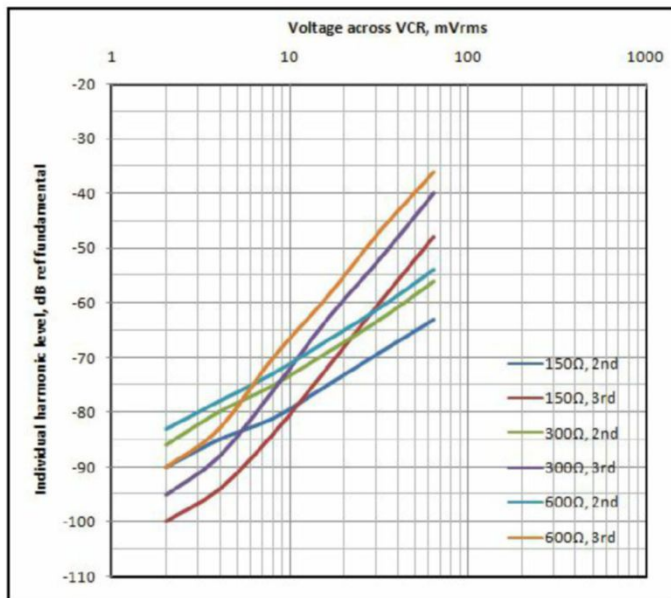


Figure 5: Second- and third-order harmonic distortion levels on the output of the attenuator (Figure 1b) as a function of the voltage across the LSK389-based VCR circuit (Figure 3a) for different R_{vcr} values.

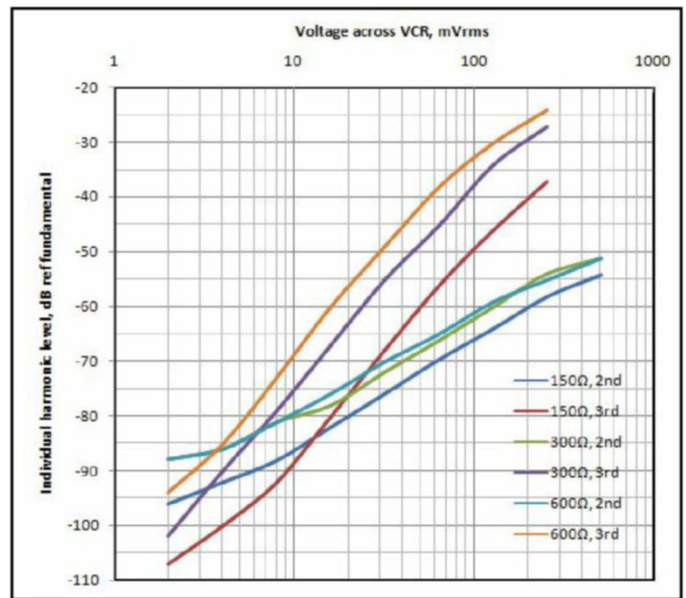
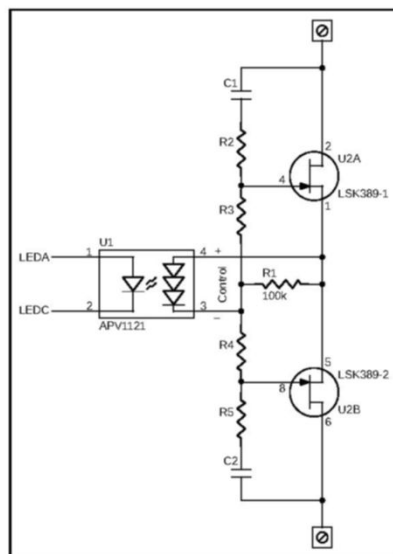


Figure 7: Second- and third-order harmonic distortion on the output of the attenuator (Figure 1b) as a function of the voltage across an LSK389-based VCR circuit (Figure 6) for different R_{vcr} values.

Figure 6: Variable resistor with reduced nonlinearity consists of photovoltaic photodiode output optocoupler, dual JFET operated in the ohmic (triode) region and resistors R2-R5. The values of resistors R2-R5 are equal.



Second- and third order harmonic distortion levels for the circuit Figure 3a are about 65dB below the fundamental (0.06%) at 10mVrms across the output terminals (**Figure 5**).

Distortion reduction in the circuit of Figure 3a occurs as $V_{ds}/2$ is conveniently added from the JFET source terminals to the control voltage. The circuit shown in Figure 3a is electrically equivalent to the circuits shown in Figure 3b and Figure 3c. All the circuits shown in Figure 3 employ the same distortion reduction mechanism.

When the drain-source voltage becomes larger than the difference between the applied control voltage V_{gs} and pinch-off voltage V_{gsoff} , the JFET moves out from the ohmic (triode) region to the constant current region (saturation). JFETs with larger V_{gsoff} (e.g., LSK489, LS26VNS, and LS26VPS) have an extended ohmic (triode) region and maintain linear resistance for larger signal amplitude. I use the LSK389 with rather low $V_{gsoff} = -1V$ in my measurements because I look for a particular range of resistance.

The next tested circuit is shown in **Figure 6**. It is based on circuit shown in Figure 3a with distortion reduction resistors for each JFET. Distortion performance of the circuit is shown in **Figure 7**. Second- and third-order harmonic distortion levels are about 70dB below fundamental (0.03%) at 10mVrms across the LSK389 with distortion reduction resistors.

Virtually any other FET (e.g., JFET and MOSFET) can be used in the circuits shown in Figure 3 and Figure 6 with the proper polarity of control voltage from the photovoltaic optocoupler.

Low-Distortion Oscillator with Different Types of VCR

To check the factual performance of the JFET with photovoltaic output optocoupler control, I decided to apply it to the highly acclaimed low total harmonic distortion (THD) oscillator designed

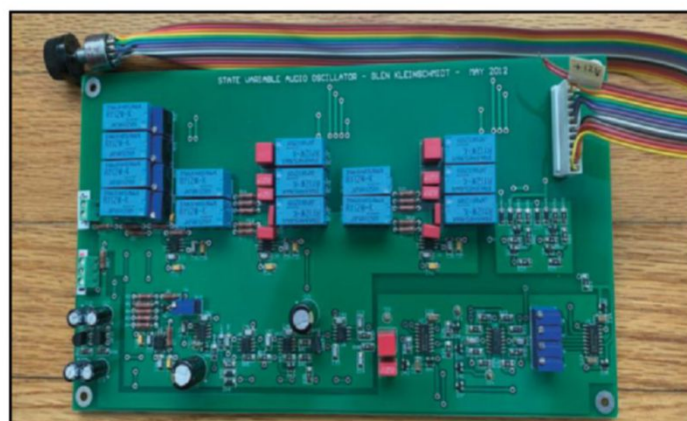


Photo 1: This is the voltage-controlled amplifier of the audio oscillator (Designed by Glen Kleinschmidt, reprinted with permission)

MENLO SCIENTIFIC

Engineering
Success
since **1983**

Engineering Consulting for the Audio Industry

We Know

- Product Development

We Provide

- Design and Technical Support

We Deliver

- Liaison, Sourcing, and Guidance

We've worked with the world's largest brands, supported large companies outsourcing production, advised manufacturing facilities on streamlining their process, and helped companies large and small to develop new products, and launch revolutionary technologies. We've helped startups find investors, and introduced entirely new concepts. Many companies learn their lesson by failing. Others have hired us first.

MENLO
SCIENTIFIC

Richmond, California, USA

- tel 510.758.9014
- mike@menloscientific.com
- www.menloscientific.com



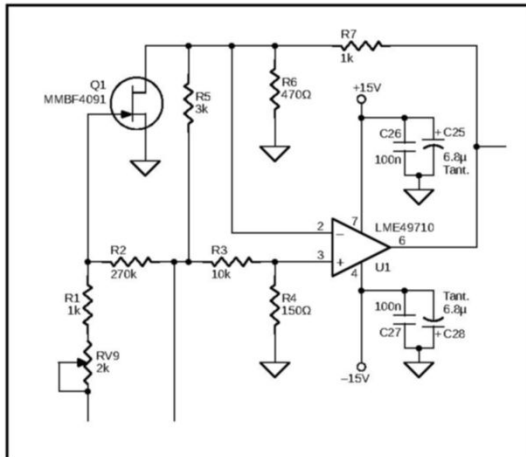


Figure 8: Here is the schematic for the voltage-controlled amplifier of the audio oscillator that is shown in Photo 1. (Designed by Glen Kleinschmidt, reprinted with permission)

by Glen Kleinschmidt [5]. The voltage-controlled amplifier (VCA) is pictured in **Photo 1** and the portion of the oscillator schematic is reproduced in **Figure 8**. The same circuit [6] was adopted in the Audio Precision System 1 (without R5). The output of the oscillator is 850mVrms at 2kHz and the voltage across the VCR (JFET Q1, Figure 8) is equal to 12.5mVrms. The distortion spectrum of the original oscillator is shown in **Figure 9**. Subsequently, Q1 was replaced with the circuits, shown in Figure 3a and Figure 6. Resistor R2 was removed; the value of R5 was reduced to $\sim 2k\Omega$ for stable oscillation. **Figure 10** shows the distortion spectrum of the oscillator with the VCR circuit shown in Figure 3a. **Figure 11** shows the distortion spectrum of the oscillator with the VCR circuit shown in Figure 6.

Noise Performance

As the output voltage range of discussed circuits is limited, it is important to evaluate noise performance. Noise was measured with the VCR attached directly to the input of a low-noise amplifier with 60dB of gain. The input stage of the low noise amplifier comprises multiply parallel LSK389 JFETs with the input voltage noise density $0.5nV/\sqrt{Hz}$ (equivalent to the thermal noise of 15Ω resistor). The noise measurement results are listed in **Table 1**.

The noise performance of the RH11FxM is surprisingly good. The equivalent noise resistance for the circuits shown in Figures 3a and Figure 6 is slightly higher than the set resistance. This can be a subject for a separate investigation.

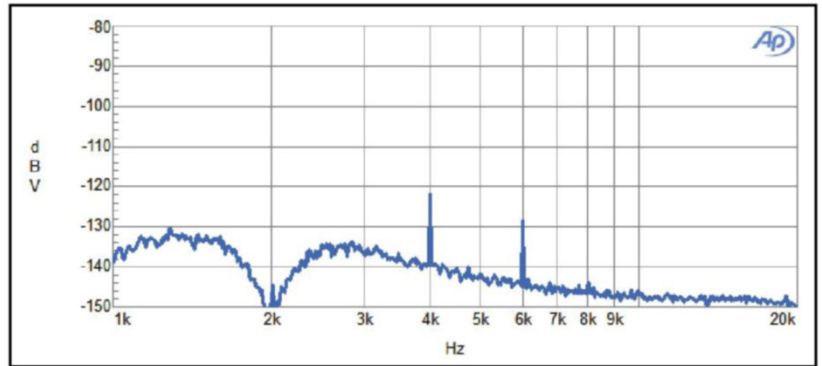


Figure 9: Distortion spectrum of the oscillator with JFET Q1 in the VCA, RV9 adjusted for lowest overall distortion

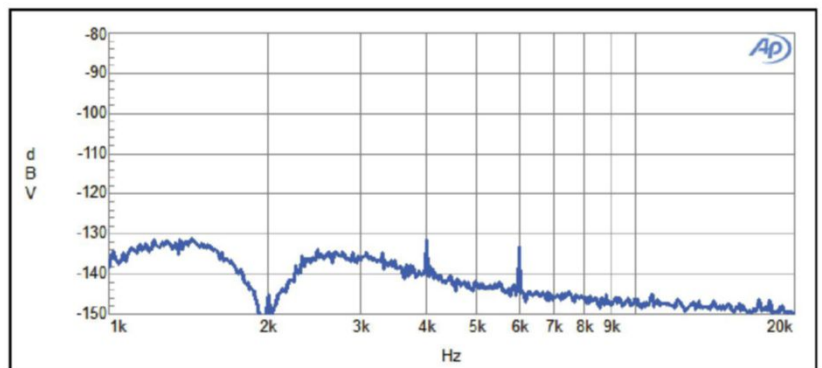


Figure 10: Distortion spectrum of the oscillator with the VCR circuit shown in Figure 3a

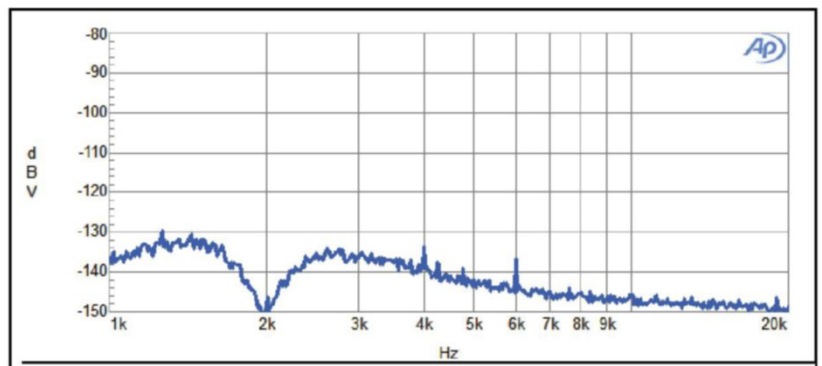


Figure 11: Distortion spectrum of the oscillator with the VCR circuit shown in Figure 6

About the Author

Dimitri Danyuk has been the principal hardware engineer at Harman Luxury Audio since 2013. He graduated with honors from Kyiv Polytechnic in 1985. His passion is to help audiophiles in all aspects of audio design. His articles have appeared in a number of magazines including the *Journal of the Audio Engineering Society* (JAES), The Institute of Electrical and Electronics Engineers (IEEE) Transactions, EDN, Electronic Design, and others. Dimitri also provides consulting services for various businesses.



Rvcr	RH11FxM (Figure 1)			APV1121 and LSK389 (Figure 3a)			APV1121 and LSK389 with a "trick" (Figure 6)		
	Noise voltage, nV, 20kHz BW	Noise voltage, density, nV/√Hz	Equivalent noise resistance	Noise voltage, nV, 20kHz BW	Noise voltage, density, nV/√Hz	Equivalent noise resistance	Noise voltage, nV, 20kHz BW	Noise voltage, density, nV/√Hz	Equivalent noise resistance
75Ω				172	1.3	100	198	1.3	100
150Ω	240	1.7	180	256	1.8	200	270	1.8	200
300Ω	338	2.4	350	350	2.5	400	430	2.5	400
600Ω	450	3.2	630	460-480	3.4	700-750	620	3.4	700-750

Table 1: The noise density and equivalent noise resistance for the RH11FxM and for the circuits shown in Figure 3a and Figure 6

CONCLUSION

The combination of an optoisolator with photovoltaic output and a JFET in ohmic (triode) region is a viable and simple approach to voltage-controlled floating resistors. The optoisolator and the JFET can be optimized for given resistance range and signal-handling capability. Fully floating VCRs allow eliminating of the interference (ground loop) from control circuit to the sensitive regulated circuit.

Isolated variable control, automatic gain control, and isolated tuning control for a filter are possible applications of the presented circuits. For other audio applications I would like to refer interested readers to the excellent review by Ron Quan [7]. If direct proportional control is required (Rvcr varies linearly with control voltage) one can use servo circuits as discussed in a Thomas L. Clarke article called "FET Pair and Op-Amp Linearize Voltage-Controlled Resistor" [8] and in an Avago Technologies' datasheet [9].

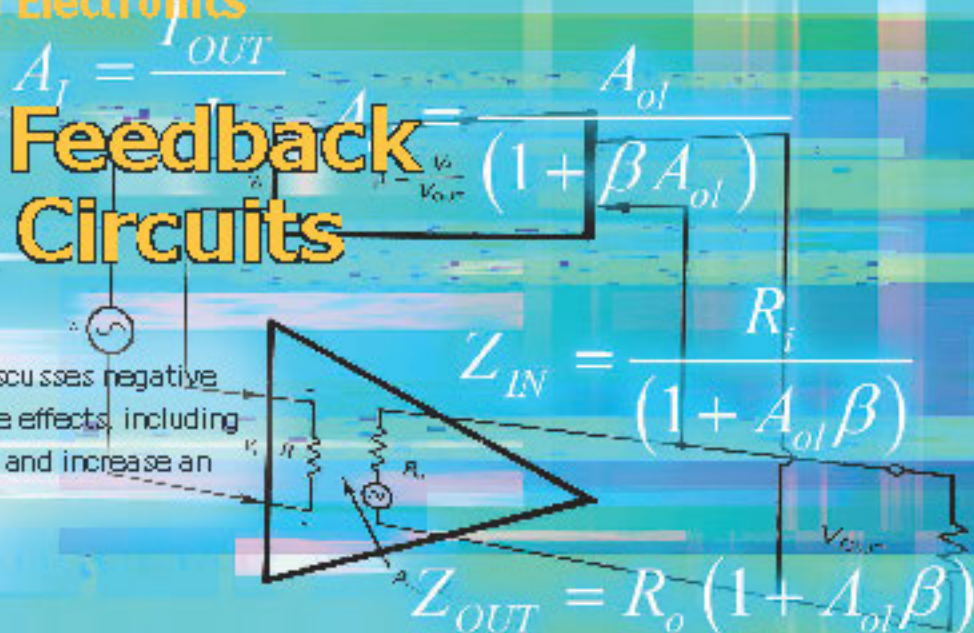
References

- [1] "H11F1M, H11F2M, H11F3M Photo FET Optocouplers," Datasheet, ON Semiconductors, May 2012.
- [2] "JFETs as Voltage-Controlled Resistors," Application Note AN105, Siliconix, 1997, www.linear-systems.com/lldata/others/AN105FETs.pdf.
- [3] P. Horowitz and W. Hill, *Art of Electronics*, Cambridge University Press; Third Edition, 2015; Chapter 3.2.7: "FETs as Variable Resistors," pp.161-163.
- [4] M. Chibana, "Variable Resistor Device Using a Field Transistor," US. Patent 3715675
- [5] G. Kleinschmidt, "An Audio T.H.D. Analyzer," oscillator schematic can be found at www.diyaudio.com/forums/group-buys/221670-sv-oscillator-pcb-buy.html or www.eevblog.com/forum/buysellwanted/who-would-like-an-ultra-low-distortion-audio-oscillator-pcb/25/Glen
- [6] A. C. Barber, "Linearization of Voltage-Controlled Amplifier Using MOSFET Gain Control Circuit," US Patent 5808516
- [7] R. Quan, "A Guide to using FETs for Voltage-Controlled circuits," Parts 1-5, *EDN*, 2017, October 13, 2017; January 2, 2018; February 6, 2018; and April 10, 2018; and July 3, 2018.
- [8] T. L. Clarke, "FET Pair and Op Amp Linearize Voltage-Controlled Resistor," *Electronics*, April 28, 1977; pp. 111-112.
- [9] "HCNR200 and HCNR201 High-Linearity Analog Optocouplers," Datasheet, Avago Technologies, July 1, 2014.

Negative Feedback Amplifier Circuits

In this article, our columnist discusses negative feedback and details its positive effects, including its abilities to reduce distortion and increase an amplifier's frequency range.

By
Richard Honeycutt
(United States)



In the early years after Lee De Forest invented the triode vacuum tube, engineers attempting to use the device had frequent problems with oscillation in their amplifier designs. In other words, their amplifier circuits whistled and howled. It was soon discovered that this problem was caused by positive feedback: Some of the output signal was being fed back in phase into the input of the amplifier. In 1927, H. S. Black of Bell Telephone Laboratories invented a way of using feedback to control tendencies toward oscillation. He applied a phase-inverted portion of the output signal to the input of the amplifier. In addition to eliminating the troublesome oscillations, this negative feedback had other benefits: It reduced distortion and increased the frequency range of the amplifier.

Types of Negative Feedback

For any amplifier circuit, we can derive a feedback signal that is proportional to either the output current or the output voltage. In addition, we can add the feedback signal to the input signal by connecting the feedback network either in series or in parallel (shunt) with the input signal source. This leads to four types of negative-feedback connections, as illustrated in **Figure 1**: voltage-series, voltage-shunt, current-series, and current-shunt. The generic amplifier symbol—a triangle—is used to represent the hollow-state gain stage.

In order for the feedback signal to be able to be fed in series with the input signal source, that source must have a low internal (Thevenin) impedance. Therefore, the series feedback connections imply voltage sources for the input signals. Likewise, since a low-impedance signal source would tend to short out a feedback signal applied in parallel, the shunt-feedback connections

imply a signal source with a high internal impedance, which means a current source.

Because negative feedback tends to stabilize the quantity to which it is made proportional, voltage-feedback circuits act as constant-voltage sources, whereas current-feedback circuits act as constant-current sources. If we put this information together, we have what is called the controlled-source method of classifying amplifiers, as discussed in the Hollow-State Electronics article published in the October 2014 issue of *audioXpress*.

Voltage-Series Feedback

Let's begin by examining the voltage-series feedback circuit. Referring to **Figure 2**, you will notice a number of symbols. The amplifier's gain without feedback is indicated as A_{vol} : open-loop voltage gain. (The words "open-loop" indicate that the feedback loop is not closed; therefore, there is no feedback when A_v is measured.) The portion of the output that is fed back is given by β (a lowercase Greek beta). (Sometimes β is given a + or - sign corresponding to positive or negative feedback. When this is done, the factor $(1 + A_{vol}\beta)$ that appears in many of the equations below must be changed to $(1 - A_{vol}\beta)$. Thus, the feedback voltage V_f is βV_o . The source voltage is V_s , and the total voltage applied to the input of the amplifier is V_i . R_i and R_o are the input and output resistances of the amplifier; and R_L is the load resistance.

With no feedback, $V_s = V_i$, so the gain is simply A_{vol} , the open-loop gain of amplifier itself.

$$A_{vol} = \frac{V_o}{V_i} = \frac{V_o}{V_s}$$

Once we connect the feedback loop, the input signal to the amplifier becomes $V_i = V_s - V_f$

Notice that we have assumed that the feedback signal is out of phase with the signal as indicated by the minus sign in front of V_f . Let us skip a few steps of algebra and calculate V_{OUT} :

$$V_{OUT} = \frac{A_{vol} \times V_s}{1 + A_{vol} \beta}$$

Now the voltage gain with feedback (the closed-loop gain) can be found, and it is given by:

$$A_{vol} = \frac{V_{OUT}}{V_s} = \frac{A_{vol}}{(1 + A_{vol} \beta)}$$

So we see that voltage-series negative feedback decreases gain in an amplifier.

The input and output impedances of an amplifier using voltage-series feedback are affected by the presence of the feedback. Thus, they are not identical to the impedances without feedback, R_i and R_o . Again skipping some algebra, we get:

$$Z_{IN} = R_i (1 + A_{ol} \beta)$$

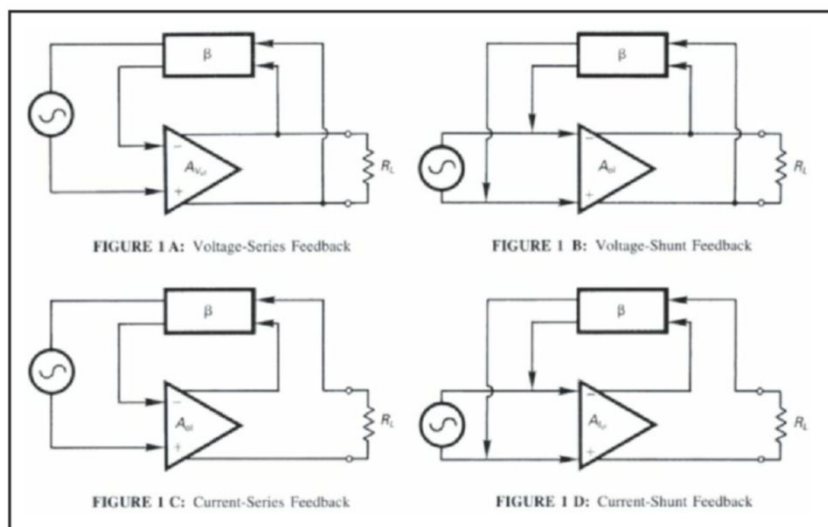
So voltage-series negative feedback increases an amplifier's input impedance. How about the output impedance? It is decreased by this form of negative feedback:

$$Z_{OUT} = \frac{R_o}{1 + A_{ol} \beta}$$

Rather than the generic amplifier schematic, let's look at an actual hollow-state stage using voltage-series feedback. **Figure 3** shows the schematic of a single-stage hollow-state power amplifier incorporating voltage-series feedback. The feedback voltage is taken from the plate of the power pentode and fed through the secondary of the input transformer, placing the feedback effectively in series with the input signal induced in the input transformer secondary by the amp's input signal in the input transformer primary. The β factor can in this case be the voltage-division ratio of R_1 and R_2 : $\beta = R_1 / (R_1 + R_2)$.

In a typical case, the values of resistors R_1 and R_2 might be 2M Ω and 10M Ω , respectively, corresponding to $\beta=0.0476$. The open-loop voltage gain of the stage (using a 6BQ5) might be about 50. The stage gain would then be reduced by the feedback by the inverse of the factor $(1+A_{vol}\beta)$, which equates to about 0.3, yielding a closed-loop gain of 15.

The input impedance would be increased by the factor $(1+A_{vol}\beta)$, decreasing loading of the input transformer. The output impedance would



be decreased by the same factor as the gain, improving the damping factor. The combination of high input impedance drawing little current from the signal source and low output impedance making V_{OUT} relatively insensitive to load-resistance variations can be summarized by calling an amplifier using voltage-series negative feedback a voltage-controlled voltage source.

It is possible to connect the feedback signal in series with the input signal without using an input transformer, but that is seldom done.

Figure 1: There are four basic configurations for negative feedback: voltage series feedback (a); voltage-shunt feedback (b); current series feedback (c); and current-shunt feedback (d).

Voltage-Shunt Feedback

The voltage-shunt feedback amplifier is sometimes called a current-controlled voltage source. (Shunt is an old word meaning parallel.) In this type of circuit, since the input quantity is a current, and the output quantity is a voltage, the gain is expressed as a transresistance:

$$A_{ol} = \frac{V_{OUT}}{I_i}$$

(The prefix trans means that the voltage in the numerator and the current in the denominator are

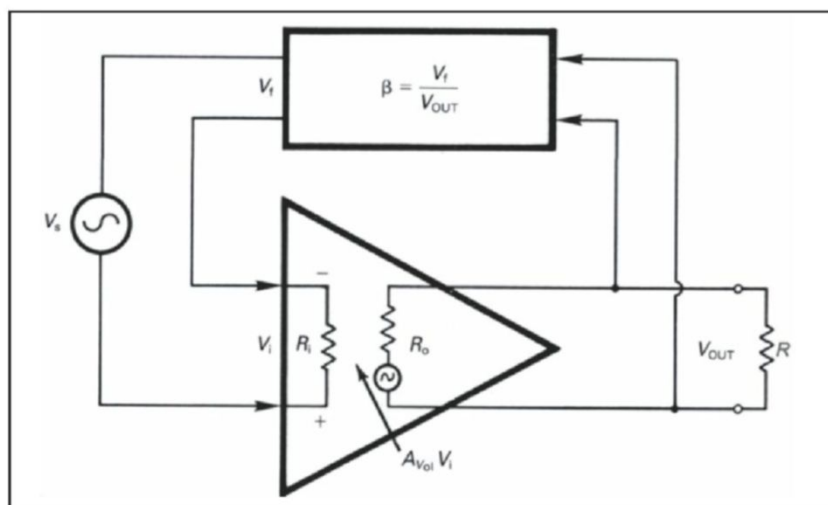


Figure 2: This enlarged schematic for a voltage-series-feedback stage clarifies some details.

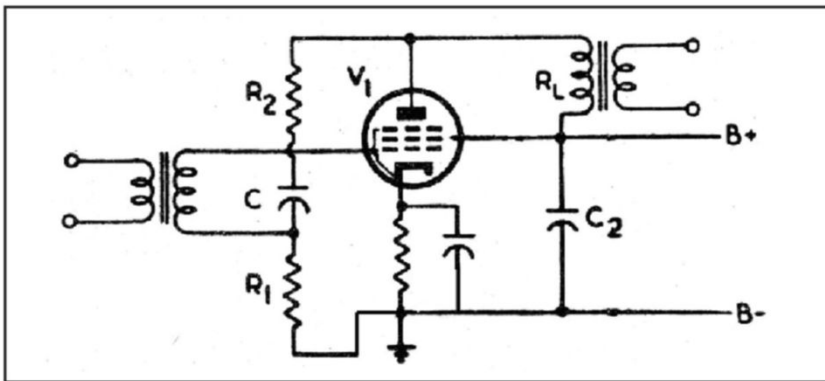


Figure 3: This single-stage hollow-state power amplifier uses voltage-series feedback.

measured at different places in the circuit. The terms in this and the following equations are as defined in **Figure 4.**) The closed-loop gain is given by the same equation that we obtained for the voltage-series circuit. The effect of voltage-shunt feedback upon input impedance is different from the effect of voltage-series feedback:

$$Z_{IN} = \frac{R_i}{1 + A_{ol}\beta}$$

and

$$Z_{OUT} = R_o (1 + A_{ol}\beta)$$

Figure 5 shows a triode stage using voltage-shunt feedback. The feedback voltage is taken from the triode's plate, then reduced in amplitude by a voltage divider comprised of 10MΩ feedback resistor R_f and 1MΩ grid resistor R_g . The open-loop voltage gain is given by:

$$A_{ol} = \frac{\mu R_L}{r_p + R_L}$$

where:

- μ is the amplification factor of the tube (100 in this case),

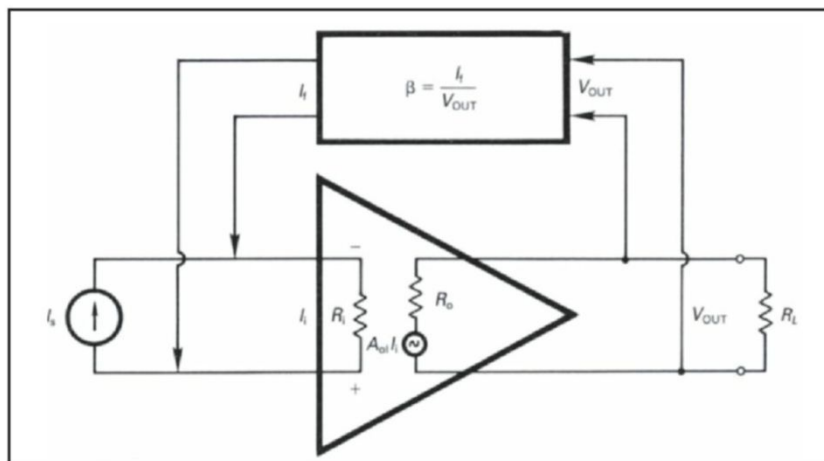


Figure 4: This schematic illustrates a generic voltage-shunt feedback stage.

- R_L is the load resistance driven by the stage (the 100kΩ plate resistor)
- r_p is the plate resistance of the tube (we'll use 62.5kΩ)

This gives a value of 61.53. Calculating using the voltage-divider equation, we find that $\beta = R_g / (R_f + R_g) = 0.091$. The resulting closed-loop gain comes out to 9.3. Using the value of R_g as R_i , the input resistance with no feedback, that value drops to 151.5kΩ with voltage-shunt negative feedback. The output resistance without feedback is the parallel combination of r_p and R_L , or 38462Ω. The voltage-shunt feedback increases this value to about 285kΩ. Of course, we have used about 16dB of feedback, and less feedback (lower value of β) would result in less decrease in gain and input impedance, and less increase in output impedance.

Current-Series Feedback

The current-series feedback circuit is a voltage-controlled current source. Therefore, its gain must be stated as a transconductance:

$$A_{ol} = \frac{I_{OUT}}{V_i}$$

Since the output voltage is just $I_{OUT} \times R_L$, we can multiply the transconductance by the load resistance to obtain the voltage gain. The generic schematic for an amplifier using current-series feedback is shown in **Figure 6**. This type of feedback decreases gain by the inverse of the factor $(1 + A_{ol}\beta)$. The input and output impedances are given by:

$$Z_{IN} = R_i (1 + A_{ol}\beta)$$

$$Z_O = (1 + A_{ol}\beta) R_o$$

The combination of a high input impedance and a high output impedance is characteristic of a transconductance amplifier, as this circuit is sometimes called.

A hollow-state version can be as simple as a pentode stage, although a pentode stage using a constant-current source as the plate load is more nearly ideal. **Figure 7** shows a common-cathode triode amplifier stage with an unbypassed cathode resistor. The feedback results from the tube's plate current passing through the cathode resistor, creating a feedback voltage proportional to that current, appearing in series with the input grid-to-cathode signal voltage. Thus, this is a current-series feedback circuit. The β factor is given by the equation $\beta = R_k / R_p$.

The usual configuration of this stage has the

lower end of R_G grounded. In this case, the input impedance of the stage is not affected by the feedback. But if the lower end of R_G is connected to the cathode instead of being grounded, the input impedance of the stage is increased as shown in the previous equation.

Sometimes part of the cathode resistor is bypassed, allowing the use of a larger resistor for biasing purposes while having less negative feedback than you would have if you left the entire cathode resistor unbypassed.

When the output is taken from the cathode rather than the plate, this stage is called a cathode-follower. Then $\beta=1$, and the voltage gain is just under unity. The output impedance of a cathode-follower is given by: $Z_{OUT}=R_K/(1+g_m R_K)$, g_m being the transconductance of the triode, given by μ/r_p .

Current-Shunt Feedback

The current-shunt feedback amplifier is a current-controlled current source or, in other words, a current amplifier:

$$A_I = \frac{I_{OUT}}{I_s}$$

The closed-loop current gain is given by:

$$A_{cl} = \frac{A_{ol}}{(1 + \beta A_{ol})}$$

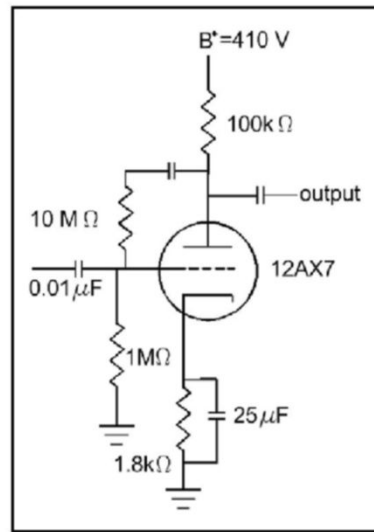


Figure 5: This hollow-state triode amplifier stage uses voltage-shunt feedback.

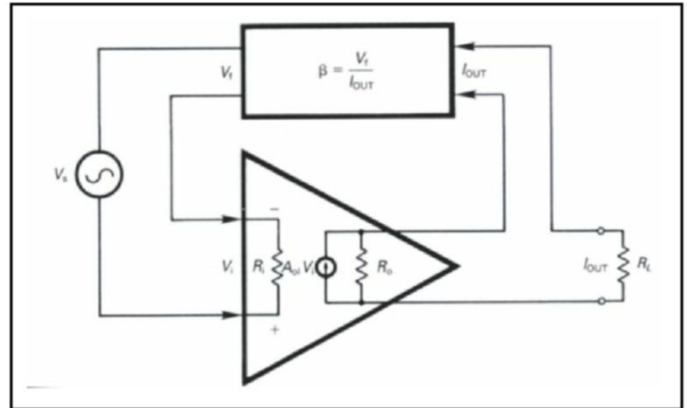


Figure 6: A generic current-series feedback stage is represented by this schematic.

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 203-502-7600

IsoThermal

TUBE TRAP DESIGN KIT #2

A: TubeTrap: Start with an Acoustic RLC Series Parallel Circuit

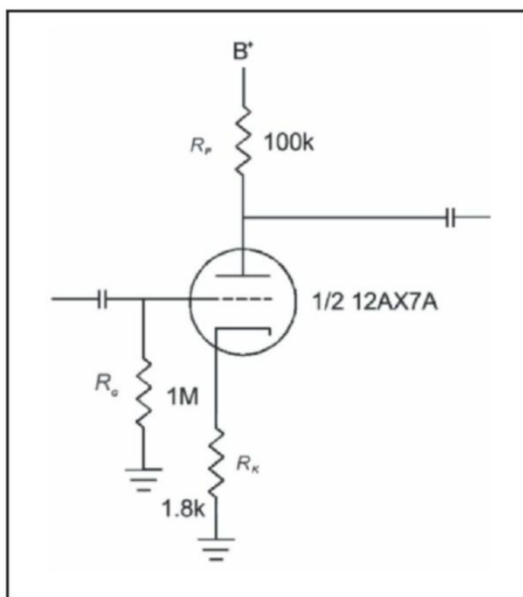
B: IsoThermal Upgrade: Add IsoThermal Matrix

C: TubeTrap + IsoThermal = IsoThermal TubeTrap

Save 10% on IsoThermals Use Coupon Code: **audioXpress10**

Acoustic Sciences Corporation | shop.acousticsscience.com

Figure 7: A common-cathode triode stage with an unbypassed cathode resistor uses current-series feedback.



The input and output impedances with feedback are given by:

$$Z_{IN} = \frac{R_i}{(1 + A_{ol}\beta)}$$

and

$$Z_{OUT} = R_o (1 + A_{ol}\beta)$$

It would be possible to connect a small resistor in series with the primary of an output transformer, between the top end of the primary and the B+ supply, then feed the signal from the lower end of the resistor through a blocking capacitor to the grid of a triode, and thus create a hollow-state stage using current-shunt feedback, but the result would have a low input impedance and a high output impedance (thus a low damping factor, for a power stage). Such a stage would not be very useful, so you are not likely to find such stages in practice.

About the Author

Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built up a "junk box" by disassembling trade-in TVs from his uncle's furniture store. His dad bought him the Van Valkenburg, Neville, and Nooger Basic Electronics book set, so he learned about tube circuits as well as transistors. He started repairing electronic devices about 1965, earning his First-Class Commercial FCC license in 1969. He has worked part-time in broadcast radio engineering and audio electronics repair since 1968, in addition to full-time work in acoustical and audio system design, plus a 20-year stint teaching electronics at the college level.



Summary

In summary, any amplifier having voltage feedback acts as a controlled voltage source, and has an output impedance that is reduced by the feedback. This makes the amplifier into a more perfect voltage source. Any amplifier having current feedback acts as a controlled current source, and its output impedance is increased by the feedback. This makes it act as a more perfect current source.

Series feedback causes the amplifier to have a higher input impedance, which is desirable if the input is fed by a voltage source. Shunt feedback causes the amplifier to have a lower input impedance, which is desirable if the input is fed by a current source. All varieties of negative feedback reduce the gain by the same factor $(1 + A_{ol}\beta)$.

Other Effects of Negative Feedback

Aside from reducing gain and changing impedances, negative feedback has three other important effects: (1) it improves gain stability; (2) it reduces distortion; and (3) it increases bandwidth. Gain stability is a measure of the extent to which the gain of a circuit remains at the design value in spite of temperature- and age-induced variations in components.

Negative feedback makes the gain of a circuit more dependent upon the values of the passive components in the feedback (β) network, and less dependent upon the active devices (transistors, ICs, etc.). Since passive devices are usually much more stable than active ones, the net effect is a great improvement in the gain stability as a result of the addition of negative feedback.

Strictly, the term distortion refers to any difference, other than amplitude, between the input and output signals of a circuit. There are two general kinds of distortion: nonlinear distortion and frequency distortion. Nonlinear distortion results whenever the output voltage of a circuit is not strictly proportional to the input voltage. The result is a change in the waveform of any signal. Frequency distortion refers to the inability of a circuit to equally handle all frequencies that are in the input signal. Thus, the output signal contains proportionally more or less of some frequencies than others.

In common usage, distortion is taken to mean only nonlinear distortion. However, it is a fact that both kinds of distortion are reduced by the addition of negative feedback. The effects of different degrees of negative feedback upon distortion and bandwidth will be the subject of a future Hollow-State Electronics article. 



ELECTRONIC TRANSFORMERS

- Original Equipment
- Replacement
- Amateur

TRIAD MAGNETICS, INCORPORATED

460 HARLEY KNOX BLVD., PERRIS, CA 92571

www.TriadMagnetics.com

Like. No. Other.

EPIQUE

by Dayton Audio

Presenting the Epique
E150HE-44 and **E180HE-44**
ultra-low distortion drivers
from Dayton Audio.

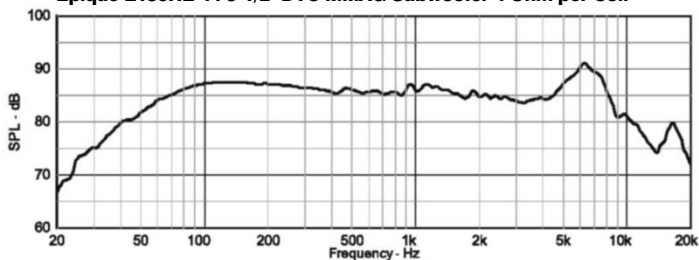
Capable of unheard-of excursion for their size, these drivers are engineered to produce exceedingly low distortion with controlled response. Allowing more versatility than ever, you now have the opportunity to create systems that will deliver game changing performance.



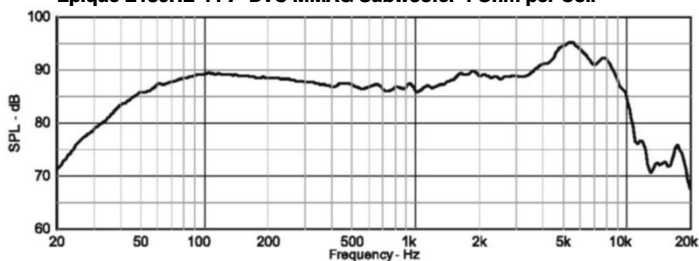
Patented MMAG motor system

- Patented dual gap motor design allows exceptional linearity
- Over 14 mm of excursion (at 70% BI), 28 mm peak to peak
- Damped woven carbon fiber cone delivers remarkably smooth response
- Inductance control for extremely low distortion

Epique E150HE-44 5-1/2" DVC MMAG Subwoofer 4 Ohm per Coil



Epique E180HE-44 7" DVC MMAG Subwoofer 4 Ohm per Coil



DAYTONAUDIO
daytonaudio.com

Distributed by:
PARTS EXPRESS
AUDIO COMPONENTS & SOLUTIONS
parts-express.com
1-800-338-0531

Custom OEM inquiries welcome