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Fresh From the Bench
Sonarworks Reference 4
Studio Edition Loudspeaker
and Headphone
Correction System
By Stuart Yaniger



Show Report
AXPONA 2018
The Renaissance of High-Quality Sounds
By Oliver A. Masciarotte

Audio Electronics
Current Feedback:
Fake News or the Real Deal?
By Christopher Paul

Sound Control
Microphone Choices
for Solo Voices and Instruments
By Richard Honeycutt

Questions & Answers
Thinking About
Creative Sound Solutions
By Shannon Becker

Audio Electronics
Transformers for
Tube Amplifiers (Part 3)
By Gerhard Haas

Speaker Builder
Inductive, Capacitive,
and Resistive Back EMF
By Andy Lewis

Hollow-State Electronics
Design and Performance
of Power Supplies
By Richard Honeycutt



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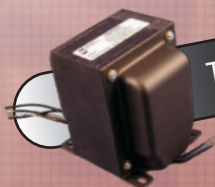
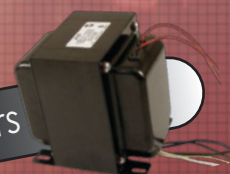


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The Sound Reference We Want?

While audio product designers see the use of digital signal processing technologies applied to headphones and speakers as a valuable tool to correct transducers nonlinearities, and as a way to fine-tune frequency response according to an envisaged target or a specific voicing, the amount of variability left to the user is a matter of great debate. In general, “personalization”—currently one of the strongest industry trends—is not well regarded by those designers and manufacturers who believe that a listening experience should always be consistent and the closest to the intended reference, avoiding the dreaded “circle of confusion.”

Current industry trends and technology progress reveal we may have reached a critical point in addressing those goals, introducing new challenges. As Floyd Toole described, the goal should be “to design the best possible listening experience for recording control rooms and home entertainment systems,” considering the whole sound reproduction chain, including room acoustics. And in the case of headphones, trying to make the listening experience as close to that “best possible” room-listening experience.

Unfortunately, achieving linear reference monitoring for audio professionals and studios, or musically fulfilling experiences and improved definition translates to vastly different realities, depending of the usage profiles and when addressing consumers expectations in different markets, even when—as the recent Harman research clearly shows—the goals shouldn’t differ that much.

The available technologies to address those targets remain quite consistent in the approach, enabling some companies to explore the possibilities to the extreme.

Such is the case with the Latvia company Sonarworks that we address in this edition of *audioXpress* with its Reference 4 Studio Monitor and Headphone Calibration Solution, targeting the needs of audio professionals. Sonarworks combined patent-pending technology and the expertise of their own music production “expert ears” to develop a consistent and practical calibration software for studio monitors and selected headphones, providing a reproduction that is closer to a studio reference. This also enables audio professionals to obtain greater consistency when working in different studios.

The same combination of computer science and psychoacoustics also allows Sonarworks to simulate different audio systems with high accuracy—which professionals use to check how their mixes will translate to other systems.

More recently, Sonarworks released True-Fi, a new software targeting consumers, essentially intended to remove unwanted sound coloration from headphones so users can have “a sonically balanced listening experience.” But the new True-Fi Headphone app can also be used by consumers to emulate the response of multiple target headphones—meaning that an affordable but good quality headphone will sound exactly as another potentially much more expensive model. We tried it, and it works, even if—as recent research published in the *Journal of the Acoustical Society of America* also shows—trying to emulate a certain frequency response curve and apply correction of tonal discrepancies caused by the headphones physical construction, could also reveal the limitations of the actual transducers, particularly in terms of audible distortion above certain thresholds, depending on the playback levels.

A similar emulation approach is used by Harman on its Harman Headphone Virtualizer App to enable listeners to make double-blind comparisons of different headphones, using a single high-quality replicator headphone.

Of course, emulation limitations also apply to personalization features, where the end-user is able to dial any type of frequency profile, potentially revealing driver limitations or simply conflicting with the device’s own design (multiple drivers, crossovers, size of transducer, earcup form factors, etc.). So, basically this is the same as saying, yes you can dial in a certain amount of bass, mid, and treble, but if you really want to have any type of response emulation, you should make certain we have a good sounding system to start with. And if our system is “good sounding” we shouldn’t have much reason to change it anyway—unless we just want to compensate for hearing deficiencies (that we are aware of).

Many companies are now betting on delivering individual profiles based on personal assessments of the users’ hearing. But we also need actual research to find out, how different the thresholds are for those profiles. The next challenge is to design equipment able to deliver the widest variability in emulation and personalization, paradoxically potentially leading users to a consistent experience.

J. Martins
Editor-in-Chief

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Avel Lindberg, Inc.	63
Celestion	11
Hafler	23
Hammond Manufacturing, Ltd.	3
Hypex Electronics BV	17
JLI Electronics, Inc.	53
KAB Electro-Acoustics	41
Marchand Electronics, Inc.	55
Menlo Scientific, Ltd.	37

OPPO Digital, Inc.	15
Parts ConneXion	2
Parts Express International, Inc.	68
Pimacoustic	23
Schiit Audio	51
Sonarworks	59
Sonoma Acoustics	21
The Loudspeaker Sourcing Show 2018	65
Triad Magnetics	67
Warkwyn	41

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COLUMNISTS

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

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

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

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

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

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● Features

8 AXPONA 2018

The Renaissance of High-Quality Sounds

By Oliver A. Masciarotte

Oliver Masciarotte provides an insider's view of Audio Expo North America (AXPONA). The three-day event featured listening rooms, the Exhibit Hall with the Ear Gear Expo and the Record Fair, several seminars, and live musical performances.

18 Back EMF Phase Relationships in Moving-Coil Loudspeakers (Part 3) "Slip" Creates Inductive, Capacitive, or Resistive Back EMF

By Andy Lewis

In the third part of this article series, Andy Lewis discusses the concept of slip in order to explain its effects on impedance.

30 Sonarworks Reference 4 Studio Edition Loudspeaker and Headphone Correction System

By Stuart Yaniger

Stuart Yaniger examines the Reference 4 Studio Edition, targeted to studio and mastering sound calibration but can also be used in home audio setups.

42 Current Feedback: Fake News or the Real Deal? (Part 1)

Untangling the Signal Path

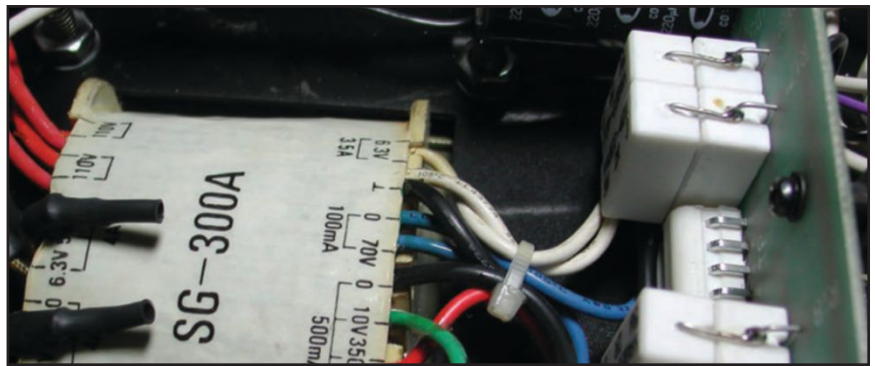
By Christopher Paul

This article employs bench tests, simulations, and equations derived from various circuits to test the existence of current feedback.

48 Transformers for Tube Amplifiers (Part 3) Choosing the Right Transformer

By Gerhard Haas

In this final article of the series, we will discuss long-term stability and how to select the right transformer.



● Columns

SOUND CONTROL

24 Miking Solo Voices and Instruments

The Mic Choices

By Richard Honeycutt

QUESTIONS & ANSWERS

38 Breathing New Life Into a DIY-Oriented Audio Company

An Interview with Kerry Armes and Dan Poinsett of Creative Sound Solutions

By Shannon Becker

HOLLOW-STATE ELECTRONICS

60 Design and Performance of Power Supplies for Hollow-State Amplifiers

By Richard Honeycutt

● Departments

4 From the Editor's Desk

5 Client Index

66 Industry Calendar

● Websites

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AXPONA 2018

The Renaissance of High-Quality Sounds

Audio Expo North America (AXPONA) is the largest high-end audio show in North America. The three-day experience (April 13–15, 2018) featured multiple hotel floors packed with listening rooms, The Exhibit Hall featuring Ear Gear Expo and The Record Fair, seminars, and live musical performances.

By
Oliver A. Masciarotte
(United States)

This year's AXPONA show was held in a new venue, which received mostly positive responses from attendees and exhibitors. The Schaumburg Convention Center and attached Renaissance hotel offered plenty of good-sized rooms, decent amenities, and a dated but pleasing vibe somewhat reminiscent of Barbarella.

The show was well attended, with a record-breaking 8,134 attendees—up 21% over 2017. This year was a bit of a change for me as well, deciding to drive with an audiophile buddy rather than my better half, who couldn't get time off. Though attendance was up, it was the rare room that was too crowded to enter. Most vendors experienced good traffic and found the rooms to be workable. Rooms around the atrium interior had a bonus view, open and airy, when you left those spaces.

Unlike years past, I didn't spend a great deal of my time in the headphone area, instead concentrating on the more than 100 exhibitors I had preselected. Despite that, I missed quite a few due to very heavy snow and winds to the west, which forced us to head home on the wreck-strewn interstate a day early.

Electronics

On to the show! I'll start my coverage with electronics, specifically Cambridge Audio (www.cambridgeaudio.com). You may remember my 2017 Rocky Mountain show coverage, where I mentioned that the company had been working on a new range of higher end electronics, which sounded very pleasing. "With a minimal signal path, intelligent biasing and no caps to degrade the audio," I stated that the prototypes was a departure from their normally budget priced offerings. Those proof-of concept separates have matured into the new Edge series, celebrating 50 years in business with a clean, aesthetically pleasing \$5,000 Edge A integrated, the well-appointed \$4,000 Edge NQ pre/streamer, and the \$3,000 Edge W power amp. Part of what makes the DC-coupled amp and preamp sound good methinks is their ultra-wide bandwidth, with a published frequency response of 3 Hz to 80 kHz ± 1 dB. Rupert Neve would be proud.

Ayon Audio (www.ayonaudio.com) was showing off its new \$28,800 hollow-state Orthos II XS Gen 4 in an all-Ayon stack driving \$49,900 Lumenwhite Kyara

Atria, a sure cure for claustrophobia

speakers. The mighty Orthos II XS yields 300 Class A W into 8 Ω in pentode mode, and 180 W in triode mode.

Axiss Audio (www.axissaudio.com) was showing Resolution Audio's 3.0 version of their \$9,500 Cantata Music Center for the first time, with an upgraded R-2R ladder DAC and analog attenuator (www.resolutionaudio.com). It also includes a pair of line level stereo analog inputs and a Red Book-only optical disc transport. Due to the proprietary DAC architecture, the USB input is limited to 192 kHz.

In another Axiss room, I had a nice visit with the A&M Ltd.



Cambridge Audio's new Edge W employs modified A/B operation for clean and efficient power...good looking as well!



Dual Orthos II XS amps heat up the space.



The newly revised Cantata Music Center 3.0, stacked on a Cantata C50 2.0 integrated amp

dudes while company Director Yutaka Miura ran me through the specifics of their new AirTight ATE-3011 phono preamp. Hewing to the company's more than 30 years of high-end hollow-state heritage (www.airtight-anm.com), Miura-san told me the ATE-3011 features three phono inputs and a solid copper chassis. Retro illuminated independent selection of low "turnover" and high frequency "rolloff" de-emphasis, including NAB, RIAA, FFRR, AES, and flat settings, covers current and many legacy standards. The model ATE-3011 will arrive at the end of the year, in time for CES, and pricing has yet to be set.

Ozan Turan, President of High End By Oz (www.highendbyoz.com), had the Vitus Audio Reference Series RI 101 integrated amp with optional DAC on demo (www.vitusaudio.com). The new \$15,800 RI 101 boasts an entirely new preamp stage, and its output boards, along with their initial PSU, are completely redesigned for even better performance. With crazy wide frequency response, way past 700 kHz, this brawny beast outputs 300 W of Class A/B power into 8 Ω and 600 W into 4 Ω .

Due to their lack of euphonic color, low output impedance, wide bandwidth, and moderate pricing, Linear Tube Audio had its gear in several suites at the show (www.lineartubeaudio.com). This visit, I was focused on the new \$6,800 ZOTL Ultralinear. As with previous ZOTL designs, the Ultralinear eschews an output transformer, instead relying on an RF carrier. A review of the ZOTL Ultralinear is in process, so stay tuned!

With a prototype having been shown at last year's Rocky Mountain Audio Fest, Classic Tube Audio was proud of a production example of its 2A3-based, \$4,900 Whammerdyne DAA-3 (whammerdyne.com). A two chassis configuration, with a PSU base "docked" to the amplifier above, the DAA-3 flaunts DC coupling, wideband operation (-1 dB from 12 Hz to 95 kHz,



AirTight's ATE-301, a final product candidate in Golden Age champagne cosmetics



The understated new Vitus Audio Reference Series RI 101

-3 dB from 6 Hz to 130 kHz) and a suggestively DIY aesthetic for audiophiles with limited means and high-end taste.

A real hollow-state blast from the past is Western Electric, and its revival of the brand as well as WE 300B manufacturing (www.westernelectric.com). They brought along an interesting prototype to the show, a Class A2 single-ended integrated



Linear Tube Audio's new ZOTL Ultralinear, in austere Fern & Roby-designed casework



The DAA-3's separate PSU boasts seven low-noise regulated DC supplies.



The homely but promising 91E prototype from Western Electric

amplifier with an MC/MM phono stage, no negative feedback, and microprocessor-controlled bias. The production 91E will run \$9,995 and makes "20+" watts of yummy, dual triode goodness.

The irrepressible Richard Schram was chillin' at AXPONA, attending the birth of his new \$5,995 Halo JC 5 stereo amp—essentially two Halo JC 1 monoblocks in one classy chassis. Designed by analog yoda John Curl, the JC 5 provides 400 W into 8 Ω of direct coupled Class AB power, with the first 12 being Class A (www.parasound.com).

A vendor I always seek out for his two-way, stand mounters is John "Fritz" Heiler (www.fritzspeakers.com). He took two new, all Scan-Speak models to the show for high-end performance at a nice price. The updated \$2,800/pair REV 7 SE includes a 7" Revelator mid/bass driver and Illuminator ring tweeter, while his \$3,800 LS 7 model is equipped with a 7" Illuminator mid/bass driver and the top-of-the-line Revelator tweeter. He mentioned that both models are 3 dB down at 30 kHz, and employ an Acoustic Reality-style capacitor and resistor-free series crossover.

Trade show trooper, longtime friend, and HRA evangelist Mark Waldrep was in residence at the AXPONA Expo, and when I stopped in passing, he said "You gotta hear this," motioning to a small soundbar on his table. Not just any soundbar, it was Comhear's \$389 to \$599 YARRA 3DX speaker array (www.yarra3dx.com), with MyBeam beam forming technology. Radio frequency beam forming has been around for many decades, especially in radar



The beefy JC 5 in black livery from Parasound



Fritz's new LS 7, at left, and REV 7 SE, at right



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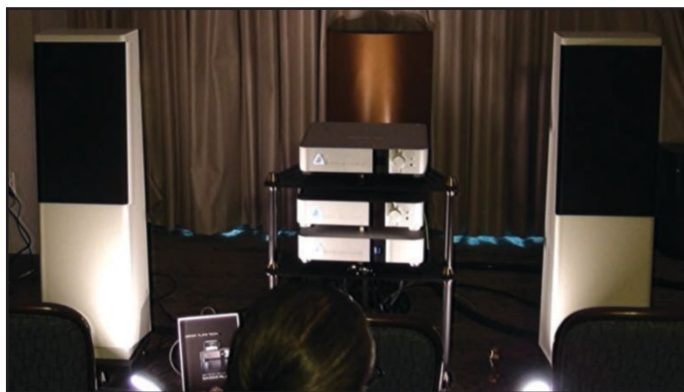
CELESTION

applications, but its use in audio has been mostly restricted to sound reinforcement. As I dropped into the YARRA 3DX sweet spot seat, I couldn't help noticing that the entire soundstage actually swung down into view from stage left. This sensation, akin to entering a stage from below, was totally unlike any I had experienced. The bar provided solid imaging that extended well outside the physical speakers. I noticed what may have been timbral issues with the unfamiliar music, but that might have been due to not enlisting their mobile app for customization.

I also visited the folks from Merging (<http://nadac.merging.com>), as we have a long relationship related to my day job at a pro audio software manufacturer. They had a sweet setup going, with PSI Audio prototype loudspeakers resting on Star Sound Technologies BSPL101 platforms (www.starsoundtechnologies.com). The floorstander speakers, still unnamed, were based on PSI's existing A25-M stand-mounted three ways with a 10" woofer. In keeping with the PSI philosophy (www.psiaudio.swiss), the active design is all-analog, with a Class AB amp for the tweeter and dual Class G/H amplifiers for everything else. Roger Roschnik, CEO of PSI, filled me in via e-mail after the show. "Phase compensation, adaptive output impedance damping, all entirely analog in to out..." he said. "Entirely handmade, tuned, and calibrated in our large anechoic chamber. The idea is to offer the best of professional monitors for domestic use."

Speaking of DSP, Gayle Sanders premiered a modern loudspeaker, comprised of a hand-sculpted, multi-way floorstander with constrained-layer damping, an integral electronics package, and lots of DSP (www.eikonaudio.com). The Image1 active speaker system includes Bernt Böhmer's (www.bohmernautio.com) wavelet-based Eikontrol to provide internal temporal and frequency compensation within the speaker, and externally for boundary compensation and uniform wavefront propagation into the room. From my brief listening session with unfamiliar material, I assert that the Image1 provided what could be the most uniform low-frequency coverage of any rig I've heard. The complete Image1 system includes paired speakers, amplifiers, and a mobile app-controlled Eikontrol for \$24,500.

Vehement Audio (www.vehementaudio.com) had its killer little \$2,495 Brezza EOS on demo. A rear-ported bookshelf model,



These still unnamed towers are PSI Audio's first high-end consumer product, reflecting the 40 years of pro audio analog design of the Swiss company.

the freshest Brezza sports a RAAL 64-10 for the very top and a custom 6" SEAS mid/woofer crossing over at 3.8 kHz. The acrylic front baffle could contribute to the 30 lb. weight but, whatever the magic employed, they sounded exceptionally nice.

Another vendor I always enjoy, for value and fidelity, is Ryan Speakers (www.ryanspeakers.com). A preview of the upcoming S610 was on hand, a time-aligned two-way with an upper 3 dB down at 35 kHz. A 1" beryllium dome crosses over at 2 kHz to a 6.5" laminated Nomex mid/woofer. A port on the front of this \$3,995 bookshelf makes for easy placement.

Do-it-all techie Steven Norber (www.pranafidelity.com) was at AXPONA with his \$4,950 per pair Bhava. Teamed up with his own purna/ca preamp (\$4,500 to \$9,950) and \$8,950 purna/ma 1200 W into 2 Ω power amp wired with Kubala-Sosna Elation! cabling, the sound was predictably reference quality. "Aside from their shape, the materials for the cabinets are different...along with changes to the crossover," he said. There were also changes made to the internal acoustic treatment. "It may be worth noting the Fifty90 cabinets were produced in China, while the Bhavas are built and



Electrostat pioneer Gayle Sanders is back with Eikon Audio.



Vehement Audio's little champ, the Brezza EOS

finished locally (in Colorado) using materials unavailable in China.” The result? In Norber’s words, an even more “...refined and focused sound without giving up dynamics or sounding too ‘polite.’”

Neat Acoustics (www.neatacoustics.com) had a new jumbo Iota on show for the first time, a production version of the \$4,995 Iota XPLOER . While I use jumbo in a relative sense, the XPLOER is a larger version of the ALPHA. The latest Iota retains a 2.5-way design, with a 170 mm mid/bass unit and an AMT tweeter in a sealed enclosure. The lower cabinet section contains a separate “iso-baric,” downward-facing dual magnet-to-cone woofer configuration, tuned to 35 Hz. Frequency response for the XPLOER is roughly 25 Hz to 25 kHz, and the sensitivity is 88 dB at 1 W.

At any show, I always find at least one loudspeaker vendor that is new to me, and this year’s example is Orinda Acoustics (www.orindaacoustics.com). Their enticingly named new “12” 2-Way Be” model is a two-way floorstander, with a beryllium compression



Ryan’s new raked cabinet two-way, the S610, will arrive this summer.



PranaFidelity’s compelling Bhava, a “step-up” from his Fifty90

driver (1” exit) fitted to a 300 mm symmetrical horn. A 12-inch long excursion, paper cone woofer in a vented cabinet takes care of the low-frequency duties. Frequency response is roughly 25 Hz to 25 kHz, and the price is \$22,000/pair.

You may have seen recent coverage of Scan-Speak’s new Ellipticor line in this magazine as well as *Voice Coil* (April 2018 Test Bench—“The New Ellipticor Drivers from Scan-Speak”). I can tell you that seeing an elliptical voice coil up close reminded me of those less than ideal race track speakers you typically find in car doors. Of course, one of the design goals for such a unique motor was reducing modal resonances. In talking with Scan-Speak’s Fran Hasselriis about these rare beasts, he said it was a tough manufacturing challenge. The hardest part of bringing the line to life was proving to themselves that they could do it (www.scan-speak.dk).

Hasselriis also tipped me off to a customer of his that was new to me; Acoustique Quality (www.aq.cz). Though they have a range of high-end ported hi-fi speakers, the Passion line, their latest is an unassuming desktop active speaker, the M24BT, making its North American debut at the show. The price is yet to be determined, but the wall wart-powered M24BT includes Bluetooth 3.0, two 40 W Class D amps, and a remote.

Ear Gear Vendors

João Martins, the hirsute editor of *audioXpress*, asked me to minimize my coverage of headphone happenings, as they’re taken care of elsewhere. While staying within that guideline, I did take a good look around, and found several Ear Gear vendors sequestered in a light-filled mezzanine suite, including Woo Audio (www.woaudio.com).

With electrostats making something of a comeback, Woo was showing its latest, the lovely \$8,999 3ES, an all-balanced headphone preamp/amp/energizer with a mad scientist vibe. Based on the



The guts of Ellipticor



The headphone suite (Photo courtesy of Michael Newman)

W33, the 3ES delivers up to 10 W of wideband power into Stax, MrSpeakers, and HiFiMAN cans.

New Gear

Another fave cans vendor is Dan Wiggins and his Periodic Audio gang (www.periodicaudio.com). This AXPONA hosted the first glimpse of their upcoming amp, the “Ni” as in Nickel, for all you science nerds. The forthcoming self-powered miniature headphone amp has a modest gain of 3 dB to accommodate a wide range of impedances without noise, and will go on sale for \$299.

Doshi Audio (www.doshiaudio.com) had two new items, a stereo amp and a headphone amp. Based on its Monoblock, the \$18,995 Stereo Amplifier is a hybrid, with JFET input and paired EL34/6CA7 output and low overall feedback. The result is 50 W of Class A power, and a maximum of 65 W. The imaginatively named Headphone Amplifier features a completely balanced design with both balanced and unbalanced outputs, three impedance ranges and direct coupling throughout.

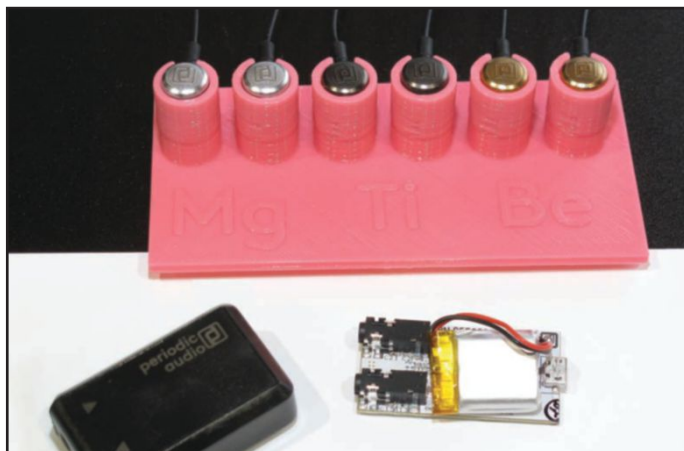
Mark Gurvey of Source Systems (www.sourcesystemsllc.com) had a new big brother to M2Tech’s EVO mini-stack (www.m2tech.biz). The Rockstars Series is composed of the \$1,699 Young MkIII DAC/preamp, the Nash MM/MC phono preamp, the DXD-capable \$1,899 Joplin MkII ADC, the \$1,299 Crosby Class-D power amp, and the

Van Der Graaf MkII low-noise PSU, also \$1,299. Pricing is not yet set on the Nash. The Joplin MkII offers variable gain with choice of digital de-emphasis and H/LPF. The revised Young handles DXD and DSD4, as well as analog and Bluetooth in. Though Der Bingle wasn’t a rock star, the Crosby may just be, packing 110 W (4 Ω) of power into the same compact form factor as the rest of the line. Adding to the allure, the Van Der Graaf MkII acts as a power hub for the whole system, and features user-defined power up/down sequencing. Gurvey also had fidata’s HFA51-XS20U high-end music server, an upgrade to the XS10U, which now supports four RAIDed 1 TB SSDs. The pairs of striped drives, which optionally can be mirrored as well for increased data integrity, reduce current draw on the PSU. In turn, this improves fidelity as any server designer knows all too well. The HFA51-XS20U runs \$7,000 to \$9,000, depending on storage complement (www.iodata.jp/fidata).

Duane Randleman of Automated Lifestyles brought along two entries in the new gear category; Bionic Cable and Aesthix’s first



Woo Audio’s (ahem) heady new 3ES



A prototype Nickel case and quad-layer PCB, accompanied by Periodic’s trio of insert-style cans



Doshi’s handsome Headphone Amplifier is priced at \$3,795.



The Rockstars of M2Tech: Young MkIII, Crosby, and Van Der Graaf MkII

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integrated amp. Developing 150 W into 8 Ω , the \$7,000 hybrid Aesthix Mimas combines elements from its award-winning Calypso line stage and Atlas power amp, and includes optional phono board and DAC. Internals are truly balanced and differential, and a 1 W into 32 Ω fully discrete, Class A headphone amp option replaces the default 1/3 W A/B version (www.aesthetix.net). Bionic Cable is a new company and this was the first show for that line (www.automatedlifestyles.tv).

There are plenty of commodity Ethernet switches available that support basic Quality-of-Service (QoS). SotM (www.sotm-audio.com) has gone one better with its sNH-10G, designed with audio in mind. This smart switch folds SotM's noise filtering and isolation tech from its \$700 iSO-CAT6 Special Edition ethernet filter/isolator, and includes a house clock output for connection to its master clocks, such as the new \$4,000 sCLK-OCX10, keeping all participants in perfect frame sync. The model also has two fibre ports for very high speed, long throw LAN connectivity. Pricing on the sNH-10G is yet to be set.

Vinyl

I don't subscribe to the Vinyl Revival, having had a turntable, originally an AR-XB, since college. A little over a decade later, I was introduced to what we now call HRA digital, and have never looked back. I still own a Pioneer PL-12D, but never use it 'cuz now I have FiDef but that's another matter. Anyway, there were two turntables at the show that caught my attention—one for its subtle tastefulness, and the other for its novel approach. Let's start with EINSTEIN Audio Components (www.einstein-audio.de) and its The Record Player. At \$12,000, this base isn't for the faint of heart. What struck me about this lovely piece was the lack of Trumpian ornamentation, a study in understated elegance. In high-end audio, nothing succeeds like excess, and The Record Player bucks that thinking.

Now for the novelty...Fern & Roby (www.fernandrobby.com), crafters of wood and iron, have a new product. The \$6,750 Montrose

Kit consists of the Montrose turntable, a Maverick phono preamp with either MM or MC daughterboard, a Denon 103r cartridge, an alignment protractor, and four isolation feet. What I found fascinating was the turntable's gray cast iron makeup, inherently massive and resonant-resistant.

The Widget

Only one widget in this year's report, the unmistakable Pnepod NP-1 from Pneumance Audio (www.pneumance.com). Employing a tried and true technique, the Pnepod consists of a synthetic rubber dome captured by a metal collar with valve. Hand-pressurizing the dome provides a pneumatic lift for any platform the Pnepods sit under. A set of four costs \$1,195, and that kit will support up to 80 lb of distributed weight.

Overall Impressions

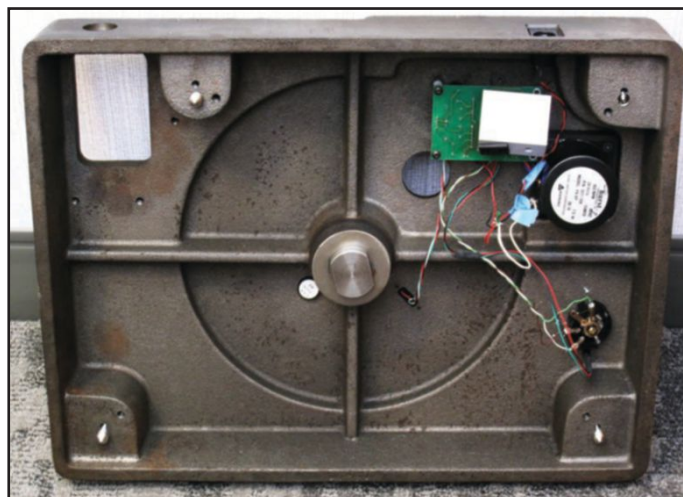
Several tendencies were evident at this year's show. First, Master Quality Authenticated (MQA): Notice that this is my only mention of the emerging format. Why? Because, at least for mainstream higher fidelity manufacturers, inclusion of MQA support is becoming a given. The new Young MkIII is an example. Look, despite what conspiracy theorists and non-oversampling DAC manufacturers may say, MQA is a great way for record labels to distribute high-resolution audio (HRA) that's not the master file. With forward and backward compatibility built in and sound that's way better than Red Book, what's not to like?

Next factoid of note was my repeated encounter with electronics that exhibit much wider frequency responses than the old school "20 Hz to 20 kHz" dictum. I mentioned pro audio legend Rupert Neve earlier...Neve holds to the sensible notion that, in order to have clean phase behavior in the audio passband, you have to design for at least two times that bandwidth. Seems that manufacturers are increasingly following that path—an outstanding example being the pre-production L2 Signature monoblock that Vinnie Rossi (www.vinnierossi.com) showed, with 250 kHz bandwidth.

Having worked with what is now "HRA" since the late '80s, I'm loving it. I asked Nick Doshi of Doshi Audio, "Is ultra-wideband



May Park shows off SotM's new sNH-10G eight port switch.



Fern & Roby's cast iron plinth



Pneunce Audio's Pneupod

a fad or simply good design sense?". He stressed that frequency extension alone is not enough, and cautioned that "...an amplifier with a wide bandwidth is also more susceptible to Radio Frequency Interference (RFI). Its design must include adequate RFI protection in our increasingly (RFI-) polluted world." Doshi also added that a "...phase margin for stability into less than ideal loads and low output impedance for control over tough loads" are also recommended. His new Doshi Amplifier exhibits frequency response within 1 dB from 20 Hz to 100 kHz and the all-tube monoblock is within 1 dB from approximately 10 Hz to 75 kHz with -3 dB down at 117 kHz.

Last item...After-hours entertainment at this year's AXPONA was decidedly not old skool, with one of them being a Bowie tribute rather than lounge lizard jazzers. This skews toward the third, and perhaps most significant trend for the audiophile industry—the ever larger participation of youth at the show, especially women. Since the saner sex has statistically better hearing than men, often accompanied by a more refined design sense and an expanding disposable income for singles, it's no wonder women are increasingly drawn to high-end hi-fi.

Overall, I find all these developments refreshing, and look forward to acceleration in all those directions. I can also look forward to future AXPONA shows, now that they've settled on a winning venue. 

About the Author

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

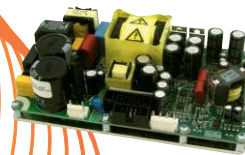
hypex NCxxxMP series

overview

The NCxxxMP amplifier module incorporates a low power standby power supply (meets 2013 ERP Lot 6 0.5W requirements), a highly efficient switch mode power supply and a high- performance Class D amplifier in one compact and easily applicable power brick.



NC122MP



NC250MP



NC252MP



NC500MP



NC502MP

Add-on Module NC100HF
The NC100HF is a dedicated tweeter-amplifier which fits the NCxxxMP series.



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Back EMF Phase Relationships in Moving-Coil Loudspeakers (Part 3)

"Slip" Creates Inductive, Capacitive, or Resistive Back EMF

In the third article in this series, we discuss the concept of slip, in order to explain how and why it causes the impedance to be inductive, resistive, or capacitive, depending on frequency.



By
Andy Lewis
(Acme Sound, LLC)

Due to inertia, there is a delay in time between the motor force driving the cone (in phase with current) and the velocity of the coil as it moves through the magnetic field. This creates a phase delay between the input signal and the Back EMF, which is in phase with velocity ($V = Blv$). This phase delay is what causes Back EMF to simulate inductors, capacitors, and resistors. For the purposes at hand, we will refer to this delay in time between input current and cone position as "slip."

There is no established convention for the use of the term "slip." I have seen it used to describe the phase relationship between input voltage and Back EMF voltage.^[1]

However, I find using input voltage as a reference in this context not to be useful, for two reasons. First, input current, as opposed to voltage, is precisely in phase with motor force, providing a link between the electrical and Newtonian systems. Input voltage is not, because voltage and current are shifted in phase with respect to one another whenever impedance is reactive. In other words, there is a phase shift between input voltage and current introduced by the reactive impedance before we can even approach

the subject of the phase relationship between input signal and cone position. Second, Back EMF voltage is precisely in phase with voice-coil velocity, which is also a Newtonian quantity. These quantities have fixed differential and integral relationships to one another, which are lacking when input voltage is used as a reference. Input current, however, has a fixed integral relationship to cone position. For this reason, we will define "slip" to be the angle between input current and cone position, not the angle between input voltage and Back EMF voltage. This is why, in the second part of the article series "Thought Experiments," we used current as a reference, not voltage.

Calculation of Angle of Slip Between Input Current and Back EMF

We know the phase angle of the Back EMF from calculations described in the first article in this series (May 2018). The voltage of the Back EMF is reversed in polarity from its impedance, just as with the DC motor. The velocity of the cone is precisely in phase with Back EMF voltage. Velocity precedes position, because velocity is defined to be rate-of-change of

position. Therefore, cone position lags behind velocity by 90° (see Figure 5 from the first part of the article series, May 2018).

So, we take the phase angle of the Back EMF impedance, subtract 180° to get the phase angle of the voltage, and then add 90° to represent the lag in time between velocity and position. The angle of Slip at a given frequency, then is calculated using the following equation:

$$S = (\Theta - 90) \text{ using degrees,}$$

where:

S = angle of slip

Θ = Back EMF Impedance Phase Angle

Figure 1 shows the angle of slip of the Fisher woofer (which is the example we have used throughout the article series) as a function of frequency. We see the angle of slip increasing as frequency is increased, with the most rapid increase in phase delay occurring near resonance. The angle of slip becomes more-or-less constant at higher and at lower frequencies, for reasons not described here. At very low frequencies, slip will approach zero, because there is very little delay between input and position. At DC, slip drops to zero.

Slip Causes Back EMF to Simulate LCR Components

A reactive load (capacitive or inductive) is characterized by its phase relationship between the input signal and the return of energy stored in the reactive component(s). In a loudspeaker driver, energy is stored in, and returned from, the moving mass of the cone assembly and the spring compliance. As this stored energy is returned, the voltages generated by the cone's motion (Back EMF) have phase relationships to input signal identical to those of the reactive components in the LCR model (see **Figure 2**).

To the source voltage, there is no difference between the LCR model and the moving-coil loudspeaker. The return of the energy to the system from the LCR model duplicates the return of energy from the mass and compliance of the loudspeaker, relative to input signal. It is the inertia of the cone, causing its motion to lag behind the input signal, which causes these phase relationships, resulting in the reactive impedance of the driver.

Back EMF and LCR Models Are Slightly Different

We can model the loudspeaker's impedance at any frequency using only two components: a resistor and

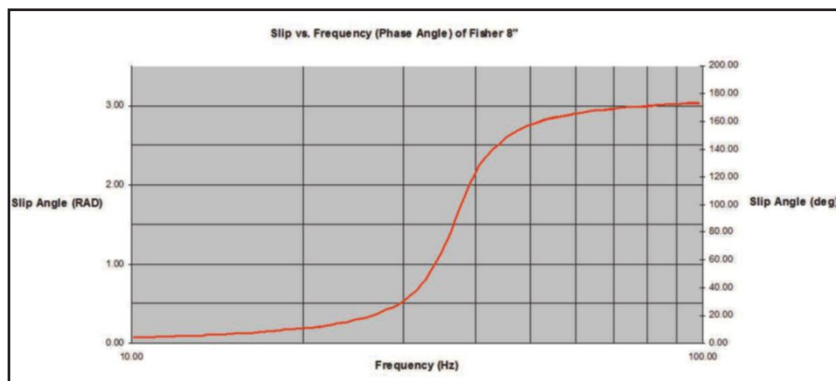


Figure 1: The phase angle of slip is shown between the input current and the cone position with respect to frequency.

a reactive component (inductor or capacitor). **Figure 3** shows the model at 23 Hz, below resonance. When we make a measurement of the complex impedance, we find a resistance in series with a reactance, as shown in Figure 3. (Above resonance, the reactive component would be a capacitor. The concept is the same.)

Figure 4 shows a more accurate representation of the moving-coil driver at this frequency. When Back EMF is involved, we can't perfectly model the loudspeaker using two components, because Back EMF itself simulates a resistor in series with another component (as shown in Figure 10 from the first part of the article series).

In a way, the two components making up Back

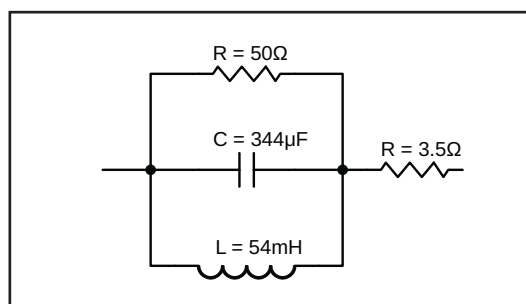


Figure 2: The LCR Model shows the equivalent real-world values of impedance for the Fisher woofer, which is used as an example throughout the article series.

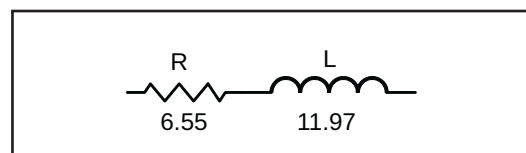


Figure 3: The resistor is shown in series with the inductor below resonance.

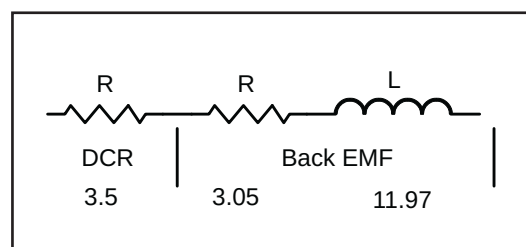


Figure 4: There is a distinction between voice-coil resistance and the resistive component of Back EMF.

EMF are in series and parallel at the same time. They share voltage, as if they were in parallel, but their impedances add as vectors, as if they were in series. Therefore, a loudspeaker's impedance can't be perfectly modeled using two components, as in Figure 3.

Inductors are the same way. They can't be perfectly modeled using a pure inductance and a pure resistance, but only using an infinite number of tiny components and integral calculus. Back EMF can be either capacitive or inductive, depending on frequency. Therefore,

when comparing the behavior of Back EMF to that of the LCR model (shown in Figure 2), it is necessary to introduce a correction factor to compensate for the difference between the pure LCR models and the actual loudspeaker.

The correction factor is equal to the difference in phase angle between the pure reactive component and that of the Back EMF: $90 - 75.6 = 14.4^\circ$ at f_1 , (inductive), and -14.4 at f_2 (capacitive). These phase angles are sort of reciprocal to angle of slip ("sort of" because the relationships are not symmetrical below and above resonance—stay tuned).

Slip Angle of Back EMF Creates Inductive Reactance Below Resonance

We can graphically illustrate how, below resonance, the angle of slip creates an inductive reactance identical to that of the LCR model, or to that of the equivalent LR model. At f_1 (23 Hz), the impedance of our Fisher woofer can be modeled using a resistor with a resistance of 6.55Ω in series with an inductor having a reactance of 11.97Ω (see Figure 3).

We can illustrate the phase relationship of input current to the impedance of self-inductance (Back EMF) in this equivalent RL filter. We can see that the self-induced voltage precedes input current by 61.3° . The effective impedance of the self-induced EMF is 180° out-of-phase with voltage, as with any Back EMF. **Figure 5** shows the phase relationships occurring in the LR filter.

How Slip Creates Inductive Reactance

At the frequency f_1 , 23 Hz, the angle of slip is 14.4° , as shown in **Figure 6**. Velocity precedes position by 90° ; therefore, Back EMF precedes slip by the same amount. We know the angle of slip, so we can calculate the other relationships, which are shown in **Figure 7**.

Figure 7 is similar to Figure 5, but not identical. The curves representing the LR model show the Back EMF 61.3° behind current, and the curves illustrating the characteristics of the loudspeaker show Back EMF 75.6° behind current. The difference would be equal to the correction factor of 14.4° of the resistive and reactive components of Back EMF on the basis of their integrated nature, as previously explained.

Once the correction is made, the curve becomes identical to Figure 5, which shows the characteristics of the LR model. The phase position of the loudspeaker's Back EMF is the same as that of the self-inductance of the inductor in the LR model. Therefore, Back EMF is inductive, as is the driver's impedance.

Back EMF is Purely Resistive at Resonance

The speaker's impedance at resonance is purely

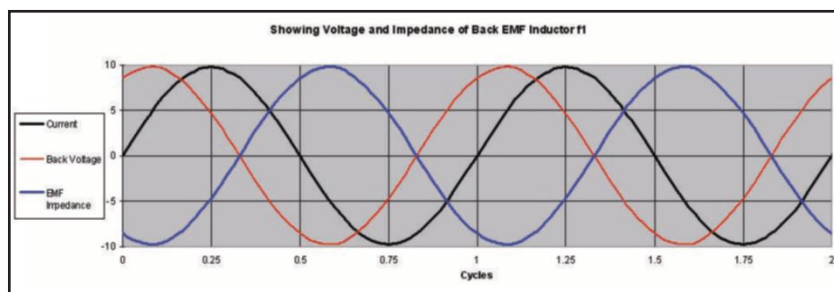


Figure 5: Here are the phase relationships of resistor/inductor model at f_1 , 23 Hz.

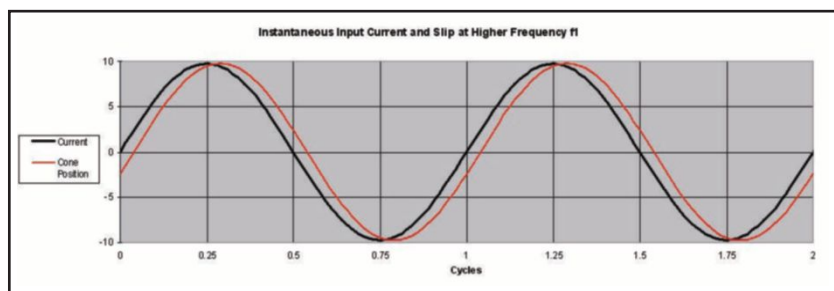


Figure 6: With a 14.4° Slip at test frequency, $f_1 = 23$ Hz.

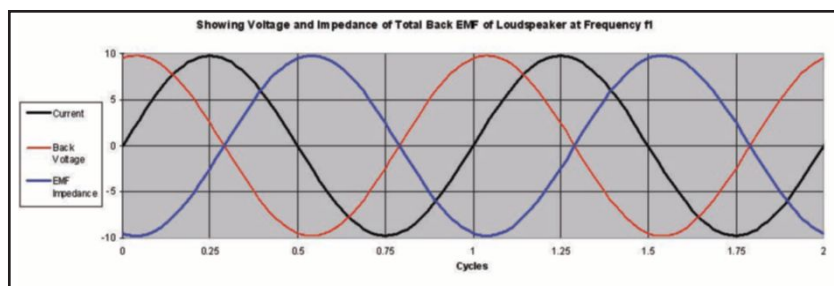


Figure 7: Here are the phase of Back EMF voltage and the loudspeaker's impedance at 23 Hz (f_1)

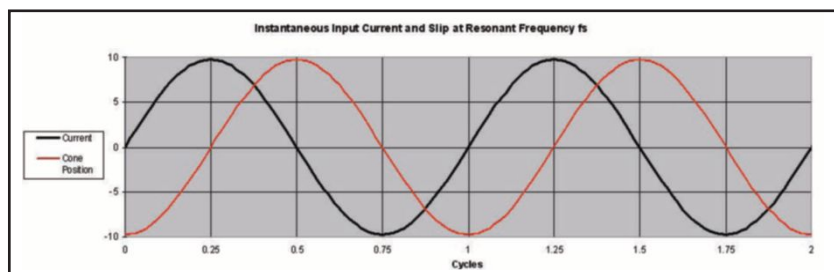


Figure 8: The slip at 90° (0.25 cycles) is at resonant frequency f_s , 36.93 Hz.

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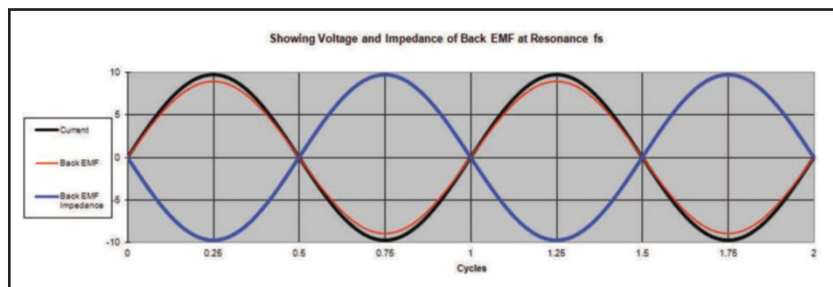


Figure 9: Back EMF impedance perfectly opposes the input signal at resonance.

Figure 10: The angle of slip creates a capacitive resistance identical to the two-component RC model of impedance at f_2 , 59.3 Hz.

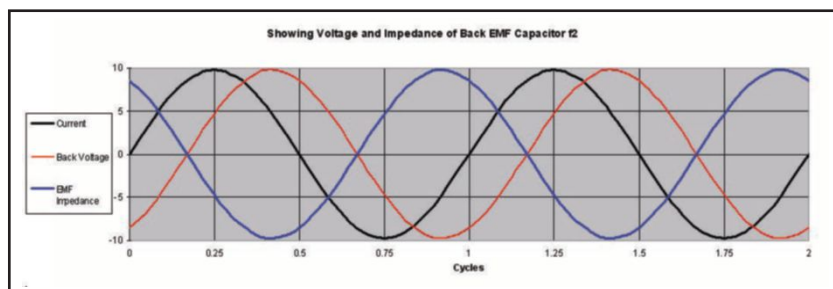
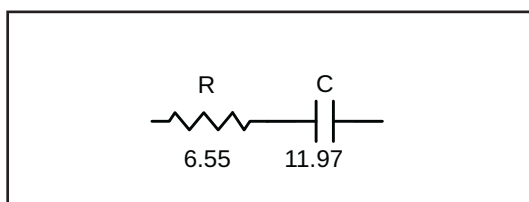


Figure 11: These are the phase relationships occurring in this equivalent RC filter at f_2 .

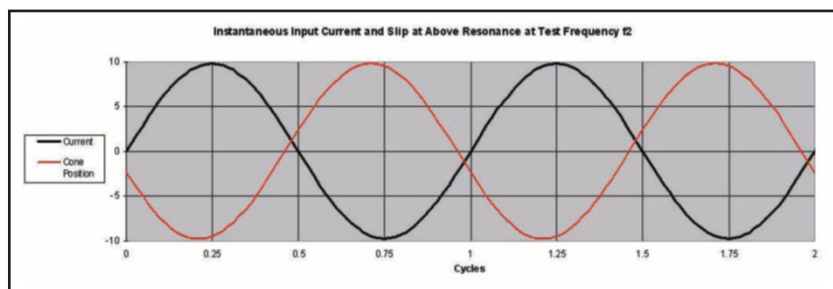


Figure 12: The 166° slip is shown at test frequency, f_2 , 59.3 Hz.

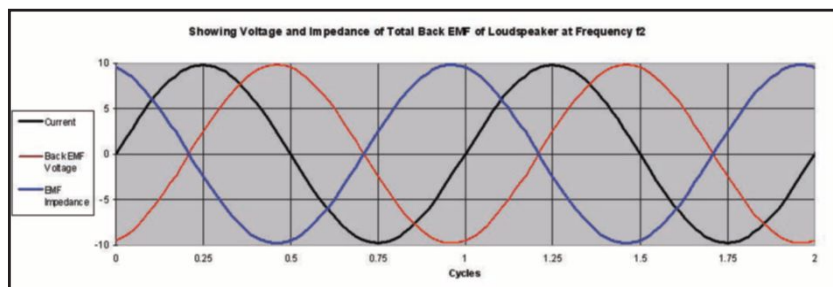


Figure 13: The phase of Back EMF voltage and the loudspeaker's impedance is shown at 59.3 Hz (f_2).

resistive, just as with the LCR model or the DC Rotary Motor (shown in Figure 2 in the first part of the article series). As always, Back EMF impedes the input voltage, but at resonance, the Back EMF is resistive, because of the angle of slip. The angle of slip has increased to 90°, as the mass of the cone causes it to lag farther behind input current (see Figure 1). The relationship between input current and cone position is shown in **Figure 8**.

Again, because velocity is in phase with Back EMF, and has a differential relationship to slip (position), it is possible to illustrate the relevant relationships graphically, as in **Figure 9**.

At resonance, the input current and voltage are in exact phase with one another, as with any purely resistive load. Cone position (slip), lags behind current by 90° (see Figure 1 and Figure 8). But because velocity (in phase with Back EMF) precedes position by 90°, the Back EMF is precisely in phase with input voltage and current. The resistance of the Back EMF presents a perfect resistive impedance to the input.

This is the exact relationship between the DC input current/voltage and the purely resistive Back EMF we see in the DC Motor, described in that section. There is no reactive component to Back EMF at resonance, and the resistor in the LCR model of Back EMF is equal to total driver impedance (resistance) minus the voice coil's DC resistance.

Slip Angle of Back EMF Creates Capacitive Reactance Above Resonance

We can illustrate how, above resonance, the angle of slip creates a capacitive reactance identical to that of the LCR model, or the two-component RC model (see **Figure 10**) at this frequency. The impedance characteristics are a mirror image of those at f_1 , below resonance. The slip characteristics, however, are not symmetrical in an analogous way, but are part of the continuous increase of slip, with frequency, as shown in Figure 1.

At the frequency at f_2 , 59.3 Hz, the impedance of our Fisher woofer can be modeled using a resistor with a resistance of 6.55 Ω in series with a capacitor having a reactance of 11.97 Ω .

The impeding voltage lags behind input current by 61.3°, as the capacitor charges. The impedance of this opposing voltage is 180° out-of-phase with voltage. **Figure 11** shows the phase relationships occurring in this equivalent RC filter.

References

- [1] R. Carver, "How to Design a High-Pressure, High Back-EMF Subwoofer," 2002.

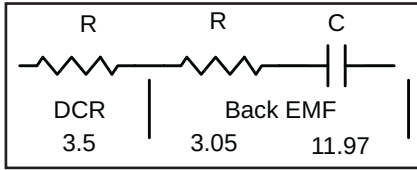


Figure 14: The dual parallel/series of the voltages on the components representing Back EMF changes the model slightly.

How Slip Creates Capacitive Reactance

At the frequency f_2 , 59.3 Hz, the angle of slip is 166° . The position of the cone lags far behind input current, as shown in **Figure 12**. As always, velocity precedes position, so Back EMF precedes slip. We know the angle of slip, so we can calculate other relationships, as shown in **Figure 13**.

As was explained, the dual parallel/series nature of the voltages on the components representing Back EMF changes the model slightly, as shown in **Figure 14** (compare with Figure 4). As before, this necessitates the introduction of a correction factor, this time of -14.4° . This results in curves identical to those of two discrete RC components (see Figure 11), which need not be replicated here.

Once the correction is made, the curve becomes identical to Figure 11, which shows the characteristics of the RC model. The phase position of the loudspeaker's Back EMF is the same as that of the voltage present on the charging capacitor in the RC model. Therefore, Back EMF is capacitive, as is the driver's impedance of the driver, as a result.

Summary

The impedance of a moving-coil loudspeaker in the region of its resonance is essentially equal to the vector sum of its DC resistance and Back EMF. The position of the moving assembly relative to input signal ("slip") is what causes Back EMF to present an inductive, a resistive, or a capacitive component to the impedance.

To the driving voltage at the loudspeaker's terminals, the characteristics a loudspeaker's Back EMF are indistinguishable from a model of the driver made of discrete LCR components, the values of which can be specified.

The inductive reactance associated with the increasing impedance of a loudspeaker below its resonance is not as a result of self-inductance, as in the case of an actual inductor, but rather of the phase relationship between the input voltage and Back EMF. The driver's impedance is purely resistive at resonance because the Back EMF itself has a phase angle of zero. Back EMF and voice-coil resistances can be added as scalars at resonance.

The capacitive reactance associated with the decreasing impedance of a loudspeaker below its resonance is not as a result of a self-inductance, as in the case of an actual inductor, but rather of the phase relationship between the input voltage and Back EMF. 

About the Author

Andy Lewis lives in Englewood, CO, plays drums, and produces "Acme Sound Low-B" low-coloration loudspeakers for bass players. He is also involved in Special Olympics and other disability activities with his son, Collin.

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Miking Solo Voices and Instruments

The Mic Choices

Although there are many microphones, from which to choose, you need to consider several important acoustical factors to obtain good results.

By
Richard Honeycutt
(United States)

For many years, there has been a widespread perception that miking for recording is a black art accessible only to a few master recording wizards, and that miking for live sound can be done by any fool with some Shure SM58s and a few XLR cables. This perception has led to idolization of a few recording engineers and disappointing live sound in many venues, especially those that rely on untrained volunteer techs.

There is a story about a renowned concert pianist who was to perform in a venue that required

sound reinforcement. When he casually mentioned the upcoming performance to his recording engineer, the engineer was horrified. He insisted on packing up his multi-thousand-dollar studio mics and stands, and accompanying the pianist to the sound check. Once at the venue, the recording engineer firmly instructed the house sound tech that he would mic the piano. He set up his mics as he normally would in a studio and plugged the cables into the stage box. When the system was powered up and the pianist started playing, the house sound tech gently pushed up the fader until the piano could barely be heard. The recording engineer, standing in the middle of the audience area, loudly called for more gain. The house tech complied, and the room was immediately filled with the wail of screaming banshees! In order to save the speakers and everyone's ears, the house tech brought the gain back down and respectfully, if somewhat smugly, asked the pianist if he could set up his mics—using his own method—on the piano. The pianist wisely agreed. The second sound check was highly satisfactory, and after a little collaboration between the two engineers, everyone

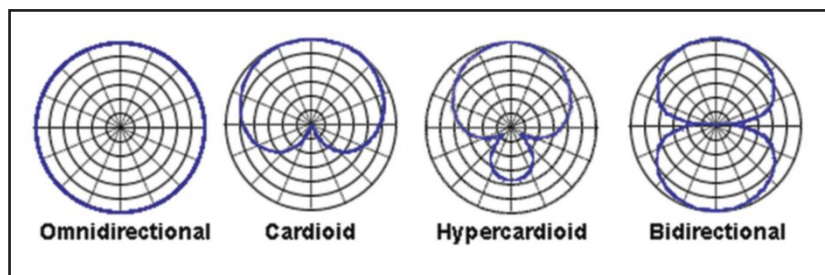


Figure 1: Most mics used for live sound have cardioid, supercardioid, or hypercardioid patterns. (The shape of a supercardioid pattern is between that of a cardioid and a hypercardioid.)

was happy, and the concert later proceeded with rave reviews.

Of course, the recording engineer in this possibly apocryphal story—though an expert in the studio environment—was inexperienced in miking for live sound. There are probably a dozen books and at least that many websites showing how to mike solo voices and instruments. The advice varies from book to book and website to website. Of course, the author's point of view affects the recommendations. Different engineers still argue. But there are important acoustical considerations that, if followed, will lead to good results. These considerations involve the microphones, the sound source (singer or instrument), and the environment (the stage, stage monitors, house speakers, etc.)

Mic Directionality

Most mics used for live sound have a cardioid, supercardioid, or hypercardioid directional response (see **Figure 1**). All these patterns preferentially pick up sound from the front of the mic and reject sound from the rear. The rear rejection (sometimes called “front-to-back ratio”) can be 20 dB or more, and varies somewhat with frequency.

Electro-Voice co-founder and mic guru Lou Burroughs (one of my heroes) recommended using omnidirectional mics wherever possible, because of their inherently flatter frequency response. However, most live-sound engineers have found that directional mics are needed in order to keep the banshees under control. The most common pattern is cardioid. You may notice from Figure 1 that the hypercardioid pattern has a rear lobe, indicating a reduction in front-to-back ratio for sounds coming directly from the rear of the mic. Supercardioids have a similar rear lobe. Thus, using these patterns in live sound can require careful placement in order to maximize rear rejection.

Proximity Effect

All directional mics are subject to proximity effect, which amounts to a bass boost when the mic is used up close. Southern Gospel bass singers have used this effect successfully to strengthen the sound of their bass notes, but for most vocalists, the effect is unnatural. **Figure 2** illustrates this effect in the frequency response of an Electro-Voice PL80a cardioid mic. Note the 12-dB boost at low frequencies when the mic is used at a distance of 1 cm from the source (dotted curve), compared to the response at 2' from the source (solid curve). The amount of bass boost depends upon the mic

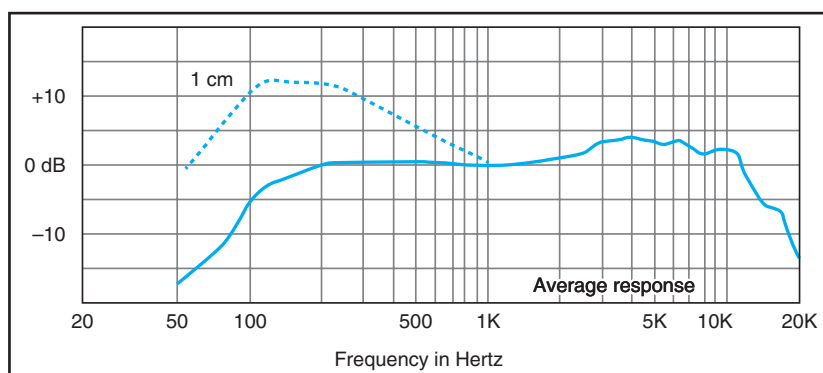


Figure 2: This frequency response curve for a cardioid mic illustrates proximity effect.

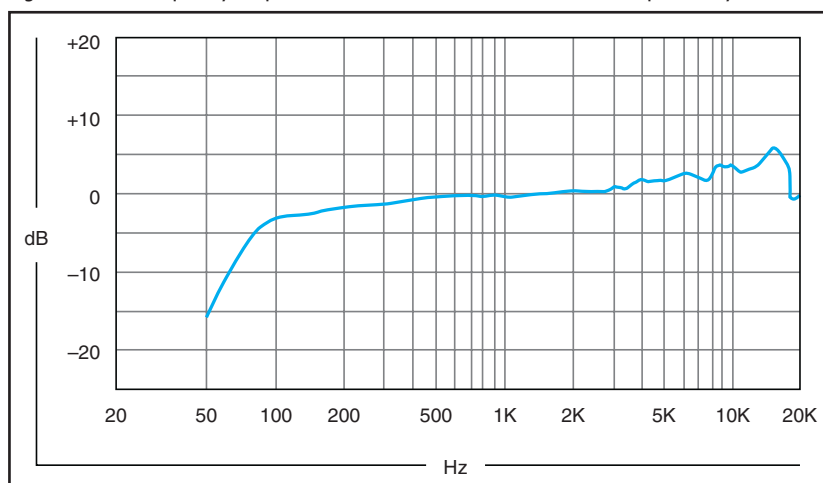


Figure 3: The low-frequency response of the Shure SM57 is deliberately limited to compensate for proximity effect.

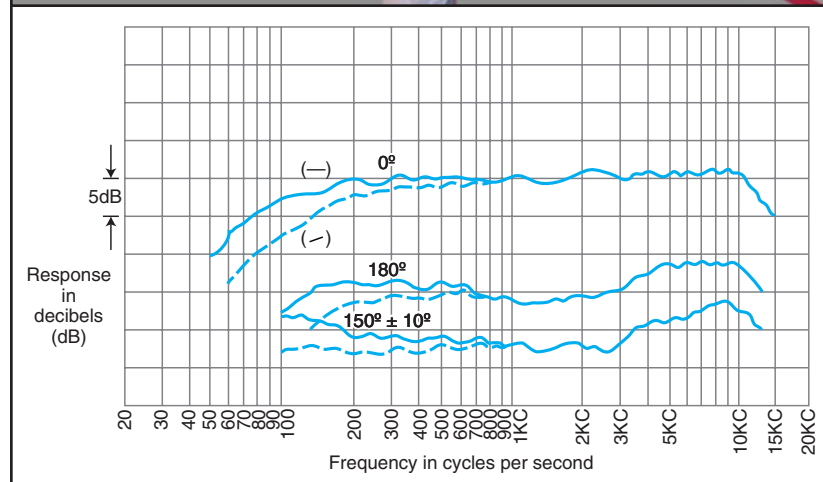


Figure 4: Variable-D mics have much less proximity effect than standard directional mics.

design: Mics designed to have an extended low-frequency response when distant from the source will have more proximity boost. The distant low-frequency response of mics designed for vocal use is often deliberately limited in order to control proximity effect when the mic is used in the 1-cm “tonsil position” (see **Figure 3**).

Variable-D Mics

Variable-D mics, used primarily in broadcast applications, avoid proximity effects by the use of a distributed rear port to provide directionality, rather than a simple hole-and-tube port. The “backbone” at the top of the Electro-Voice RE10 shown in **Figure 4** is the distributed port. If this port is covered by the fingers during handheld use, the mic’s frequency response will suffer. As you can see from the response curves in Figure 4, proximity effect is much less severe in a Variable-D mic than in other directional mics. The dashed curves represent the mic’s frequency response with the response switch in the “low-cut” position.

In many Variable-D mics, off-axis response is also more uniform than in more common mics.

Aside from proximity effect, many microphones provide an intentional boost over some range of high frequencies. Initially introduced to provide a greater sense of “presence” in an era when the high-frequency response of live-sound speakers might be weak, especially for those listeners (usually the majority) located off-axis, this “presence peak” caused more than one engineer to speculate whether we also needed an “absence dip.” All wisecracks aside, presence peaks are an important part of the voicing of a microphone, and can make or kill a mic’s suitability for a particular vocalist or instrument.

Frequency Responses

Figure 5 shows the frequency response of three classic microphones. The oldest is the RCA 44BX (see **Photo 1**). This bidirectional ribbon mic was a favorite for radio interviews, with the interviewer sitting on one side of the mic and the interviewee

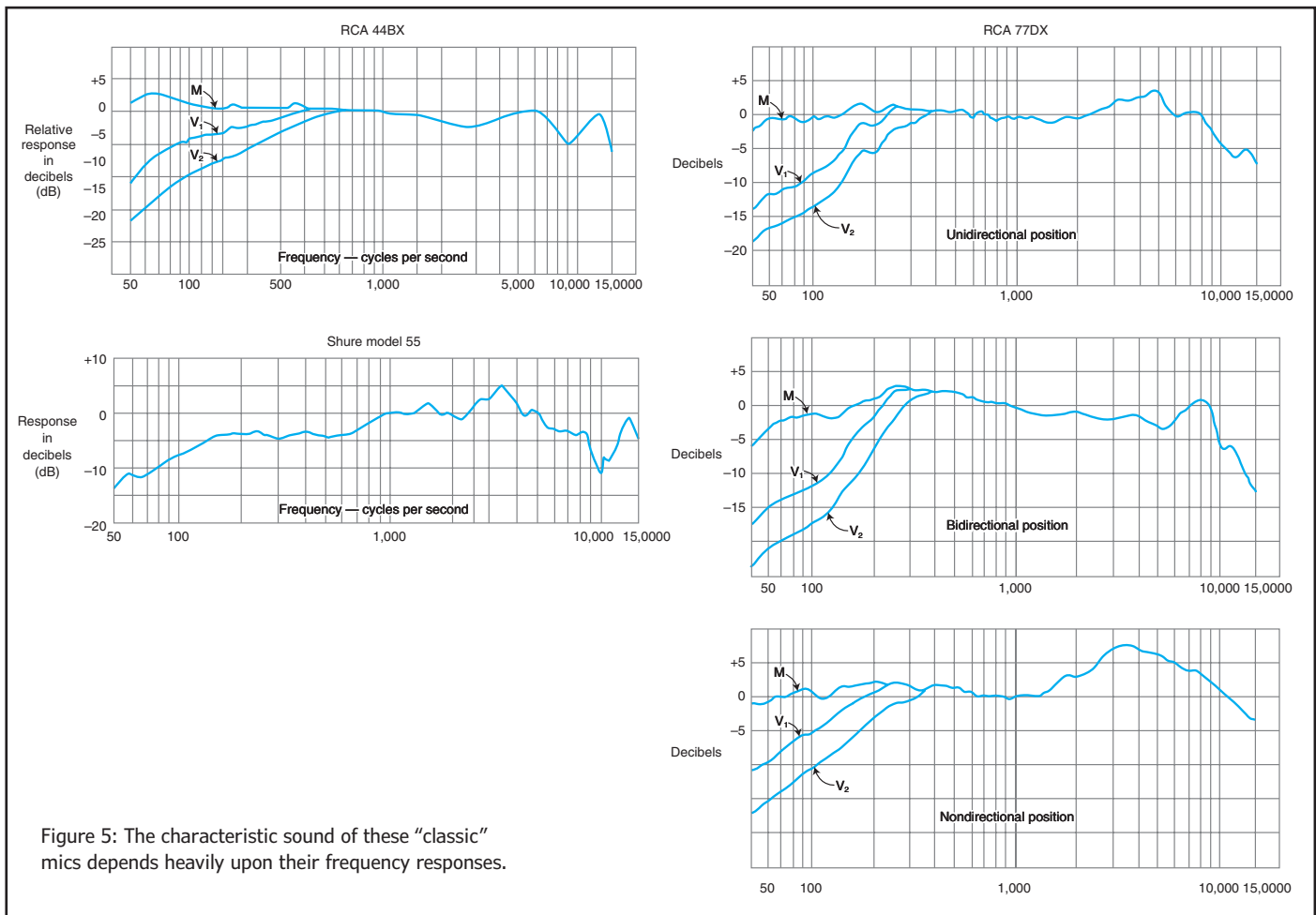


Figure 5: The characteristic sound of these “classic” mics depends heavily upon their frequency responses.

facing the other side. The three frequency response curves (top to bottom on the graph) are for music (working distance greater than 3'), close vocal/speech (working distance about 12"), and closer vocal/speech (working distance as close as 7"). The response was selected according to which terminal the output cable was connected. This mic was subject to damage from breath blast if care was not taken. (An aside: At a convention of the Audio Engineering Society in the 1980s, a respected international instrumental group performed a concert. The announcer, a US government official involved in arts exchange programs, walked onstage, and before introducing the group, blew into the microphone. A collective groan arose from the roomful of audio engineers. Thankfully, the mic was a dynamic model rather than a ribbon, so no damage was sustained!) The 44BX had some response ripples—less than 5 dB peak-to-trough—in the upper treble range, but was among the best mics of its era.

The Shure Model 55 (see **Photo 2**) may well be the most recognized microphone, largely because it was one of Frank Sinatra's favorite concert mics, along with the Altec 639B (not shown, but visually somewhat similar). Its low-frequency response was deliberately limited to compensate for the proximity effect when used up close (usually!). The response ramped up about 8 dB at 1 kHz, giving increased vocal clarity, but also imparting a sound that some have described as "strident." The 2-dB bump between 1 kHz and 2 kHz did not complement some voices. The marked presence peak between 4 kHz and 5 kHz often invited banshees to the performance. And the drooping response between 6 kHz and 12 kHz made this mic an unlikely choice for studio use, although this effect did not much hamper its usefulness as a live vocal mic.

The RCA 77 DX ribbon mic (see **Photo 3**) had three switchable directional patterns: omni (nondirectional), cardioid (unidirectional), and bidirectional. The frequency response curves in



Photo 1: This RCA Model 44BX was ubiquitous in the late 1940s and early 1950s.



Photo 2: The Shure Model 55 was a favorite vocal mic in the early 1950s.



Photo 3: The RCA 77DX was Johnny Carson's first trademark mic.



Photo 4: The Electro-Voice 635 is one of the all-time favorite broadcast interview mics.



Photo 5: The Neumann U87Ai FET condenser mic has been a favorite for recording since its introduction in the 1960s.

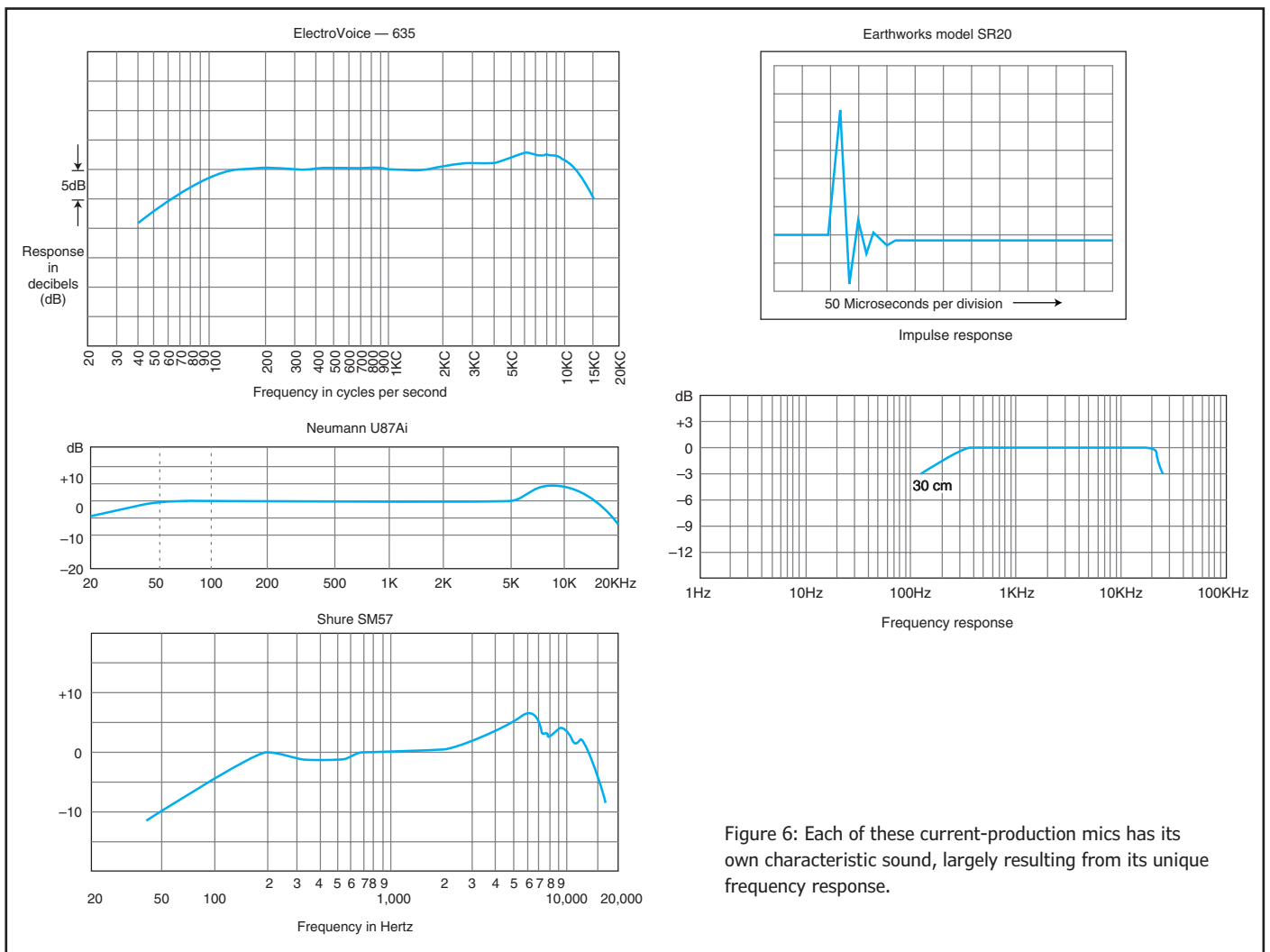


Figure 5 show the effects of the built-in low-cut filters, which were similar to those in the 44BX. Note that the choice of directional patten also significantly influenced the high-frequency response.

The Electro-Voice Model 635 omnidirectional handheld mic (see **Photo 4**) has arguably been the most popular mic used for handheld pickup of broadcast interviews and press conferences, because of its relatively flat frequency response and lack of susceptibility to handling noise. (All directional mics are more sensitive to handling noise than are omnis.) Because of its nondirectionality, the 635 was seldom used in live sound.

The Neumann U87Ai multipattern condenser mic (see **Photo 5**) could be used as an omni or a cardioid, depending upon a switch setting. The manufacturer's response curve given in **Figure 6** is not identified as to whether it was made using the omni or the cardioid setting, but I suspect it was the omni one. The gentle presence peak centered about 8 kHz was responsible for the U87's signature sound. Although used mostly in recording studios, the U87 has also seen much use picking up guitar and piano in live-sound applications, particularly for classical performances.

Although presently marketed as an instrument mic, the Shure SM57 (see **Photo 6**) took the industry by storm for both vocals and instrumentals when first introduced in the 1970s. The aggressive presence peak at 6 kHz made it very susceptible to feedback, unless sound from the house and monitor speakers could be well isolated from the mic.

The Earthworks SR20 (see **Photo 7**) is one of a few handheld mics offered by a company mainly noted for its studio and measurement mics. As you can see from the response curve in Figure 6, the manufacturer claims ruler-flat response from 300 Hz to 20 kHz at a working distance of 30 cm. Low frequencies are rolled off to compensate for proximity effect. Earthworks founder David Blackmer, while aware that the Fourier Transform of a time-domain ("impulse") response yields the frequency response, nevertheless believed that the mic industry paid too little attention to impulse response when designing mics. Since sound reflections from structures in front of the mic diaphragm clearly affect impulse response, Blackmer always strove to avoid detrimental effects from such reflections. His influence still affects the design philosophy of Earthworks today, as shown by the impulse response of the SR20 being included on the datasheet, along with the frequency response. As you can see, ringing disappears by about 125 μ s after the impulse.



Photo 6: The Shure SM57 has been widely used for both vocals and instruments, although it is now touted primarily for instrument pickup.



Photo 7: The Earthworks SR20 handheld mic is exquisitely accurate.

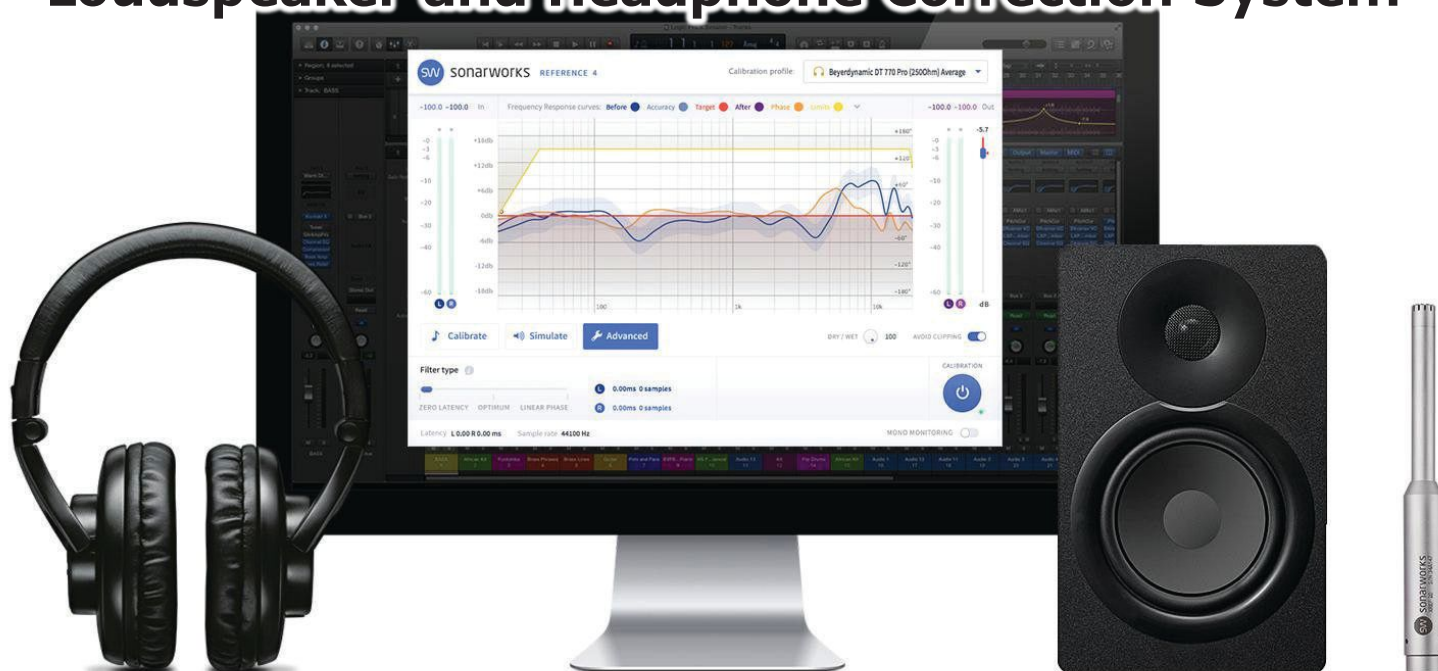
Final Note

One final note concerning mic choice for solo voice and instruments: Remember the sound pressure level (SPL)! Some instruments—especially percussion instruments such as bass drum—can produce very high sound pressure levels. Most professional mics can handle at least 120 dB without adding significant distortion, but a few vocalists and instruments can produce higher levels. Although most professional mics produce under 1% total harmonic distortion (THD) with sound levels up to 139 dB to 140 dB, some less expensive mics, and especially some wireless mics, can give a clean output only for SPLs below 110 dB or even lower.

In another article, we will discuss techniques for working within the laws of electroacoustics to provide high-quality live sound. 

Sonarworks Reference 4 Studio Edition

Loudspeaker and Headphone Correction System



Stuart Yaniger takes a look at Sonarworks, a relatively new entry into this market and its main offering—the Reference 4 Studio Edition, which is clearly targeted to studio and mastering use but can certainly be used in home audio setups.

By
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 (United States)

Although various loudspeaker and room equalization schemes have been around for decades, recent advances in digital signal processing have enabled far more sophisticated and effective methods to be implemented. As with nearly everything involving digital electronics, the costs have dropped as rapidly as versatility and efficacy have increased. Even freeware systems such as Room Equalization Wizard (REW) do a remarkable job of improving the sound in home audio setups. We've come a long way from the manual third-octave setups of my youth.

A relatively new entry into this market is Sonarworks, whose main offering—the Reference 4 Studio Edition—is clearly targeted to studio and mastering use, but can certainly be used in home audio setups with great success. The basic idea is

that mixing and mastering through monitors that have frequency response peculiarities can result in recordings that don't sound as good as they could—it's like measuring the length of something with a crooked ruler. The intent of the Reference 4 software is to straighten the ruler out, giving a recording engineer a neutral and accurate tool to evaluate the sound of a mix.

The Products

Sonarworks Reference 4 has four basic built-in functions. First, software to make measurements of the speakers in the monitoring room. Second, an engine that takes the results of these measurements and converts them to a custom frequency-response-correction plug-in to use with any DAW having the

standard capability of using VST, AU, AAX, or RTAS formats. Third, a mode called Systemwide, which only works on audio outputs, but does so for all computer audio functions, not just the DAW. In Windows, it uses Direct Sound to implement the frequency response correction on all outputs. For Macs, it works as a Core virtual audio device with no inputs and two outputs to accomplish the same goal. And fourth, for both the plug-in and Systemwide modes, Sonarworks Reference 4 contains a collection of pre-determined frequency response correction curves for the most popular headphones.

Several versions of the software are available at different price points. The full Reference 4 package incorporating all these functions sells for \$249. A serialized measurement mic can be added for an additional \$50. For headphone mastering where there's no need to do measurements and generate custom frequency response correction curves, the Reference 4 Headphone edition incorporates the fourth function only and sells for \$99. Sonarworks also sells the Reference 4 system bundled with Sennheiser HD650 headphones for \$699, which includes an individual calibration for the headphones.

Speaking of which, Sonarworks will also provide you with an individual calibration for your headphones for \$99 extra, plus shipping back and forth to their headquarters in Latvia. They claim that individual calibration will reduce frequency response deviations from ± 3 dB using the general calibration curve to better than ± 0.9 dB. Additionally, Sonarworks resells a range of high-quality headphones that are also supplied with individual calibration profiles.

For a typical headphone-oriented audio enthusiast, as opposed to a music producer, Sonarworks offers a product called True-Fi, which includes the same group of headphone calibration curves as the Reference 4 Headphone Edition, but only has Systemwide capability (i.e., no plug-ins for DAWs).

Note that although you can default to an equalization to a flat frequency response, the Reference 4 (and True-Fi) software has the capability of equalizing to a user-specified target response. There are also some pre-loaded target responses corresponding to typical responses for large and small home speakers, as well as the B&K 1974 Optimum Hifi Curve (which basically is a downward tilt in the frequency response to make up for beamy tweeters). So if you want something like a house curve, or to master with particular target speaker types in mind, you can easily implement that.

Additionally, the software enables you to simulate the playback sound from six different popular headphones and loudspeakers so that you can see how your mix sounds to users of these products. I

will confess to spending a lot of time playing with these simulations and (very unkindly) laughing at how well some pretty awful-yet-popular products could be mimicked. In my opinion, this is a very useful and unique feature.

The True-Fi version also has an EQ function meant to compensate for age- and gender-related presbycusis. That strikes me as not terribly useful for someone who wants to recreate live sound since "live sound" is judged by those same defective ears (see **Figure 1**), but it's certainly a sobering reminder of one's humanity.

The Technology

The core of Sonarworks's technology is outlined in its US Patent 9,380,400, "Optimizing Audio Systems," and the pending application 2016/0366518, "Method of, and Apparatus for, Determining an Equalization Filter," with company principal Kaspars Sprogis listed as the inventor. Essentially, their method consists of measuring frequency response at a variety of

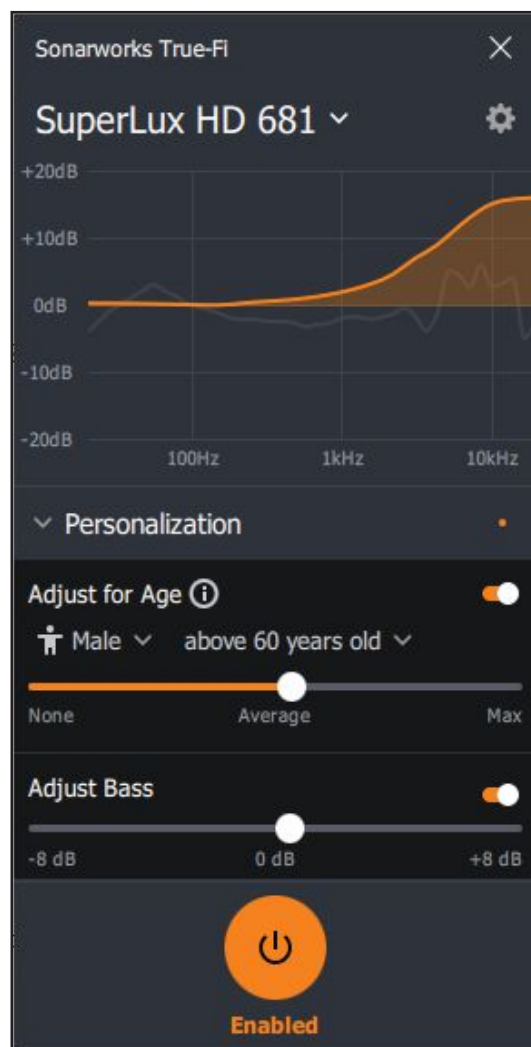


Figure 1: The True-Fi headphone correction software enables the user to enter gender and age, then shows an equalization curve. This was depressing.

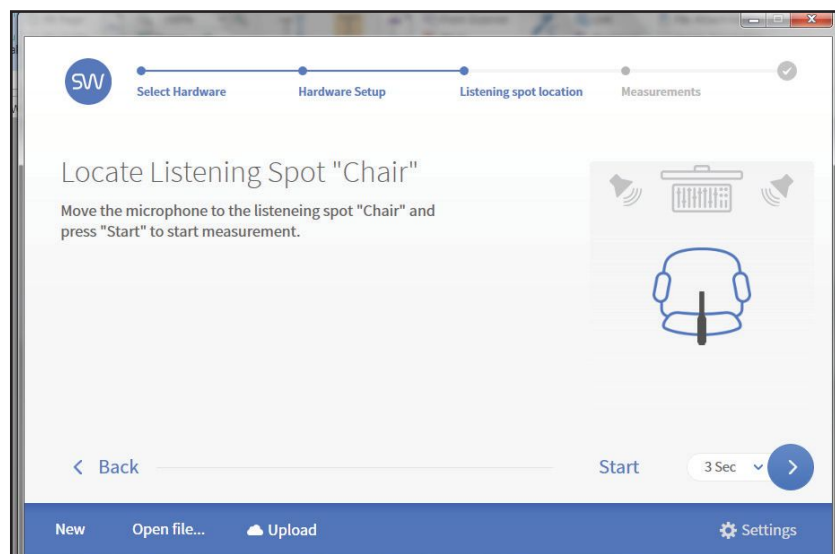


Figure 2: The Reference 4 software uses test signals to automatically determine the distance between the speakers and from the speakers to the listening position.

points, then deriving the equalization curve from a weighted average of responses gathered at those points. The weighted averages can be manipulated to only incorporate parts of the audio spectrum, if desired (e.g., a measurement from a meter away from the listening spot might have the top octave removed from the weighting), and the weighting can be varied as a function of the measurement location for that particular curve (e.g., frequency response curves from a meter away from the listening spot would have a lower weighting than those taken 10 cm from the listening spot). For loudspeakers, the user

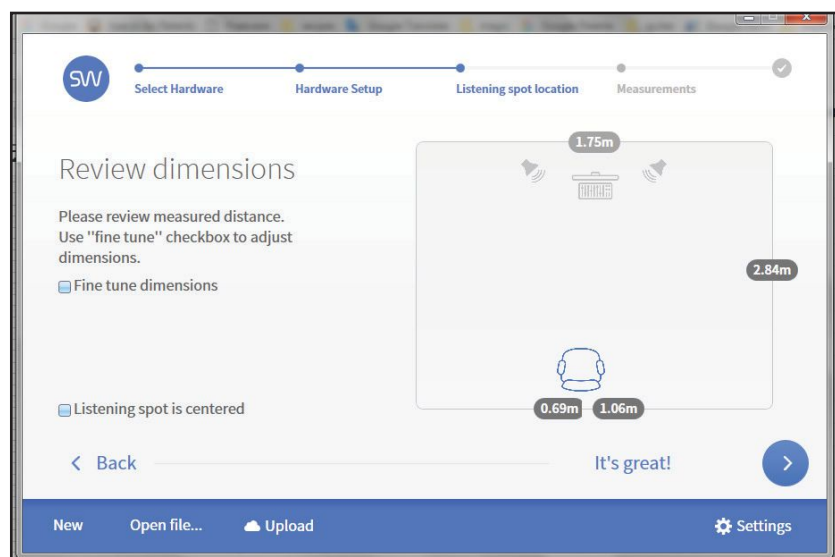


Figure 3: Significant room reflections can cause the position to be incorrectly determined.

measures them in situ so that room acoustics and angle to the listening position can be accounted for in the response corrections.

One interesting twist is that the Reference 4 system uses signals to automatically determine the spacing between the speakers, the distance from the speakers to the listening spot, and during the measurement cycles, the position of the test mic at each test point so that weightings can be accurately calculated.

This triangulation function makes the measurement process go very quickly and painlessly. I was able to obtain the required 24 measurement points in about 10 minutes. Additionally, since the software “knows” where the mic is placed at each point in the measurement process, it can rationally average out peaks and dips due to reflections, weighting them more heavily toward measurements near the listening position.

For headphone calibration curves, the process is similar, but since the user is likely unable to be able to do headphone measurements, this is done at Sonarworks and the calibration curves are provided. As with the speaker corrections, Sonarworks’ method consists of measuring the response in multiple locations within the headphone cup and similarly taking a weighted average to determine the final correction curve.

According to its website, this is done with multiple samples of each headphone model so that an average calibration can be determined. Because there’s a lot of response variation between headphone cavities with different pinnae in them (and no two pinnae are alike), Sonarworks also has the correction curves judged by a listening panel.

Rudi Putnins at Sonarworks says, “We use different measurement methods depending on headphone type and even then a committee of no less than three trained audio engineers check the outcome by ear to correct for measurement artefacts. They all have to greenlight the curve, before we can publish it. Sadly, there have been cases where we’ve chosen to not publish the calibration profile, as it wasn’t to our standards. Most often it happens due to high sample-to-sample variance or hard-to-correct resonances.”

Interestingly, you’d expect that if two different types of headphones are equalized to give the same frequency response, they would sound extremely similar. Sonarworks goes a bit farther and claims, aside from any issues with high distortion in some headphones, that once equalized, different model headphones will essentially sound the same. This is not an unreasonable claim, given the human ear’s extreme sensitivity to frequency response and relative insensitivity to low levels of distortion.

Sonarworks Reference 4 Setup

The first step is installation and license activation. This is a bit convoluted, but can be done relatively quickly via a download manager application provided by Sonarworks, along with the activation code. If your studio computer is not hooked to the Internet, activation will require a separate computer to obtain the validated key, which is then transferred to the studio computer. The license with purchase is valid for three computers, which for someone like me (with a separate lab and living room setups) is a great convenience.

Unless you're going to just use the Reference 4 system with headphones, the first step is to set up the speaker measurement to generate the correction curves. I did this on two setups, my lab/studio monitoring (a pair of Vanatoo Transparent Zeros used near-field) and my living room (four-way floor-standing speakers based on NHT M3.3 with the listening spot about 3 meters away). The software allows the option of using a mic provided by Sonarworks, serialized to enable its frequency response to be automatically accounted for, or a user-chosen mic, with provisions for entering calibration data. In my testing, I used a reference mic I had on hand, the PCB Piezotronics 376A32 1/2" phantom power condenser mic, which has a very flat frequency response and low distortion as well as nearly identical response on and within $\pm 45^\circ$ off axis.

The measurements begin with a short tutorial on mic positioning and a setup checklist. I tried cheating a bit on one checklist item—make sure you're using the same interface for recording and playback. This is because my usual analog interface (a Scarlett 2i2) used for phantom-power mics, does not have an SPDIF output to send to the optical input I use for my lab/studio speakers or the AES/EBU input to the electronic crossover for my living room setup. Initially, I tried using the Scarlett for the mic and a separate USB-connected digital coax and optical adapter, which I soon found out caused the measurement process to not work correctly. I got nonsense for the speaker and listening seat spacing. The timing is apparently quite critical. The moral of the story is that when they say to use the same interface for both, they mean use the same interface for both! The software then has you set the input and the output channels for the mic and the loudspeakers, as well as the sample rate, which needs to be 44.1 kHz for these measurements.

Next, place the mic at the listening position, and the software begins test signals, a combination of clicks and sweeps, for the purpose of setting volume and mic sensitivity levels. The user can select the delay time between the triangulation signals

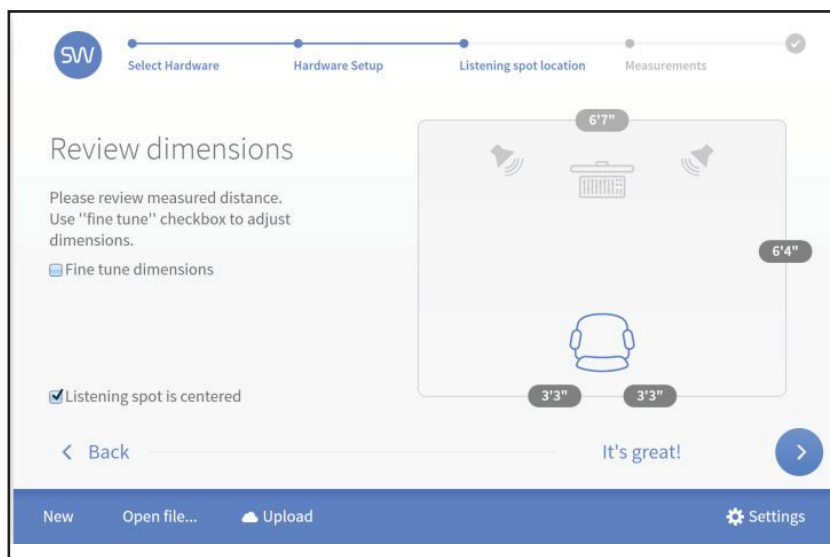


Figure 4: The software correctly locates positions in normal room or studio setups.

(clicks) and the measurement signals (sweeps) if it's necessary for (say) positioning the mic, then returning to the computer before starting the next measurement step.

In this first step, the software will give an error message if the levels aren't sufficient to give a high-quality result. Confusingly, the error is stated as "measured high signal to noise ratio" when it's actually the opposite. I attribute this minor hiccup to language issues (Sonarworks is in Latvia), of which a few pop up here and there. The prompts do, however, tell you to increase the sound level or the mic gain, so this is more of an aesthetic quibble on my part.

Once the test levels are set, the software directs you to place the mic close to each speaker in turn while test signals are played. The distance between the speakers is then automatically determined and displayed. If the distance displayed is not exactly correct, the user can adjust it manually, but if it's more

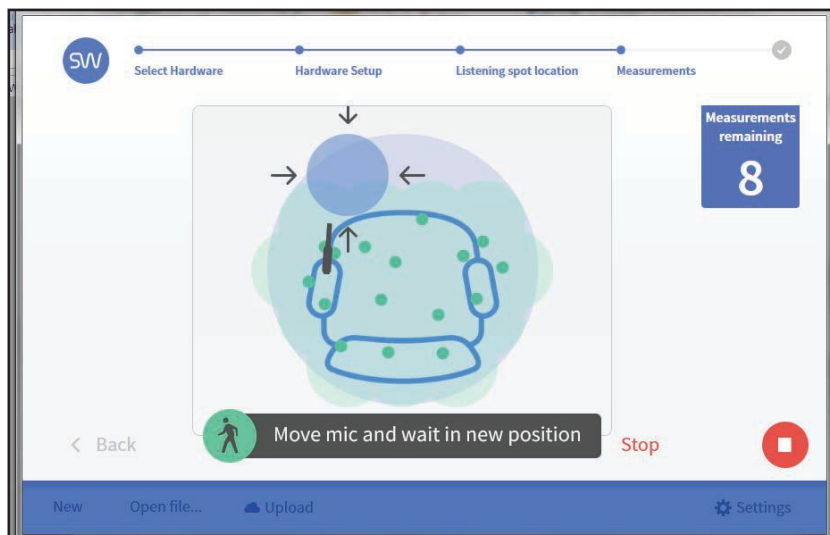


Figure 5: During the measurement phase, the Reference 4 software displays the mic position and the measurement position to allow the mic to be accurately located.



Figure 6: After completion of the 24 response measurements, the software generates frequency response correction curves for left and right channels.

than a few centimeters, that's usually an indication that something is not set correctly. (This is how I discovered that my setup using two different interfaces for mic and speakers was not going to work.)

With the speaker-to-speaker distance determined, the software directs you to place the mic at the listening position, then triangulates on that position

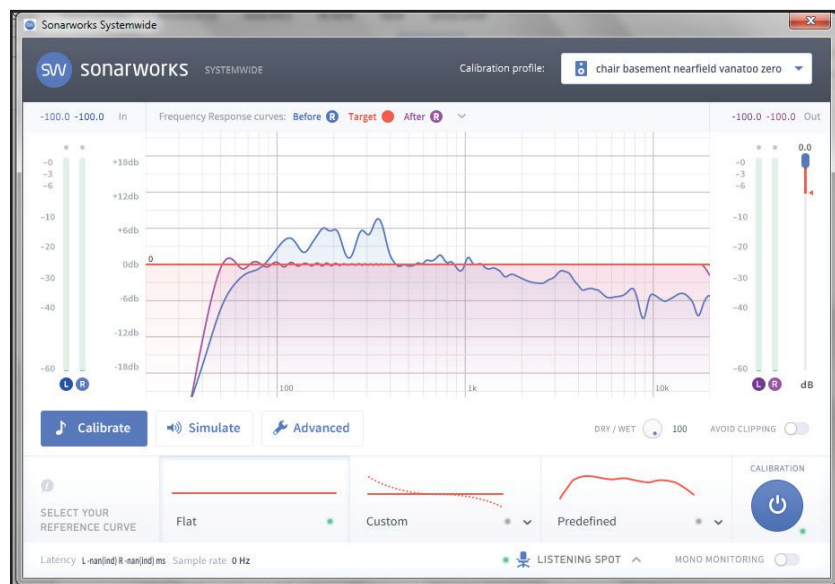


Figure 7: My studio monitors in the near field generated this response and correction curve.

About the Author

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

to set it as a reference (see **Figure 2**). As before, the software enables you to adjust the triangulated distance to the listening seat, and if it's off by more than 10 cm, this is an indication that something may be amiss in the measurements. This was the case when I attempted to do a measurement in my lab at a distance of 3 meters or so from the speakers. The display showed that the mic was very far from the center line between the speakers, when in reality it was dead center (see **Figure 3**). I traced this to some room reflections, which confused the triangulation. Adding some acoustic panels at a critical spot cleared up the issue.

Nonetheless, this brings up an interesting point that I'll return to later. And in my much less reflective living room, as well as in the lab speakers' near field (1.7 meters distance, which is representative of typical studio use), the triangulation functioned perfectly with no extra effort being needed (see **Figure 4**). As you can see here, the user has the option to select metric or Imperial units.

Now the fun begins. The software guides you through 24 separate measurements. The software display draws a circle in the area where the mic needs to be placed for each measurement and the triangulation function shows you where the mic actually is so it can be moved to the desired area (see **Figure 5**). Once it's located within the circle, the emitted sounds from the speakers change from the triangulating clicks to frequency sweeps after the selected delay time. The software then directs you to place the mic in the next spot, and so on until measurements at 24 points have been completed.

I should note that Sonarworks suggests using a mic stand, which is fine except that measurements in the seating area become problematic. After some experimentation, I found that if I attached the mic to the end of an approximately one meter boom, I could stand well behind the listening chair and move the mic around to the various measurement spots without my body (which over the years has transformed from reflective to absorbing) interfering too much with the measurement. At the same time, I could also see the computer screen enabling me to adjust the mic positioning to the spots where the software directed.

With all 24 measurements completed, the software returns the measured response used for the correction curve (see **Figure 6**). To more clearly demonstrate that the channels are individually measured and calibrated, one channel's gain in my living room system was reduced by 4 dB or so, which is reflected in the measurements. For my actual monitoring work, I used the Vanatoo speakers in the near field (see **Figure 7**).

Sonarworks Reference 4 in Action

For the in-use testing, I tried the software in two ways—as a VST plug-in for my DAW (Adobe Audition) and in the Systemwide mode. The results either way were the same, so I ended up just using Systemwide, which is very easy to customize or to switch in and out for comparisons. Your preferences might differ if you're doing more intensive mixing and mastering than I do, so it's nice having both options. But be sure to pull out the plug-in when the mixing or mastering is completed!

I first tried using the Systemwide with the pre-loaded correction curve for my cheap Superlux headphones, which can be seen in **Figure 8**. Although I have a healthy skepticism about headphone response measurement in general, and Sonarworks's particular claim that the equalization will make headphones of different models sound the same, I have to admit that the sonic results were subjectively spectacular.

The cheap headphones didn't sound cheap at all and were almost unrecognizable. I didn't have a reference set of headphones on hand for comparison, but even if the correction didn't make the Superlux sound exactly like Stax, it was still very impressive. I'd suggest that the \$79 True-Fi software (which uses the same correction curve) and a decent, inexpensive pair of headphones would be an amazing listening bargain for audio enthusiasts.

Next, I used the Vanatoos and selected the correction curve used for near-field monitoring, shown in Figure 7. For curiosity's sake, I checked the response using an Audio Precision APx515 and got the results shown in **Figure 9**. This matches the correction shown in the Systemwide display, so the latter can be presumed to be the actual correction used.

From a subjective standpoint, the correction did exactly what you'd expect from the curves—it removed the slow treble droop, compensated for the tweeter's 9 kHz dip, and smoothed out the bass. This came across as an increased clarity and liveness, but for recordings made with less-than-stellar microphones, the sound could be a bit fierce. This means the producer needs to know what the sound really is and that can help guide improvements in the mix. Beware though, once you get used to a colored sound quality and then fix the spectral balance, it may sound odd and unfamiliar until your ears adjust.

Both Systemwide and the plug-in have three filtering options with varying amounts of latency, which trades off latency for phase response accuracy. I measured the impulse response for the correction using each of the three filtering options provided (see **Figure 10**). Note that the Audio Precision APx515 has

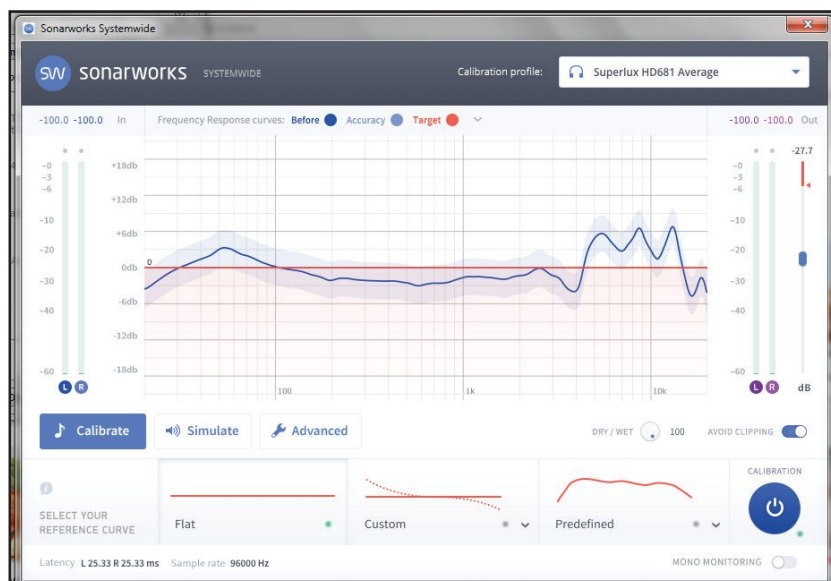


Figure 8: The correction curve for my Superlux headphones was pre-loaded in the Reference 4 software.

a delay before the impulse, so this graph should be interpreted in a relative sense. The “zero latency” option really does have close to zero latency, with the “optimum” filter having about 24 ms of delay, and the “linear phase” filter having about 47 ms of delay. Sonarworks recommends the “optimum” option for most uses, but “zero latency” should be chosen for live tracking or overdubbing.

One other thing of note here—as can be seen from the impulse responses, the Reference 4 software only does a frequency response correction. There is no time domain correction for things such as reflections, otherwise the impulse responses would indicate reflections at several milliseconds from the main

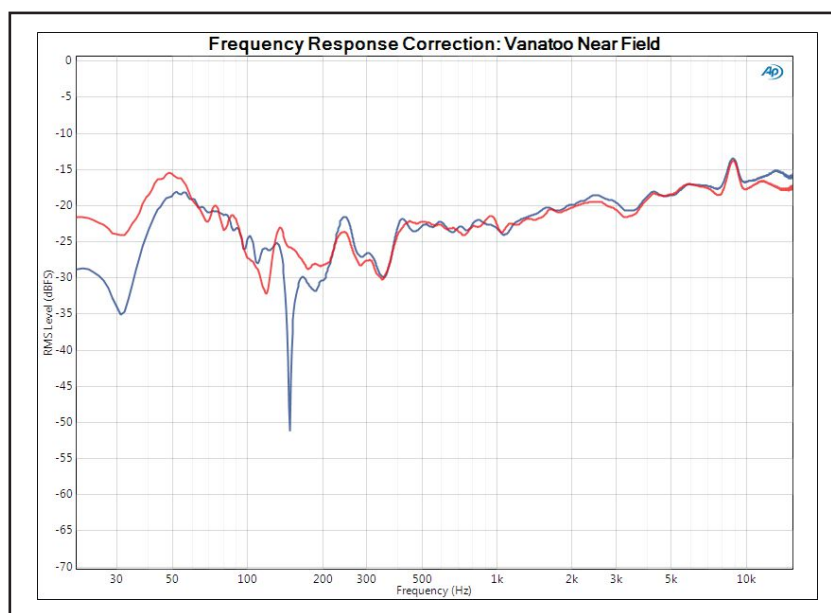


Figure 9: Measurement of the actual correction applied for my Vanatoo studio monitors closely matches the inverse of their measured response.

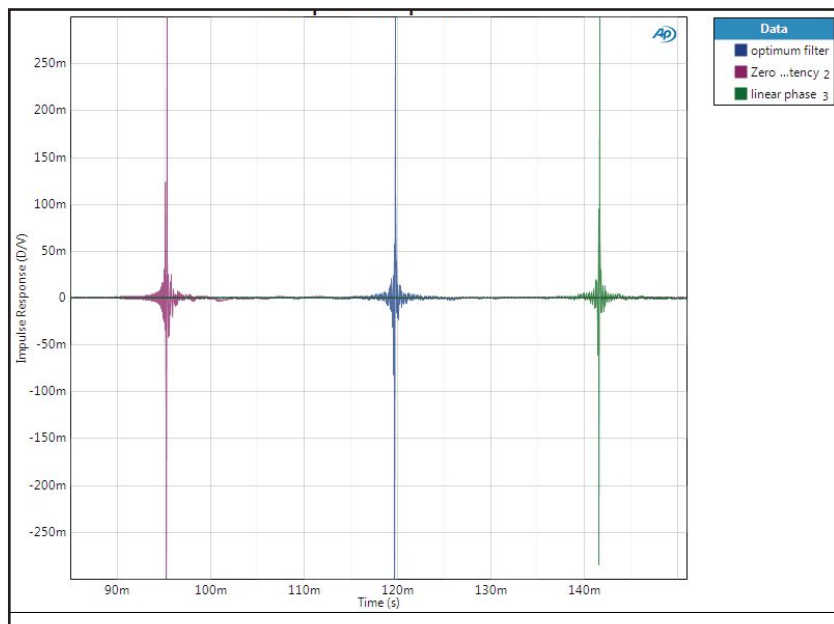


Figure 10: The Reference 4 software has three filtering options with different amounts of latency.



Figure 11. Target functions other than Flat can be selected, in this case the 1974 B&K Optimum Hifi Curve.

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Sonarworks Reference 4, www.sonarworks.com/reference

pulse to compensate for wall and floor reflections.

As I previously mentioned, the Reference 4 software enables the user to select a correction target other than flat, including the 1974 B&K Optimum Hifi Curve. I noted that the uncorrected response of my Vanatoo monitors generally followed that curve, and indeed, when I set it as a target, the difference between corrected and uncorrected became much more subtle (see **Figure 11**).

Summing Up

There's a lot of competition in the room/loudspeaker correction software market. Sonarworks's claimed advantages of the Reference 4 Studio Edition are a larger sweet spot for listening and a rapid and mostly foolproof measurement/analysis cycle. I could not really confirm the former, but the latter was certainly true. The triangulation function is very slick, and the graphical display of mic positions made the process very straightforward.

The major shortcoming is the lack of time domain correction that can be had in packages such as the ARC 2.5 Room Correction System, recently reviewed by Ron Tipton (*audioXpress*, March 2018). But for me, at least, the differences in the resulting sound were close to non-existent in my near-field monitoring (where, really, this is speaker correction rather than room correction) and very slight in my use as room correction software.

So the choice really came down to ease of generating the correction curve and the feature set that could be used in mixing and mastering, where the advantage is clearly to the Sonarworks package. I very much appreciated the ability to set my own reference curve for the correction so that I could optimize the master with specific listening targets in mind. The Systemwide function in Sonarworks was also incredibly convenient, allowing me to easily use this package for normal music listening and do mixing and mastering without the risk of me forgetting to remove the plug-in when I was finished.

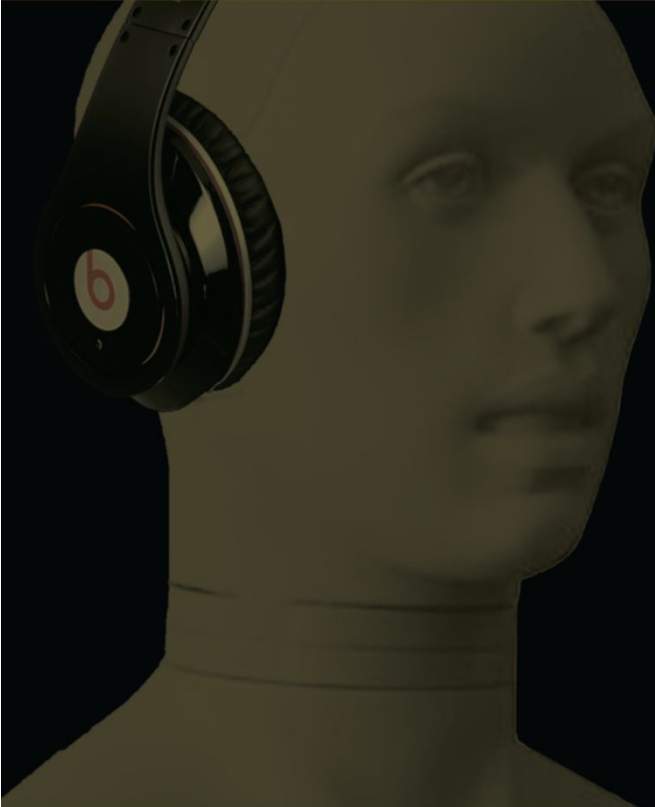
Though it's an enhancement, not a substitute for high-quality monitors and basic room treatment, for studio work the Reference 4 is a clear winner. Highly recommended. 

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Breathing New Life Into a DIY-Oriented Audio Company

An Interview with Kerry Armes and Dan Poinsett of Creative Sound Solutions

Creative Sound Solutions (CSS) is operated out of the Southeast Michigan area by two friends with a passion for great audio. Ownership of CSS was transferred from Bob Reimer, the original founder, to Kerry Armes and Dan Pointsett in June 2017 and they've been working on bringing back some of the drivers CSS is known for, as well as introducing some new kits and products to make it easier to get a world-class speaker at a great price.

By
Shannon Becker
(United States)



Kerry Armes (top) and Dan Poinsett revived Creative Sound Solutions after its original owner decided the time had come to close up shop.

SHANNON BECKER: Tell us a little about your background.

DAN: I grew up in the 1980s and enjoyed playing piano and coronet, but wasn't interested in recorded music until I was introduced to the previous generation's music—Led Zeppelin, The Who, The Beatles, etc. I then began collecting music and upgrading my audio system whenever I could, starting with a cheap boombox in middle school and working my way up to Adcom separates and Paradigm floorstanding speakers by the time I graduated high school. I attended Lawrence Technological University in Michigan where I received a bachelor's degree in Physics. My intention was to use my Physics degree to pursue a career in audio/acoustics, but right out of college I began work as an engineer in the additive manufacturing field and enjoyed it so much I stayed there, but continued studying loudspeaker development in

my free time. The manufacturing experience has been invaluable on the operations side of running Creative Sound Solutions.

KERRY: My formal education is in business. I have a B.S. in Business Administration with a minor in Industrial Design. I later got an MBA from IE Business School in Madrid. Most of my technical knowledge came from the eight years I spent in the Navy as a nuclear trained submarine officer. That's where I studied electrical and acoustical principles, which helped to provide the foundation needed to understand speaker design.

I think the first time I really noticed the sound quality of a speaker was when I was about 16 years old. I was walking through a mall and passed by an audio store and heard this great, clean bass coming from inside. I don't remember what subwoofer was playing, but I just remember thinking, "That is what I want. I want my music to sound like that."

SHANNON: How did you become interested in audio electronics?

KERRY: I've always been interested in music. I played trumpet from elementary school through a little bit of college and picked up the guitar somewhere in there as well. I always had some kind of stereo in my room as a kid to listen to music so I think that is one aspect of it. The other is that I've always liked the challenge of doing something myself and learning a new skill or trying to figure out how something works. My parents could probably tell you how many times I broke something as a kid trying to take it apart and failing to put it back together again.

Technically, I guess I got started doing car audio installs. Good speakers always interested me more than loud, which is what most of my friends cared about. While everyone else in high school was trying to get the most decibels from their car system, I wanted the best sound quality I could afford. Once I started looking into home audio, I found all these articles talking about concepts I had never heard of before around crossover design. When I was doing car audio, it was just plug and play. Home audio was a whole different realm and really took a lot of work to understand. Luckily, there are some great resources out there now (e.g., books from Vance Dickason and Floyd Toole as well as software packages for modest prices) so it is not hard to get started as long as you have the patience to keep learning.

DAN: As soon as I got hooked on popular music, albeit a couple decades late, I began experimenting with my cheap stereo systems to see how I could get them to play louder. We never had a proper stereo system in our house growing up, so I had no frame of reference—I just knew I wanted more. Some of these experiments consisted of changing speakers from one system to another. I quickly realized that speakers were the most limiting factor in an audio system and the one that most defines its sound, which greatly piqued my interest. At some point, I came across four smallish Panasonic speakers in damaged enclosures, each with a cone tweeter and a 5" to 6" woofer. With zero carpentry experience, I decided to construct new enclosures for the drivers. Likewise, with zero audio knowledge, a friend informed me that the larger I made the enclosure, the deeper the bass my speakers would produce. I made a pair of the largest boxes I could reasonably make at the time and put two woofers and a single tweeter in each enclosure. I fired them up and the bass was crazy!



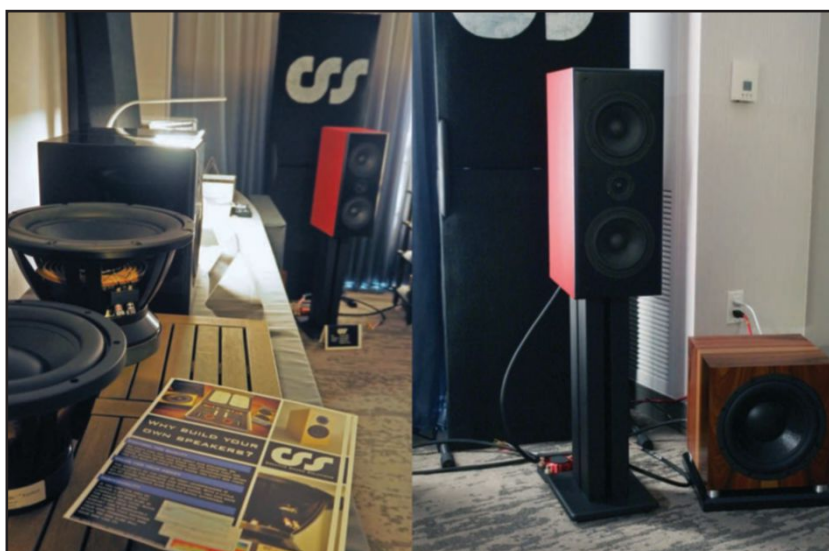
The new CSS P215 Kit uses a Satori MW 13P-8 Egyptian papyrus cone woofer and a CSS LD22 tweeter. The crossovers come with clearly labeled components and pre-formed/drilled fully labeled plates.

I was so enamored with the combination of woodworking, electronics, and physics that produced that sound, it started a lifelong obsession with great sound and speakers in particular. I immediately began reading and learning everything I could and then building and experimenting as much as time and money allowed. To this day, I'm still just as excited for the next experiment as I was 25 years ago.

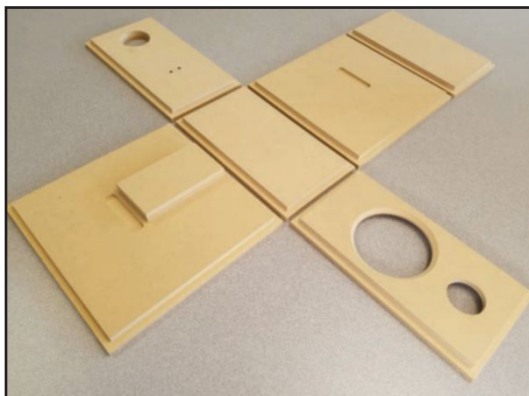
SHANNON: Was that your first audio project? Is it still in use?

DAN: It was, but it is not still in use and in fact was decommissioned shortly after its creation. The drivers were cheap and the enclosure atrocious and

Kits that are thoroughly designed, not to be the cheapest, but induce great results. Playing at AXPONA 2018 was the Criton 2TD v2.0 two-way D'Appolito loudspeaker kit with a SDX12 sub.



The knockdown cabinets are designed similarly to the SmartNode boards so that they only go together one way, which makes assembly easy even if you don't have experience in woodworking. This kit can easily be assembled, even in small apartments, so you can have access to great sound even if you don't have the tools to build it yourself.



I was quickly drawn to the books and raw drivers on the shelves at RadioShack. My next project wasn't much better, but it was a step in the right direction and at least had a little knowledge behind it.

SHANNON: Kerry, what was your first audio project?

KERRY: My first home-built speaker was actually a kit speaker someone else designed. I say kit, but it was really just a design I found on the Internet for a cheap single driver speaker with a baffle step circuit to try some things out. It didn't come all packaged together. All the parts had to be ordered online by looking them up myself. I think I ended up giving it to my parents as a gift when I started building more.

SHANNON: When did you first hear about Creative Sound Solutions (www.css-audio.com) and when and why did you decide to buy it?

KERRY: We both knew of Creative Sound Solutions for a while. The SDX series subwoofers had made a good name for themselves online in some of the places we

frequent. I had heard back in January of 2017 that Bob Reimer was having health issues and closing up shop. At the time, he was clearancing inventory to try and get rid of it. I called him up to discuss a potential option to buy the company outright.

Dan and I knew each other from some audio clubs we are both a part of and I had just moved close to him for my job. I discussed the idea with him and it just happened to be a good situation that worked out for both of us. We had both been looking for ways to start our own business and our dream job if we could pick one would be owning an audio company, so it seemed like a perfect match.

SHANNON: After almost a year of ownership, can you tell us what has changed with Creative Sound Solutions (CSS)?

KERRY: Well, the big one is that we are located in Michigan now instead of Canada. But there are numerous other things we've changed about the strategy for the company. The previous owner focused a lot on reselling existing products, whereas we are more interested in new development and CSS-branded products. We've also moved a lot of the design work in-house and are focusing on making high-end kits that are easily accessible even for a complete beginner who has no knowledge of how to read a circuit diagram or ability to do woodworking.

SHANNON: What are some of the challenges you have faced in your first year?

KERRY: One of the biggest is finding suppliers to work with when you are just starting out. Most have pretty high minimum order quantities for custom-built parts, which can be challenging when attempting to expand your product line.

SHANNON: And what has stayed the same? Are any of the products sold under the previous owner still available?

KERRY: Most of the CSS-branded drivers Bob sold were very good but many had gone out of production before we took over. We kept the SDX10 and the SDX12 subwoofers with their matching passives and the LD22 and the LDW7.

We really knew the company from its SDX series subwoofers. They have amazingly low distortion and the Thiele-Small (T-S) parameters are about as ideal as you could want. We hadn't really heard anything about the LD22 and the LDW7 before we took over the company. When we got the inventory in hand, we did some testing with them and were really impressed

The CSS P215 finished speakers show the company's passion for speaker design, with affordable options for those who want to have assured quality results.



by what both had to offer. They are well-built, low distortion drivers that easily compete with other high-dollar offerings on the market.

SHANNON: Can you tell us about some of the new products you have to offer?

KERRY: We kept the Criton line of kits but we changed the crossover from what the previous owner used to sell. We thought there were improvements that could be made from a redesign and so far everyone who has heard them agrees. We also introduced the completely new P215 kit using the 5" Satori woofer.

All of our kits come standard with our unique SmartNode crossover boards. These boards were designed to make crossover assembly easy even if you don't know the first thing about electrical circuits. Components have designated spots on the board and labels underneath detailing how to connect everything. We considered using PCB boards put kind of felt that took away from some of the fun of putting the kit together. This solution offered basically the same reduction in assembly error with a more engaging build for the customer.

SHANNON: What products are you most proud of?

KERRY: We are very proud of our kits overall. We put a lot of effort into making them something that looks good and is easy to assemble. The knockdown cabinets are designed similarly to the SmartNode

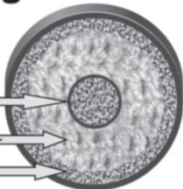
boards so that they only go together one way. In addition, we didn't scrimp on the sound at all. We recently were named one of the top 5 under \$20,000 at AXPONA by *The Absolute Sound* for our P215 speaker. The components we use are all much higher quality than you can get anywhere close to this price point because we are reducing labor and overhead around the cabinets. *The Absolute Sound* noted that even if you buy our finished offering you are getting a great speaker for the price. At the kit price, it is an amazing deal.

SHANNON: What's next for CSS?

KERRY: Outside of growing our brand awareness and business as a whole, we have basically three things in our long-term plan. The first is to expand our in-house driver line so that we have a focused and easily understandable product line. The ultimate goal would be to have an ultra high-end XBL² offering for our tweeter, mid, midwoofer, and subs and expand the second-tier offering to have more traditional motors with low distortion like our LDW7. We are looking specifically to bring back modified versions of some of the old drivers CSS used to carry like the LD25X. The second would be to expand the kit offerings to include a large three-way and finish our Criton 2TD with all the bells and whistles of our other kits. We're also working on knockdown cabinets for a passive radiator version of our subs. The third is to start carrying custom amplification for our subwoofers. 



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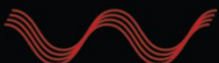
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Current Feedback: Fake News or the Real Deal? (Part 1)

Untangling the Signal Path

Recently, the very existence of current feedback as opposed to voltage feedback has been called into question. This article employs bench tests, simulations, and equations derived from various circuits under investigation to address this controversy.

By
Christopher Paul

(United States)

Audio electronics is no stranger to controversies. Several examples come to mind.

First, it's generally agreed that if a circuit's output has a voltage, the impedance of that output can be determined. First, the voltage is measured. Then the output is connected to a load of known value and the voltage is measured again. A simple calculation reveals the impedance. But we can also set the output's signal to zero and drive that output with a current. The ratio of the resultant voltage to that current gives another impedance. Must these "output" and "input" impedances be identical? Thevenin's theorem provides the answer.

Second, the ground-referenced impedance at a Cathodyne phase splitter's cathode is widely accepted to be somewhere near $1/g_m$, the reciprocal of the triode transconductance. But is the impedance at the anode the same? B+ power supply noise rejection tests at that electrode answer the question.

Third, recently, the existence of current (as opposed to voltage) feedback has been challenged, specifically in ICs that the semiconductor industry has come to call "Current Feedback Amplifiers." In this article, I hope to shed some light on this wrangle.

A Diversionary Thought Experiment

Before we consider feedback, let's engage with something simpler. There is a functional equivalency

between certain signal sources. A voltage source V in series with impedance Z is indistinguishable from a current source I in parallel with Z —that is, if we exclude from $Z = V / I$ values of zero and infinity. Suppose that one of these sources drives a load L . Does the load see more of one source type than the other? Well, if L is less than Z , it's a better approximation to say that a current I flows through it than that a voltage V appears across it. The reverse is true if L is greater than Z . And if Z equals L , one approximation is as good (or as bad) as the other. We'll see that this relationship of source to load impedances and that of amplifier feedback networks to feedback inputs are similar.

Some Amplifier Input Structures

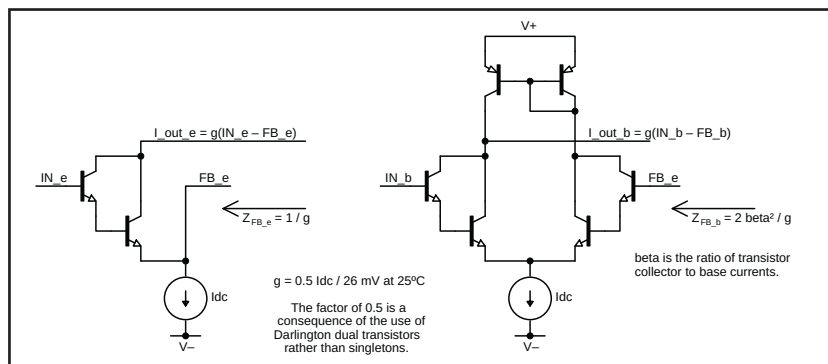
Figure 1 shows two different Darlington pair input stages. Let's refer to them by the junction to which feedback is applied: e-fed (emitter-fed) and b-fed (base-fed). The transconductance term g is approximated as shown in Figure 1. Perhaps surprisingly, each stage has the same value of g when all Darlington pairs are identically biased. The e-fed Darlington sees the full differential voltage applied between FB and IN. Although each b-fed Darlington sees half of that, the PNP current mirror adds these b-fed half currents. And so each stage offers the same signal output current. Let's follow

these currents to their sources. In the b-fed case, they come from “steering” the I_{dc} bias current. Negligible currents arrive from the FB (feedback) base—hence, voltage (no current) feedback. But in the e-fed case, no steering of I_{dc} is possible—the feedback portion of the signal current is sourced exclusively from the amplifier’s output through the feedback network. This is one possible definition of current feedback.

Consider the impedances of the FB inputs. The e-fed circuit’s is approximately $1/g \, \Omega$ (see the Figure 1.) The b-fed’s impedance is larger by a factor of β^2 , where β (often greater than 100) is the ratio of a transistor’s collector to base current. With all Darlington’s biased at 1 mA, $Z_{FB_e} \approx 52 \, \Omega$, while Z_{FB_b} is much larger. Such different input stages can lead to amplifiers with some noticeably different characteristics.

Amplifiers with E-fed and B-fed Input Stages

In Figure 2, the e-fed amplifier is on the left and the b-fed one is on the right. For functionally corresponding components, e-fed and b-fed reference designators differ only in whether they contain an “e” or a “b”. Henceforth, when possible, we will generalize by suppressing those letters. Current sources are implemented with R_1 and Q_5 . R_8 through R_{11} , C_2 and U_2 form servos that keep



Out near 0 VDC. U_1 provides gain and low impedance outputs. +9 VDC at the U_1 (+) inputs establishes +9 V at the Q_1 and Q_2 collectors. R_{b7} works with R_{b4} to help shift Out close to 0 VDC. We shall see later that R_4 and C_3 have been selected to provide matching loop gains for the two amplifiers at 100% feedback (infinite R_6 .) For the moment, we’ll set C_3 and all sources with “middle” in their names to zero. This makes the voltage sources short circuits and the current sources open ones.

And now for something completely different (with apologies to Monty Python.) In analyzing these designs, I believe it would be wise to take guidance from an episode of *The Adventures of Young Indiana Jones*. In the 1920s, Indiana Jones (Indy) was quite taken with Jazz. He had an opportunity to play

Figure 1: Input stages to which feedback is applied to an emitter (left, e-fed) and a base (right, b-fed).

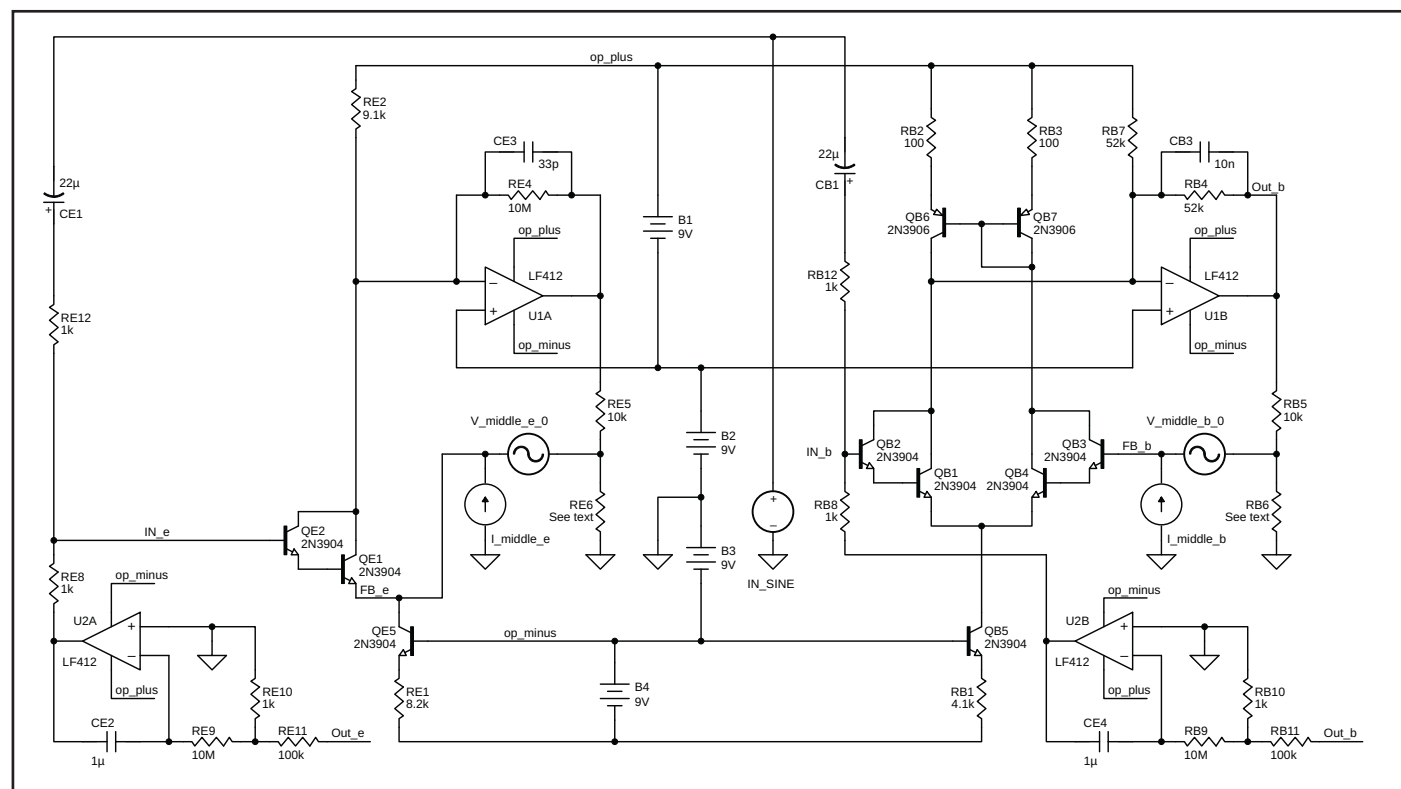


Figure 2: Amplifiers composed of e-fed (left) and b-fed (right) input stages

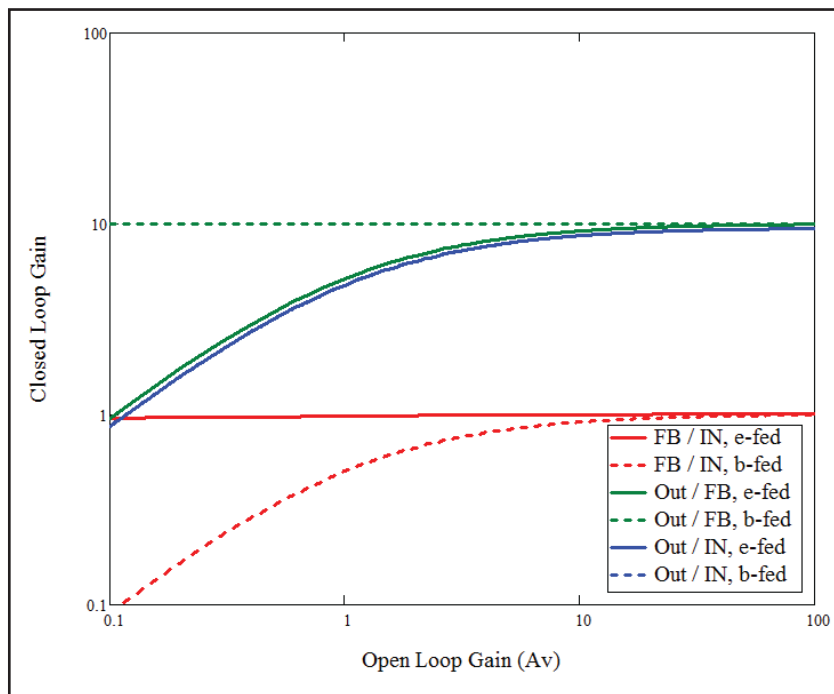


Figure 3: Graphs of equations 5 through 7 as a function of A_v for R_4 as shown in Figure 2 and $R_6 = 1100 \Omega$.

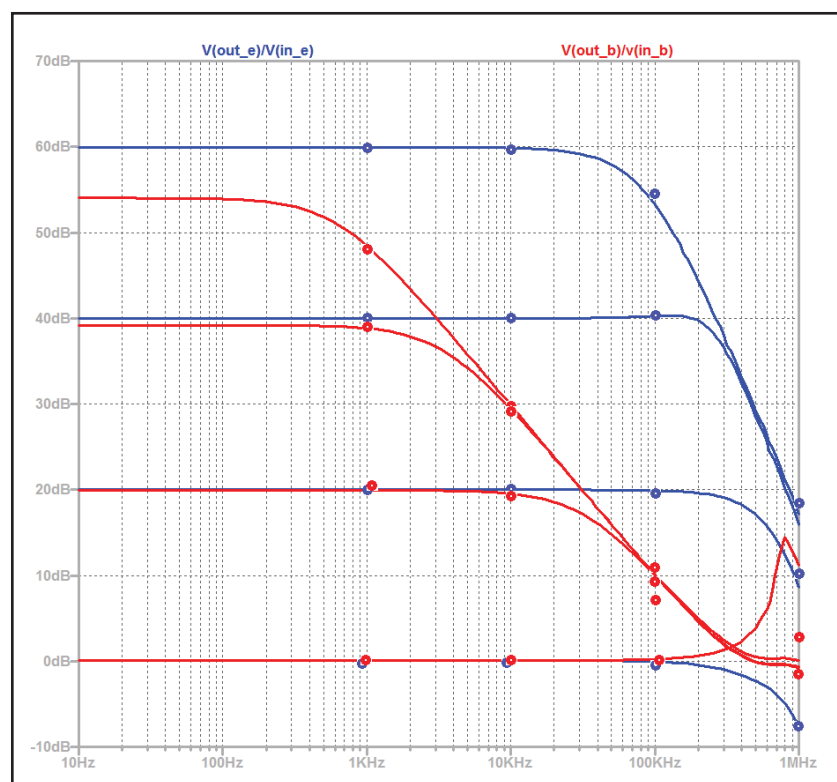


Figure 4: Measured and calculated closed loop gains of the b-fed and e-fed amplifiers shown in Figure 2 for $G = 1 + R_5/R_6 = 0, 20, 40, \text{ and } 60 \text{ dB}$.

saxophone with some of the greats of that era. But they remonstrated him for not knowing a song well enough to improvise on it. He was told, "You've got to know it forward and backward, inside out and upside down before you can understand it well enough to play it," or something to that effect. Let's follow that advice.

We can determine the open loop gains A_v of the Figure 2 circuits by inspection by setting the source IN_SINE to zero, disconnecting R_5 from Out, and driving R_5 from a voltage source V_{test} (not shown.) We obtain:

$$A_v = \frac{Out}{V_{test}} = g \times R_4 \times \frac{R_b}{R_b + R_5} \quad [1]$$

where $R_b = R_6 \parallel Z_{FB}$

Solving this equation for R_4 while setting R_6 to infinity and A_v to 1,000 allows us to calculate the values for the R_4 s that you see in the schematic:

$$R_4 = \frac{A_v \times (Z_{FB} + R_5)}{g \times Z_{FB}} \quad [2]$$

For the following equations, the first is obvious and the second comes from conservation of current at the FB node:

$$Out = g \times R_4 \times (IN - FB) \quad [3]$$

$$\frac{FB}{R_6} + \frac{(FB - Out)}{R_5} + \frac{(FB - IN)}{Z_{FB}} = 0 \quad [4]$$

From the above, we can derive three equations of interest:

$$\frac{FB}{IN} = \frac{1}{1 + \frac{1}{A_v + \frac{(A_v + 1)}{Z_{FB} \times \left(\frac{1}{R_5} + \frac{1}{R_6} \right)}}} \quad [5]$$

$$\frac{Out}{FB} = \frac{1 + \frac{R_5}{R_6}}{1 + \frac{1}{A_v \times \left[1 + \frac{Z_{FB} + Z_{FB}}{R_5 + R_6} \right]}} \quad [6]$$

$$\frac{Out}{IN} = \frac{\left(\frac{R_5}{R_6} + 1 \right)}{\left(1 + \frac{1}{A_v} \right)} \quad [7]$$

Figure 3 shows these as dashed (b-fed) and solid (e-fed) lines as A_v varies, with R_6 arbitrarily set to 1100Ω to make $G = 1 + R_5/R_6 \approx 10$. In these

circuits, Z_{FB_e} measures 52 Ω and $Z_{FB_b} \gg R_6$.

At high values of A_v , results are what we expect: FB is approximately equal to IN, and Out is $G \approx 10$ times both. But look at what happens at low gains. The blue Out/ V_{in} traces are indistinguishable because A_v varies in lockstep for these two circuits. They also coincide with the solid green OUT/FB e-fed curve (I've purposely offset the blues slightly in the graph so that both they and the solid green can be seen.) The red dashed b-fed FB/IN is very much less than unity, though it remains approximately 1 throughout in the solid red e-fed case. The Out/FB dashed green b-fed curve holds steady at 10, but the solid green e-fed one falls to 1. The differences between these circuits, while obvious at low open loop gains, are simply masked at the high A_v to G ratios, which we are well advised to employ.

Combining equations [3] and [4] yields:

$$\frac{\text{Out}}{\text{IN}} = \frac{\left(\frac{R_5}{R_6} + 1 \right)}{\left[\frac{\frac{R_5}{Z_{FB}} + \frac{R_5}{R_6} + 1}{1 + \frac{g \times R_4}{R_6}} \right]} \quad [8]$$

The $K = g \times R_4$ term is the voltage gain element in our design. Since we desire Out/IN to equal G , good practice requires K to be large enough that the term in the square brackets is very close to unity. However, when we account for the effects of C_3 , (see Figure 3; C_3 is in parallel with R_4 and therefore affects K), we see that K falls with frequency. Varying G by varying R_5 will affect the coefficient of the $1/K$ term. This will change the frequency at which our "good practice" falters. However, the small value of Z_{FB} in the e-fed design affords us an opportunity. We simply hold R_5 constant and vary R_6 . While R_6 remains greater than Z_{FB} , there is little variation in the coefficient of $1/K$ and therefore, little variation in bandwidth. Additionally, equation [1] shows us that this keeps the loop gain relatively constant. This strategy is unfortunately unavailable in the b-fed case. Here, Z_{FB} is typically quite large, R_5/Z_{FB} is a lot less than 1, and the $1/K$ coefficient varies with G . That explains the constant gain-bandwidth product typical of b-fed input stage designs.

The simulation in **Figure 4** of the Figure 2 circuits accounts for the effects of C_3 (we hold $R_4 \times C_3$ equal for both amplifiers) and demonstrates the results as we vary R_6 so that $G = 0, 20, 40$, and 60 dB. The small circles on the graph represent values measured on the test bench, where the Q1-Q2 and Q3-Q4 combinations are replaced with single package Darlingtons, BC517s. The peak of the b-fed curve in Figure 4 around 1 MHz is due to inadequate

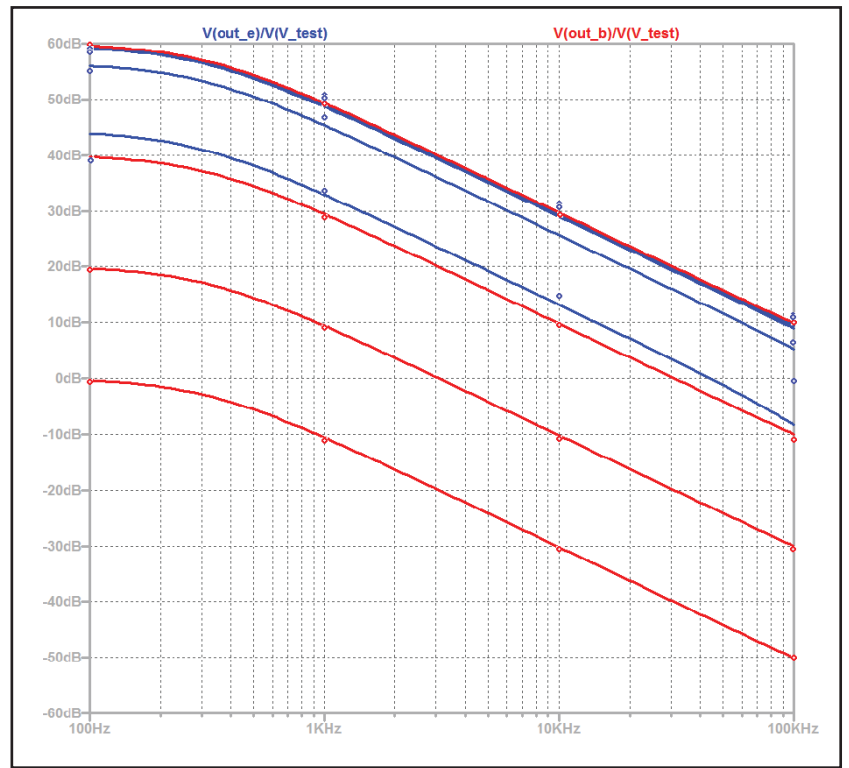


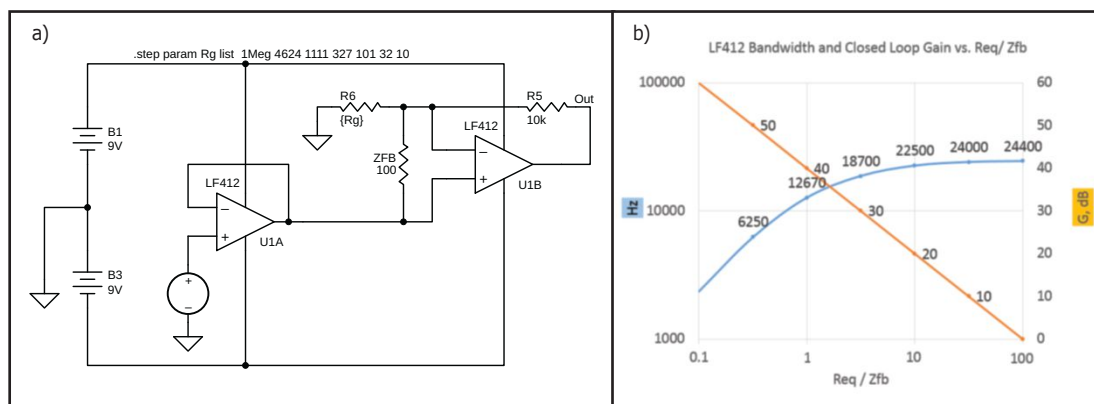
Figure 5: Measured and calculated open loop gains of the b-fed and e-fed amplifiers of Figure 2. IN_SINE is set to zero and R_5 is disconnected from Out and driven by a separate signal source, V_{test} (not shown).

compensation. Overall, the curves are a bit difficult to believe. Did we really design for the same A_v in both amplifiers? We can check this by zeroing IN_SINE, disconnecting the R_5 s from the Outs and driving the R_5 s from a separate signal source V_{test} (again, not shown). The simulation and bench test results appear in **Figure 5**.

The top blue (e-fed) and red (b-fed) curves in Figure 5 (which correspond to $R_6 = \text{infinity}$ and $G = 0$ dB) confirm that the two amplifier open loop gains are practically identical under these circumstances, and are 1000 = 60 dB at low frequencies. The b-fed amplifier behavior is unremarkable. Its response is inversely proportional to $G = R_5/R_6 + 1$, as expected. But look at the e-fed performance! A_v does not fall as $1/G$ as G increases as we would expect, but rather much more slowly. And in Figure 4, it reaches a closed loop gain of exactly 60 dB, even though A_v itself is only 60 dB (when $R_6 = \text{infinity}$.) What is going on here?

The e-fed performance can be explained by the feedback network looking like a degenerative resistor to Q_1 . Unlike in the b-fed case, it prevents the full voltage $\text{IN} - \text{Out}/G$ from appearing across Q_1 and Q_2 . But as we reduce G by reducing R_6 , we also reduce the value of the degenerative resistance. More of that voltage appears across Q_1 and Q_2 . This acts to increase the gain generated by this Darlington, partially counteracting the fall in loop gain. It is

Figure 6: (a) Closed loop gains of an LF412 op-amp for $G = 1 + R_5/R_6 = 0, 10, 20, 30, 40, 50,$ and 60 dB. (b) Z_{FB} drastically lowers the impedance at the inverting input of this b-fed-like op-amp.



almost anticlimactic to note that Figure 4 confirms that the e-fed design has a nearly constant bandwidth for $G = 0$ dB ($R_6 \parallel R_5 = \text{infinity} \parallel 10 \text{ k}\Omega$) and 20 dB ($R_6 \parallel R_5 = 1100 \parallel 10 \text{ k}\Omega$.) Both resistances are greater than $1/g = 52 \Omega$. But bandwidth begins to fall when $R_6 \parallel R_5$ is $100 \Omega \parallel 10 \text{ k}\Omega$ and more dramatically when $10 \Omega \parallel 10 \text{ k}\Omega$, which are comparable to or less than $1/g$. This evokes our “diversionary thought experiment” at the start of the article and brings us to...

Indy's Final Gambit

There's at least one more way to look at these designs. In 1975, R.A. Middlebrook described methods to precisely determine circuit loop gains. He established the following relationship between the total loop gain T of a given circuit, and its purely voltage T_v and current T_i loop gains:

$$(1+T)^{-1} = (1+T_i)^{-1} + (1+T_v)^{-1} \quad [9]$$

There are a number of things that surprise about this relationship. First, although the value of T is unchanged by the chosen point of analysis within a given loop, this selection will affect T_v and T_i (although [9] will still hold.) For this reason, we apply our analysis at FB where we find the question of feedback type to be relevant. Second, because

the applicable terms are in the denominators, the smaller of the voltage and current loop gains will have the stronger influence on the total loop gain T . Referring again to Figure 2, Middlebrook requires us to zero IN_SINE and to alternately activate V_middle and zero I_middle , and then to activate I_middle and zero V_middle . When V_middle is active, T_v is the negative of the ratio of the voltage at the junction of R_5 and R_6 ($FB - V_middle$) to that at the emitter of Q_1 (FB). To determine T_v , we first write the following current conservation equation at the FB node:

$$\frac{FB - 0}{Z_{FB}} + \frac{FB - V_middle}{R_6} + \frac{FB - V_middle - Out}{R_5} = 0 \quad [10]$$

Combining equation [3] (with $IN_SINE = 0$) and [10], we obtain:

$$T_v = \left(g \times R_4 + \frac{R_5}{Z_{FB}} \right) \times \left(\frac{R_6}{R_6 + R_5} \right) \quad [11]$$

With only I_middle active, T_i is the negative of the ratio of the current flowing from the junction of R_5 and R_6 ($FB / Z_{FB} - I_middle$) to that flowing into the emitter of Q_1 (FB / Z_{FB}). To determine T_i , we first write the current conservation equation at the FB node:

$$\frac{FB - 0}{Z_{FB}} + \frac{FB}{R_6} + \frac{FB - Out}{R_5} - I_middle = 0 \quad [12]$$

Combining equation [3] (again with $IN_SINE = 0$) with [12], we arrive at:

$$T_i = \left(1 + \frac{R_5}{R_6} + g \times R_4 \right) \times \frac{Z_{FB}}{R_5} \quad [13]$$

The term responsible for amplifier gain is $g \times R_4 = K$. When equations [11] and [13] describe properly designed circuits, at low and moderate frequencies K dwarves any term with which it is summed. Under these conditions, we see that:

$$\frac{T_v}{T_i} \approx \frac{Req}{Z_{FB}}, \text{ where } Req = R_5 \parallel R_6 \quad [14]$$

About the Author

Christopher Paul was born in the US the year before the Nobel prize was awarded for the invention of the transistor, the development of which he claims no credit for. His love of music from an early age led to some puzzling experiments with musical instrument amplifiers, and from there to an interest in electronics. Chris finds it interesting to delve into relatively simple circuits and to investigate and write about aspects of their performance that are little known or appreciated. He enjoys deriving equations that can be used to evaluate and design them, although he admits to being in therapy for this. Chris has worked for companies whose products are in the fields of communications, electronic article surveillance, and enterprise hand-held computers. He holds two M.Sc. degrees in electrical engineering from Brooklyn Polytechnic (now part of New York) University.

This tells us that when $R_{eq} > Z_{FB}$, the voltage loop gain exceeds the current loop gain. And when that inequality is reversed, the current loop gain is larger. However, the larger the loop gain, the less the feedback. And so, $R_{eq} > Z_{FB}$ implies that current feedback predominates at this point in the circuit and is the major influence on total loop gain. And, $R_{eq} < Z_{FB}$ implies that voltage feedback predominates at this point.

These results lead to an interesting question: Are the types of feedback and associated phenomena dependent on the kind of input stage, or only on the above inequality? Easy enough to find out...

Hold On to Your Hats

Figure 6 shows the results of a simple bench test. Start with an amplifier with a b-fed-like input—the LF412 (actually, its JFET inputs draw even less current than a b-fed.) Can we coax it to behave as if it had an e-fed input? The graph answers, “yes.” Bandwidth is practically constant while $R_{eq} = R_5$ || $R_6 > Z_{FB}$, but falls with R_{eq} when that inequality fails.

Apparently, circuits have distinctly different characteristics depending on this inequality, regardless of input stage type.

Current Feedback Shakes Another Tree

Many of us are familiar with the four types of signal routings shown in **Figure 7**. The source of the feedback is the output, either the voltage across the load (shunt-derived) or the current through it (series-derived). But what is the feedback destination? It must be the op-amp feedback (inverting) input, right? There is no other place to “feed back” a signal to that will influence circuit operation! Typically, we are told that series applied is voltage feedback. So voltage is applied, but no current flows into the FB input, right? Wrong, if this is an e-fed device. The signal current flowing through such a transistor must come through the feedback network only. So how can this be exclusively voltage feedback? We are also told that shunt applied is current feedback. So this current flows into the op amp FB input, right? Not if we are working with a b-fed or FET input device (which accepts negligible current) it isn’t. So how is this current feedback?

My takeaway is this: Describing signal routing as series/shunt/applied/derived is appropriate and a sufficient means of distinction. Insisting that each type is always associated with only current or only voltage feedback belies the differing natures of op-amps. It’s bad enough that “current feedback” could be defined alternatively by input stage type or by Middlebrookian inequalities—consider that

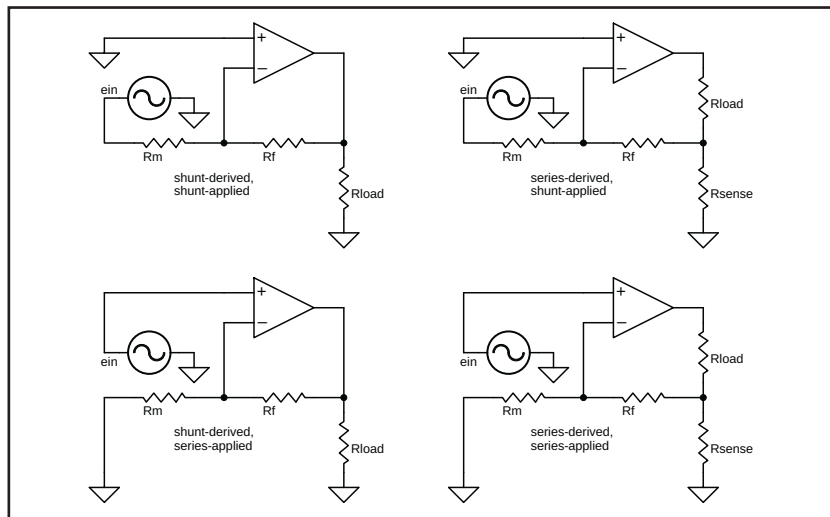


Figure 7: The four arrangements of signal routings

to some, it has also meant load current (series derived) feedback!

So Where Does This Leave Us?

Regardless of input stage type, while $R_{eq} > Z_{FB}$, we can maintain a constant bandwidth while varying the circuit’s closed loop gain by adjusting only R_6 . This is easier if the amplifier has an e-fed rather than a b-fed input stage, but it is not strictly necessary. With e-fed stages, as G is increased by reducing R_6 , loop gain falls more slowly than the $1/G$ of the b-fed case. This greater than expected loop gain likely has beneficial effects when it comes to limiting the increase in distortion at high closed loop gains, at least from the output stage.

The matching of A_v at $G = 0$ dB for the two input stage type amplifiers was done for purposes of comparison. Nothing presented here is meant to suggest that one type of input stage inherently leads to wider bandwidth or better performance in an amplifier.

In the second part of this article, we’ll look at extensions of these input stage types as employed in modern op-amps and see how they lead to even further distinctions in performance. Op-amps built around some of these are what the industry has taken to calling Current and Voltage Feedback Amplifiers. We’ll also delve further into feedback types. 

Transformers for Tube Amplifiers (Part 3)

Choosing the Right Transformer

In Part 1 of this article series we discussed output transformers and their fundamental properties. In Part 2, we discussed the ideal core sizes and core types. In this final article of the series, we will discuss long-term stability and how to choose the right transformer.

By
Gerhard Haas
(Germany)



A very important point concerning a transformer's long-term stability is often disregarded. Around 1970, the Radio Corporation of America (RCA)—when it still manufactured tubes and semiconductor components—carried out fundamental examinations on thermal cycles and the expected service life of power semiconductors. What applies to semiconductors, also applies to transformers and other components.

When a component at room temperature is put into operation, it heats up to a maximum temperature defined by the given circumstances. After it is switched off, it cools down to room temperature again. Now, we can verify how often a component can go through this cycle until it becomes unusable. It is a mistake to think that a transformer will withstand that forever. A transformer is usually made from three main materials: iron core, copper wire winding, and insulation—the latter also includes the coil body. Now, take into consideration that copper expands by about 48% more than pure iron when heating. The insulation must compensate for the copper expansion. If crossing wires are present, as with toroidal core winding or inaccurate layer winding, those are potential sources for shorts as mechanical

pressure occurs. The higher the temperature differences between idle condition and operating condition, the higher the pressure differences. Furthermore, it is also possible that the insulation layer is pushed through causing a winding short over all the layers. In this case, the transformer becomes hot because many windings are shorted and the electrical energy is converted into heat. If no fuse is tripped, the transformer will literally go up in smoke.

If mechanical pressure is applied to the insulation, they start to flow. Everyone knows this experiment from physics class. When a weight on a thin wire loop is attached to an ice block, the loop will slowly go through the ice block. The ice will melt because of the pressure applied by the weight hanging on the thin wire. Since physics applies to everything, this principle also applies to a transformer winding. The insulation will deform due to the occurring pressure and the increased operating temperatures. The weak point of the crossing wires is eliminated by winding layer-wise with layer insulation.

In connection with the laminations and the core dimensions, the primary inductance needs to be given

close attention. The primary inductance is important for reaching the lower limit frequency. There are several formulas to determine the right inductance. Some are relatively complicated and require exact data about the core material, which is often not available to the desired extent. Therefore, a workable formula is shown, which is applied to the amplifier example:

$$L_p = \frac{Ra}{2 \times \pi \times f_u}$$

L_p is the primary inductance. R_a is the primary impedance. And, f_u is the lower limit frequency. This formula is intended for a 3 dB drop at the lower frequency. The primary inductance must already be present for low modulation in order to reach the lower limit frequency. It is also referred to as the initial inductance. The primary inductance L_p and R_a represent a frequency dependent voltage divider the behavior of which is shown in Figure 5 from Part 1 of the article series. When assuming $R_a = 1.9 \text{ k}\Omega$ and a lower limit frequency of 20 Hz (−3 dB), L_p must have the minimum value of:

$$L_p = \frac{Ra}{2 \times \pi \times f_u} = \frac{1.9 \text{ k}\Omega}{2 \times \pi \times 20 \text{ Hz}} = 15.13 \text{ H}$$

If we compare the calculated value with the values from the transformer B-434 S, you will see that this transformer shows a very high primary inductance at 100 mW. Therefore, it is able to transmit 20 Hz at full power without any noteworthy drop. Even before the level significantly drops at low frequencies, the core is saturated.

The question arises as to what causes this high primary inductance of this transformer. It depends on the permeability of the iron and on the number of primary windings. Due to the sheet quality, the grain-oriented sheets feature a higher permeability than EI- or M-type cores. In contrast to EI- and M-type sheets, MD-type sheets feature only a hair-thin gap. Overall the EI-type core is expected to feature an inevitable air gap. The larger the air gap, the higher the magnetic negative feedback, which reduces the maximum inductance possible.

Those trying to measure the primary inductance using an LCR meter will fail. The iron's permeability depends on the modulation. It increases with the current until it reaches a maximum in order to drop again with higher modulation. Analogously, the same applies to mains filter chokes. A specific current must be sent to the coil in order to specifically excite the iron. The desired inductance values are only reached in this way. The LCR meter, which supplies a measuring voltage of some 100 mV up to 1 V, is not

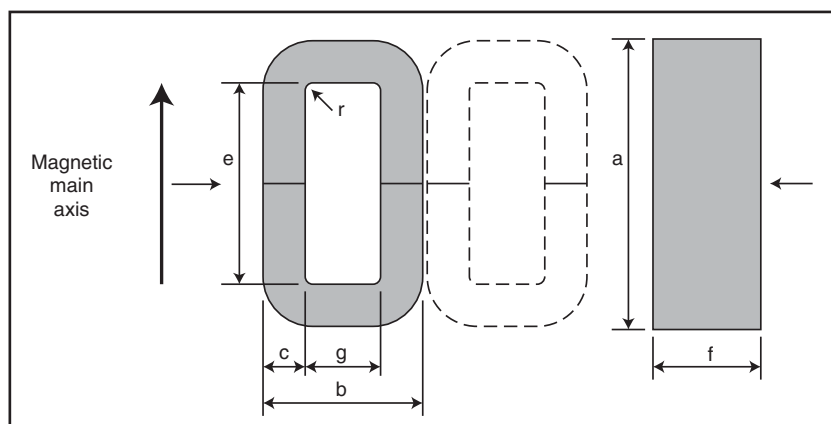


Figure 1: The basic structure of a cut-tape wound core (C-core, respectively double C-core) is shown.

even able to sufficiently excite a small iron core—you can measure something, but nothing realistic. Air coils without iron core and small ferrite core coils can be measured with LCR meters.

The transformation ratio of a transformer also has a big influence on the achievable final result. Not only the resistances, but also the capacitances are translated by the square of the transformation ratio. Consequently, the capacitances have an increasing effect with increasing transformation ratio. Then, the frequency response at the upper band end can be limited so that it has an audible effect. Furthermore, a waviness in the frequency response, resonance points with alternating increases, and level drops as well as signal distortion may occur together or individually.

On the other hand, a higher number of windings is required on the secondary side in case of lower transformation ratios. In this case, the copper resistance increases on the secondary side, resulting in power losses. So, if the copper resistance of the secondary winding is 30% to 50% of the speaker impedance, a substantial part of the amplifier power inside the transformer is converted into heat instead of being properly used. So, under certain circumstances a much larger core than is actually necessary must be used to increase the winding space, reduce the number of windings, and reduce the copper resistance.

The examples referred to in this article apply to single-ended type-A and push-pull AB transformers. Provided that with single-ended type-A transformers the lower limit frequency of 20 Hz shall be reached at full modulation, a correspondingly larger core must be chosen. Because of the direct current bias of the iron core, an air gap needs to be provided for compensation. Due to the air gap, the effective permeability of the iron core is reduced, which needs to be compensated by a higher number of windings. However, the copper resistance increases,

Specifications of the A-6C33 Transformer	
Frequency response	20 Hz to 20 kHz (-0.2 dB)
	90 kHz (-3 dB)
Primary inductance	>48 H at 200 mW
Leakage inductance	4.6 mH
Ra	600 Ω
Secondary impedances	4 Ω , 8 Ω
Copper resistance of the primary winding	115 Ω
Core	SM 102 B (SM = double C-core)
Dimensions	
Mounting frame	113 mm $\times$ 113 mm
Height above coil body	92 mm
Weight	approx. 4 kg

Table 1: The capabilities of a well-designed single-ended type-A transformer are displayed.

which also requires a larger core. Also, EI-type cores are preferable as an air gap can be easily inserted between the E and the I.

Cut-tape wound cores, however, are much better, because the magnetic load of these cores can be much higher and they are normally made from a very thin and low-loss M097-30 strip material measuring only 0.3 mm in thickness. In this case, the losses are only 0.97 W per kilogram of iron. **Figure 1** shows the basic structure of a cut-tape wound core (C-core, respectively double C-core). The long arrow in the picture shows the magnetic main axis. The capabilities of a well-designed single-ended type-A transformer can be seen in the datasheet of the A-6C33 transformer, using the 6C33 tube (see **Table 1**).

The degree of nesting a transformer winding is an issue, too. It is often said that the higher the degree of nesting of the winding the better the transformer. Now, the question is: Is it the actual degree of nesting or the end result that is important to the user? The user can either believe the supplier as to the degree of nesting of a transformer, or check it by disassembling it. However, in this case, the transformer is ruined in the process. Now, the user knows its degree of nesting but the transformer can no longer be used.

Finding the Ideal Transformer

There are many ways to achieve a good frequency response. The path is not important, but the goal is. So, if a transformer has a linear frequency response over the entire audio range and beyond that, it is not important how nested the winding is. It is also not important whether a low-frequency transformer features 50 kHz, 70 kHz, or 100 kHz as its upper limit

frequency. However, it is important that no significant drop occurs at 20 kHz.

What is even more important and constantly underrated is the lower limit frequency. Whether a transformer reaches 40 Hz or 20 Hz at full power, makes all the difference! The lowest note occurring in music is the subcontra C with 16.67 Hz, which can also be generated in original quality in large churches and using the appropriate organs. The E string of an electric bass guitar features 41.2 Hz as the lowest frequency, the five-string electric bass guitar reaches 30.87 Hz. Musically speaking, the frequency jump from 40 Hz to 20 Hz is a whole octave! Here, making efforts is more valuable than discussing whether a transformer is supposed to reach 70 kHz or 100 kHz.

If you want to get from 40 Hz to 20 Hz at full power, that means the transformer core needs to have double the cross-section or the number of windings needs to be doubled when leaving the cross-section unchanged. The truth is often found in the middle. Making the core arbitrarily large is uneconomic and inconvenient. Just doubling the number of windings results in a high copper resistance. So, the art of the transformer developer is to find an ideal compromise. Fundamental knowledge, knowledge of material characteristics, a sophisticated winding technique, dealing with more unknowns than equations, much practical experience, and more are required to build good transformers.

The ideal output transformer as well as the mains transformer basically have neither iron nor copper losses. As already described, you could make total material usage to reduce the losses, but that would be uneconomical. Some copper resistance may be allowed. With our model amplifier featuring the B-434 S transformer, a voltage drop of 9 V to 10 V per primary winding half would be bearable, which with an anode current of 100 mA would correspond to just 2 W. An EL 34, for example, reaches over 45 W of power loss at maximum heating power! The missing 10 V can easily be compensated by a slight increase of the operating voltage. With transformer A-6C33, the voltage drop would be about 25 V to 30 V, corresponding to about 5 W to 7 W, which will barely heat up the transformer. Also, the voltage drop can be easily compensated by a higher operating voltage.

Besides the maximum transmissible power, the frequency range a transformer masters is an important criterion. In this respect, it is crucial how the frequency response was measured. When measuring with a sine generator that features a source resistance of 50 Ω , at a power of 1 W, at 10 W, at high power, or under original conditions with the tubes provided for this purpose. Datasheets



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10 Hz	-22 dB	-0.6 dB
20 Hz	-13 dB	-0.2 dB
30 Hz	-3 dB	-0.1 dB
100 Hz	- 3 dB	0 dB
1 kHz	0 dB	0 dB
100 kHz	-1 dB	-1.4 dB
300 kHz	-3 dB	-2 dB

Table 2: The transformer measurement was measured with a source resistance of 50 Ω , a real resistance with 8 Ω was used as the load.

and brochures may, for example, list frequency response from 0.4 Hz to 187 kHz (-3 dB), or 0.8 Hz to 120 kHz (-1 dB), featuring even three decimal places. In other words, the transformer can almost transmit direct current. Furthermore, a very exact frequency response information with several decimal places simulates accuracies that can never be adhered to. Wires, insulating foils, and cores have tolerances. Equally, there are deviations during winding. Specifications rounded off to the worst values would be absolutely sufficient.

However, when reading the small print, you will see that these fantastic frequency responses were measured at 1 mW using a signal generator. When measuring under original conditions, using the intended tubes under load, clearly worse but real and usable values will be the result! At 1 mW the iron core is not excited. The winding and wiring capacitances have only minimal impact. When measuring under real conditions, such a “wonder transformer” suddenly has a frequency response of >40 Hz to <120 kHz (-3 dB). In this case, it is also no use if a toroidal core or an oversized EI-type core was used.

However, an interesting measurement with a transformer was made, which normally is not possible for the user because a transformer is bought as a ready-to-use component. If this transformer is disassembled for such a measurement, it will be damaged and useless—expensive fun. But a typical output transformer for tube power amplifiers was used as measuring object. It was measured at 1 mW—one measurement was done with the winding without the core and the other measurement included the core. The source was a signal generator with a source resistance of 50 Ω , a real resistance with 8 Ω was used as the load. The frequency response at 1 mW referring to 1 kHz can be seen in **Table 2** (the values are rounded).

As you can see from Table 2, the differences at the upper end of the frequency are minimal, no matter whether the core is inside the winding or not. At

this low power, the signal is transmitted just by the windings. At the lower end of the frequency band, the stray field gets larger because of the increasing field strength, which can be seen in a significant drop during the measurement without the core. If the core is inside the windings, the iron concentrates the lines of force and leads them through the windings so that the drop is reduced to a minimum. If these values are specified in a datasheet, the frequency response indication would be as follows: 10 Hz (-0.6 dB) to 300 kHz (-2 dB).

However, the same transformer measured at full load and modulated using the corresponding tube reaches a frequency response of 20 Hz to 20 kHz (-0.1 dB) and 50 kHz (-3 dB)—which at first glance and in contrast to preceding measurements does not look that good, but reflects reality. When comparing frequency response indications, one should always determine how the measurements were carried out! If this information is not contained in the datasheet, it is recommended to ask the manufacturer or the supplier whether the specified frequency response is reached at full load and under real conditions, so that there are no unpleasant surprises after the purchase.

Frequency responses of output transformers should always be measured near the output maximum that has the highest significance. It is a good gauge when measuring with an amplifier at approximately 1 dB below full modulation. This ensures that the amplifier does not reach the limitation. The value of 1 dB below full modulation corresponds to around 80% of the maximum output of an amplifier and produces realistic values. It also makes sense to measure the amplifier or the transformer at about 10% of the maximum output power. In this way, one obtains the frequency response behavior as it occurs when listening under normal conditions in living rooms.

Magnetism

In connection with the considerations of iron cores up to this point, another important detail needs to be addressed. In this respect, much has been discussed as to the fine resolution of the sound by using special alloys of iron cores. At first, one should think about how magnets work. When you put a magnet under a sheet of paper and pour iron filing onto the sheet, those iron filings will align along the magnetic lines of force. For a long time, the magnetism inside iron cores and magnets has been explained using this model. The smallest magnetic particles align according to the magnetic field. However, it is a fact that there is absolutely no externally visible change in the magnetic material of the particles.

The invention of audio recording on magnetic

tapes, has made magnetism usable. However, the development of computer hard disks required delving further into the subject matter. More data was supposed to be recorded and arbitrarily played back into a smaller and smaller space. Actually, it is not the iron particles that turn under the influence of a magnetic field, it is the electrons that change their behavior. In order to understand, you need to look at the structure of atoms.

The classic atomic model presented by Niels Bohr in 1913 assumed that negatively charged electrons revolve around the positively charged nucleus. The filling of the shells with electrons is defined by the expression $2n^2$ whereas n defines the relative shell. Therefore, the first shell is completely filled with two electrons, the second one with eight, the third one with 18, etc. Now, the chemical elements have the feature, for example, the fourth shell gets filled although the third one is not yet completely filled. Furthermore, for example the maximum of 18 electrons of the third shell do not move around the nucleus on a spherical surface. The electrons have different distances from the nucleus and feature different energy levels, which are named using the letters s, p, and d.

Basically, the electrons have the characteristics of jumping to the lowest possible energy level. Within each shell, two electrons are at the lowest energy level s. To demonstrate that using the example of iron, the atomic nucleus has 26 positively charged protons and 26 electrons that revolve around the nucleus. In the first shell, two electrons are at the lowest energy level s and this shell is completely filled. In the second shell, two electrons are at energy level s and six electrons are on the p level. So, this shell is completely filled. Now, there is the third shell, which can contain a maximum of 18 electrons. As always, the energy level s is completely filled with two electrons. With six electrons, the p level is full. The third shell of the iron has six electrons at the d level. Instead of filling the third shell, the fourth shell is created with two electrons at the s level.

Now, the question is, what does all this have to do with magnetism? Research published in 1988 found that the electrons do not just circle around the atomic core, but they also turn around their own axes. This is similar to our solar system. The sun revolves around its own axis. The planets revolve around the sun and also rotate around their own axes which stabilizes the entire system, the physical principle of the rotating top. It is the same with the electrons that revolve around the atomic nucleus. This happens pairwise, whereas one electron spins anticlockwise around its axis and the other one clockwise.

About the Author

Gerhard Haas has written books as well as articles for *Elektor* Germany since 1995. *The Roehren Sonderheft* magazine was his creation, along with some partners. The first issue came out in 2005. Now, there are 10 issues containing a collection of interesting articles. Gerhard has also developed many electronic circuits and equipment. In the last four decades, he primarily worked on tube amplifiers and the circuitry around them, including the audio transformers, filter chokes, power transformers. He sold his company, Experience Electronics, several years ago to focus on designing tube amps and audio transformers, write articles for *Sonderheft* and finish his book *High-End Mit Roehren*. The first copy of his book was printed in 1995. The follow up was published in March 2017.

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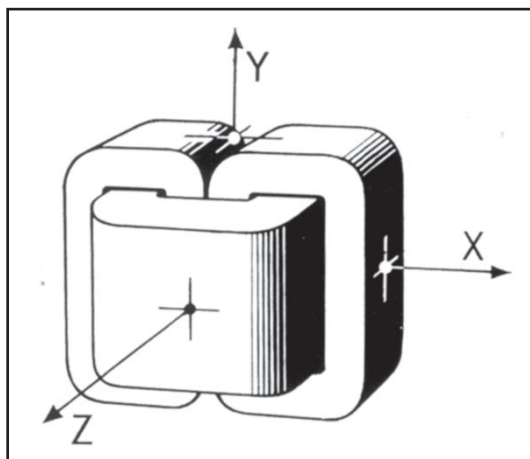
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Figure 2: The magnetic main axes of a transformer are shown based on this cut-tape wound core.



In the third shell of the iron with its six electrons on the energy level d comes into play. If the iron is non-magnetic, three electrons rotate anticlockwise and three electrons rotate clockwise. However, if an (electro)magnetic field is generated, two electrons of the third shell at the energy level d change their sense of rotation; the iron atom is magnetic. Here, the two electrons of the fourth shell at the energy level s act like a shielding from the third shell. This explains what is going on magnetically. It is not the material particles that rotate, but only two electrons per iron atom change their sense of rotation.

Changing the sense of rotation of an electron requires a certain unit of energy. Since the German physicist Max Planck laid the foundation for quantum physics, it is known that there is a smallest unit of energy, the Planck's constant. Energy is not continuous, but it develops by smallest fixed jumps established as natural constants. Therefore, a certain energy quantum is required to change the sense of rotation of two electrons in the shell of an iron atom. This ends the discussion about the influence of the iron core on the resolution. The relative alloy, depending on the iron content, determines the behavior.

Magnetic Iron

Now, the question is, why does the iron sometimes stay magnetic and sometimes it doesn't? The iron atoms organize in a cubic crystal shape. When this crystal lattice is disturbed by foreign atoms (e.g., carbon), the magnetism is preserved and the result is a permanent magnet. To describe these effects in detail would be beyond the scope of this article and it is also too complex. It is not the focus, because here, we are looking at the characteristics of iron cores for transformers that must not be permanently magnetic.

Only so-called soft iron is worth considering for building a transformer, normally, this is iron

alloyed with silicon. The easier it is to magnetize and demagnetize or remagnetize the iron, the more suitable the iron is for alternating current applications in transformers. Nickel also has magnetic properties, it is ferromagnetic. The term "ferromagnetic" is derived from the most frequent magnetic element iron (in Latin: ferrum). The third ferromagnetic element is cobalt which is mainly used in magnets. An alloy containing 75% nickel and 25% iron is used for producing the so-called mu-metal. MI stands for magnetically impermeable (in German MU-metal). This alloy offers an excellent magnetic conductivity, but it can't handle high modulation. Grain-oriented silicon-iron alloys can be modulated up to 1.8 T. The degree of the magnetic modulation also depends on the lamination.

As I already mentioned in Part 1 of this article, the proof voltage and, therefore, the operational safety of a transformer is of vital importance in tube power amplifiers. If there is a static shield between the primary and secondary winding in a mains transformer, and a voltage proof of 4,000 V has been tested between the two windings, there are very high mains safety reserves. Normally, the test is carried out with an alternating current of 50 Hz or 60 Hz. The voltage peak is by $\sqrt{2}$ higher, which corresponds to DC voltage of 5,656 V.

Output transformers for tube amplifiers can be tested in the same way. You have the certainty that if there is an interruption of the speaker cable, no high voltage will reach the speaker terminals. With a push-pull AB and single-ended type-A power amplifiers, the famous ramping of the primary voltage occurs if no load is present on the secondary side. A well-built transformer can withstand these voltage peaks. Then, the voltage flashovers occur inside the tube, at the socket, or inside the wiring, if no fuse is tripped. The repair of such defects is still cheaper than replacing the output transformer. In former times, glossy paper was used as insulation and defective transformers were nothing unusual. A galvanically isolated transformer tested for high-voltage resistance is basically a kind of insurance policy and also provides good protection for the speakers.

Assembling the Pieces

As for the assembly of transformers, chokes, and mains transformers, many consumers prefer the classic installation on the chassis being equipped with a cover. The most important thing is that the magnetic main axes of the mains transformer and the output transformer are turned by 90°, otherwise their magnetic stray field will influence each other. The axes are shown based on a cut-tape wound core shown in **Figure 2**. Y is the main stray axis. The stray

field of the coil is the Z-axis, and X is the the core. A wrong arrangement can express itself by increased humming and a reduced crosstalk attenuation in stereo amplifiers. If both output transformers are installed with their main stray axes being located one behind the other, there will be magnetic crosstalk. If they are installed so that the Y-axes are parallel to each other, they influence each other, too. The magnetic main stray axis Y of the mains transformer should not be aimed at the Y-axis of the output transformers, but at the X-axis. In this case, the interaction is lowest. Likewise, the Y-axis of the mains transformer should not be aimed at the amplifier input, even if the arrangement would be more comfortable this way.

If the amplifier is humming, the mains transformer is quickly held responsible, although the wrong installation often is the cause. If the chassis is made from magnetic steel sheet, it will also be excited leading to magnetic distortions. Non-magnetic material is always preferable. Those who install single-ended type-A transformers or chokes directly onto the steel chassis with steel covers, from a magnetic point of view, partially bypass the air gap. This changes the inductance and the permissible direct current preload. Normally, the required fixing components for these

components consist of non-magnetic aluminum or brass. Where steel parts are inevitable, spacers made from non-magnetic material are inserted between them and the core, or the air gap is increased. However, if transformers and chokes are sometimes installed on steel sheet and sometimes on non-magnetic chassis, you get a different behavior from components featuring an air gap. Furthermore, a steel sheet chassis can act as resonance body due to the magnetic excitement of the mains transformer resulting in audible mechanical humming.

Matching Transformers

Now, let's take a look at the topic of matching transformers (e.g., between the preamplifier and the power amplifier). People often want transformers with a matching of 600 Ω to 600 Ω . In this case, a 1:1 transformer is meant, but it is referred to as 600- Ω matching. Discussing the entire transformer theory would go beyond the scope of this article. Basically, looking for a 600- Ω -to-600- Ω transformer actually means a voltage matching of 1:1 with the desired galvanic isolation, but that is mostly misinterpreted.

A real resistance matching over a wide frequency range, and this is what the audio range is, as from

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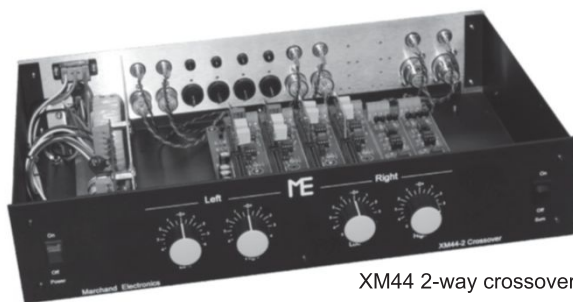
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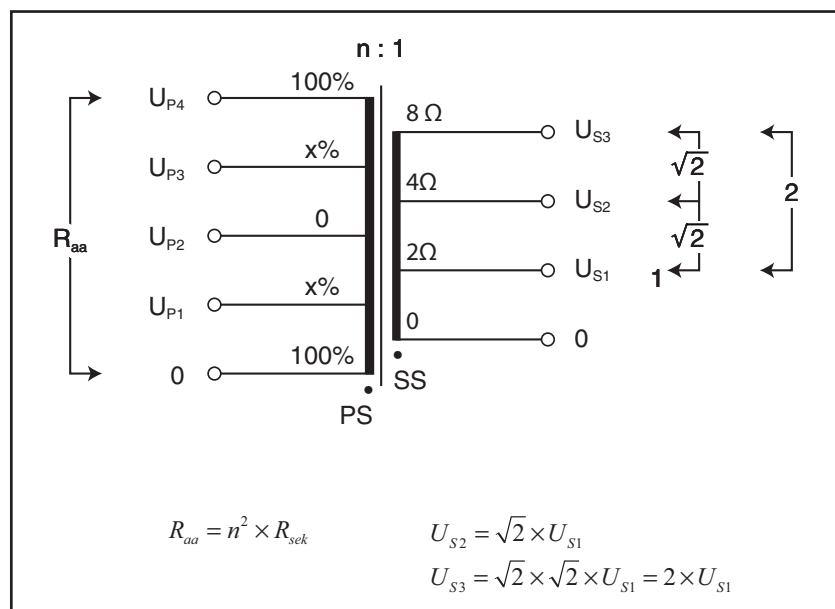


Figure 3: This is the circuit diagram of a typical output transformer for classic push-pull AB power amplifiers.

20 Hz to 20 kHz, the ratio is 1:1000, though, is virtually unfeasible. What is demanded and meant in today's audio technology is the voltage matching. There is also the current matching, which doesn't have any significance here, and the power matching.

If a galvanic isolation of two devices is desired, mostly a voltage level of 1:1 is implemented. Ideally, the signal source has an internal resistance of 0 Ω . In practice, this is never the case. So, when speaking of a matching 600 Ω to 600 Ω , this statement is closer to the power matching than to voltage matching. The usual standard level of 6 dBm means nothing but 1.55 V at 600 Ω , which corresponds to 1 mW. This definition still dates back to the era when a phone speaker with an impedance of 600 Ω and with 1 mW was considered sufficient volume for audibility.

Furthermore, back then, long-life tubes with minimum power output and amplifiers with high source resistances were used. Semiconductor technology and modern tube technology have changed that a lot.

What is demanded with respect to transformers between amplifiers in today's audio technology is the implementation and galvanic isolation of a voltage. Everything stays the same with a transformation ratio of 1:1. Input and output impedances are transformed 1:1, as well as it is the case for the current and the voltage; the use of such a transformer is uncomplicated. However, if the transformation ratio differs, it is absolutely essential to observe the rules of the Ohm's law. The impedances behave depending on the transformation ratio according to the formula:

$$n = \sqrt{\frac{R1}{R2}}$$

which results from Ohm's law. So, if a source has an internal resistance of $100\ \Omega$ and the transformer features a transformation ratio of 1:4, the result on the secondary side is $1600\ \Omega$. Consequently, the voltage matching would require the input impedance to be at least 5 to 10 times higher than the occurring source impedance, thus $8\ \text{k}\Omega$ to $16\ \text{k}\Omega$. So, the level error stays relatively small. If the transformer is, for example, terminated with $32\ \text{k}\Omega$ on the secondary side and the transformation ratio is 1:4, $2\ \text{k}\Omega$ will be present at the input.

Furthermore, the copper resistance needs to be considered with real transformer. If it is, for example, 200 Ω , it acts like an additional voltage divider. Normally, that can be compensated without any problems by turning up the volume or by increasing the amplification in the amplification stage. Basically, for all transformers, when optimizing the amplifier design, the copper resistances of the transformers as well as those of the mains transformer and filter chokes have to be considered in the calculations. However, it depends on the abilities of the developer to determine to which extent the components can be maxed out and used optimally.

Source and Load Impedances

If you want to optimally use transformers, the source and load impedances must be correct. If a transformer is very highly modulated on its primary side and if there is no real termination on the secondary side, the frequency response can be considerably bent. A tube can be used as driver stage and the transformer works on the control grid of the power tube, the typical case of the intermediate transformer. It is always best if the source impedance approaches zero and the load represents a defined real termination. This is possible in an almost ideal way using transformer coupling between the preamplifier and the power amplifier. In this way, a smooth frequency response is achieved which reaches far beyond the audio range.

The procedure for determining the terminals is relatively easy. An ohmmeter can be used to determine which windings belong to each other. When windings have thin wire and a high number of windings, it is possible to determine the sequence due to the resistance values. It is often somewhat more difficult to determine the resistances of the low-impedance speaker windings. The ratios can be determined by the voltage test.

Figure 3 shows the circuit diagram of a typical output transformer for classic push-pull AB power amplifiers. The terminal designations have been intentionally marked abstractly and not using the typical designations for anode, screen grid, center,

etc., as the presence of an unknown object has to be assumed first. Since normally the primary winding is applied first, the start of the winding lies near the core. In general, the primary winding of tube output transformers can also be recognized by the thinner wire. This connection is marked as PS in Figure 3, which stands for Primary Start. Now, the ohmmeter can be used for checking how the other primary terminals behave.

The DC resistances of the tapings should develop with increasing ohm values. In this way, you already obtain the sequence of the tapings that is 0, UP1, UP2, etc. Now, there are ways to determine the transformation ratio and the phase relation. For example, you can use a mains transformer to feed a voltage of about 10 V to 30 V between 0 and UP1, which surely excites the iron and makes the transformer work as such. The voltage must increase as the tapings increase. The voltage ratios that help determine the center is U_{p2} in Figure 3. Likewise, the percentage of the screen grid tapings can be determined.

Ideally, the DC resistances are equal to the tapings and the winding ends, when seen from the center tapping. Absolute symmetry can only be achieved using a complex two-chamber winding. However, in this case, there may be already minor differences as there are always manufacturing tolerances. But that has no real influence on the transformer's overall behavior. If, for example, the difference of the two half windings is 100 Ω and a current of 50 mA flows through the tube, the voltage drop is just 5 V. When comparing that to an operating voltage of 300 V to 500 V, this is negligible. An impairment of the transformer behavior in terms of the low-frequency signal is not to be expected. The reason for the deviation of the DC resistances is that the higher impedance winding lies further outside in the wrap, showing a higher copper resistance due to the higher wire length.

After clearly determining the start and end of the primary winding, again apply an AC voltage, this time from 0 to UP4, which requires a mains transformer with 10 V to 30 V. Now, measure the secondary voltages. Use a dual-channel oscilloscope to check the phase relation. If this is known, the secondary voltages US1 to US3 can be easily determined. With tapings measuring 2 Ω , 4 Ω , and 8 Ω , they have to act as shown in Figure 7. From 2 Ω to 4 Ω , the voltage increases by the factor $\sqrt{2}$ and again by $\sqrt{2}$ to 8 Ω . The voltage from 2 Ω to 8 Ω is doubled. In this way, it is easy to determine the position of the terminals.

Classic tube amplifiers from 1950 to 1970 often feature speaker terminals of 4 Ω and 16 Ω , but also 15 Ω instead of 16 Ω . The 15 Ω are not really

significant, as they are rather a nominal value in order to handle the at that time common speaker impedances. It was much easier for the transformer designers to come up with 4 Ω and 16 Ω than with the tapping also at 8 Ω . With 4 Ω to 16 Ω , you could just double the number of windings. For operation with 4 Ω , the two windings are connected in parallel, which reduces the copper losses. In contrast to 16 Ω , with 4 Ω there flows the current increased by $\sqrt{2}$. So, because of the ratios, the frequency response and the performance remain unchanged. The greatest art is to achieve that also with an intermediate tapping, which features the winding number ratio of 1: $\sqrt{2}$ for 8 Ω .

Now, the transformation ratio has to be determined. To do so, feed a slightly higher voltage of about 20 V to 40 V on the primary side between the start and the end of the winding, because the step-down transformation needs to be considered. If the secondary voltages are too low, the measuring error may increase. If the windings are not terminated on the primary or on the secondary side, the transformer is operated in idle mode. Now, the secondary voltage is measured, whereby the entire secondary winding between the points 0 and US3 would be best. If the result is a voltage ratio of 24:1, the impedances relevant for AC operation can be determined. If the winding from 0 to US3 is terminated with 8 Ω , according to the formula $R_{aa} = n^2 \times R_{sek}$, therefore, $R_{aa} = 24^2 \times 8\Omega = 4608\Omega$, or 4.6 k Ω . In this case, the DC resistances of the copper windings are not considered. They don't play any role when considering the impedance. It is also possible to load the 8- Ω winding with 12 Ω , then, according to formula, the value R_{aa} will be 6912 Ω .

In principle, it doesn't matter to the transformer, how it is terminated, it just transforms what it "sees." However, the termination impedances should not vary arbitrarily, since each transformer has been optimized for a specific task. Here, the copper resistance also plays a role. When operating a winding designed for 8 Ω at 4 Ω , higher losses occur inside the transformer because of the copper resistance and the increased current flow. This is connected with power losses as instead of the maximum power possible arriving at the speaker, a considerable percentage is converted into heat inside the transformer.

For output transformers that have to transform power, the copper resistance can be compensated by modified transformation ratios, provided that the termination is correct. Here, the aim is not the mathematically optimal transformation ratio, and therefore the theoretical value R_{aa} , but the adaptation is carried out so that the copper resistance doesn't cause any noteworthy power losses. At the same time, that should not distort the frequency response.

Likewise, the primary impedance may differ from the classic values in old data books in favor of better distortion properties of the amplifier. These adaptations require a lot of know-how, many years of experience in material selection, and a sophisticated winding technique in transformer construction.

Determining Frequency Response

The basic frequency response of a transformer can be determined by simple means. A sine generator is used for feeding a voltage of 5 V or more, depending on what the generator can provide. It should be checked whether the generator has a stable level over the entire frequency range in question, in order to avoid incorrect measurements. The voltage is applied to the entire primary winding, from the start of the winding to its end. The entire secondary winding is terminated with a resistance of 8 Ω, to stay with the previous example. Now, the level is measured at the 8-Ω terminal at 1 kHz using it at reference. Afterward, the frequency of the generator is changed downward and upward and the points -1 dB and -3 dB are determined. This can be easily done with a level meter. The primary injection by the waveform generator has a very low impedance, normally it is a source impedance of 50 Ω, so that the resonance frequencies in the transmission range are sufficiently attenuated (as already described in the Figures 1 to 6 in Part 1 of this article series). The secondary termination is necessary because if a transformer is not terminated, you do not get useful figures. This measurement is used for checking the frequency range of a transformer within a range of around 10 mW, which permits you to make an initial judgment. However, this measurement does not replace the real performance test under real conditions in the relevant tube amplifier.

Further important parameters of a transformer are the ohmic resistances of the windings, the primary and the stray inductance. The copper resistances can be easily determined with the ohmmeter. Here, the primary resistance is of the most interest. The secondary resistances of the windings should be less than about 20% of the nominal impedance, so at 8 Ω they should be less than 1.6 Ω. If the copper resistance is too high, together with the speaker impedance, it acts as a voltage divider, which results in power losses.

Since the copper resistance has a very low impedance, it needs to be considered that the supply resistances of the measuring cables together with the resistance of the connection terminals already have a higher value than the secondary winding of the transformer. You need to be absolutely careful about what is really measured here! Because of the higher

resistance of the primary winding, the measuring error is negligible.

The stray inductance is very easy to measure. The transformer's entire secondary winding is short-circuited and the LCR meter is connected to the entire primary winding. Normally, the value lies in the low millihenry range.

Measuring the primary inductance is slightly more complex. It is best to measure more than one generator measurement. Also take a measurement referring to an output of about 100 mW. In this case, the speakers are highly effective to some extent in the moderate volume range. And, a significant value is obtained. In case of a secondary impedance of 8 Ω, 100 mW correspond to a secondary voltage of 0.9 V. When assuming a transformation ratio of 1:25, 22.5 V need to be fed at the entire primary winding.

An adjustable mains transformer is a good voltage source as the measuring frequency is fixed to 50 Hz or 60 Hz. A mains transformer with several secondary tapings between about 20 V and 50 V is just as good. Since the primary inductance is important at the lower frequency band end, one easily obtains a practical value. Now, the current consumption needs to be measured. The primary inductance can be determined using a simple calculation. Ohm's law says $U = R \times I$. Here, R is not an ohmic resistance, but the frequency dependent impedance $\omega \times L = 2 \times \pi \times f \times L$. That is inserted in the basic formula and then flipped for L:

$$L = \frac{U}{2 \times \pi \times f \times I}$$

As an example, we assume an input voltage of 22.5 V, a frequency of 50 Hz and a measured current draw of 5 mA, so the primary inductance is:

$$L = \frac{U}{2 \times \pi \times f \times I} = \frac{22.5V}{2 \times \pi \times 50 \times 0.005A} = 14.33H$$

With a transformer with very high-quality sheets and a high number of primary windings, which can provide the full power even at 20 Hz, the primary current may be very low, which results in a higher measuring error. In this case, a higher input voltage is to be applied in order to obtain useful results.

Conclusion

With these observations and relatively simple measuring procedures, anyone will be able to test the quality and assess terminal assignments of a transformer. This will enable us to predict the behavior of a transformer in an amplifier circuit with a preliminary evaluation. Finally, we can evaluate whether a transformer is suitable for the intended purpose. 



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Design and Performance of Power Supplies for Hollow-State Amplifiers

By
Richard Honeycutt
 (United States)

The design of power supplies for hollow-state amplifiers affects amplifier performance. **Figure 1** illustrates the block diagram of a generic power supply. The input to the supply is the AC mains voltage. The transformer changes the voltage to typically 30 V to 500 V needed as B+ for the output tubes. The rectifier converts the AC voltage to pulsating DC. Rectifiers can be either half-wave (conducting only on the positive half-cycle) or full-wave (conducting on both half-cycles). The filter levels out the pulsating DC, producing almost pure DC when there is no load current drawn. Filters can be categorized as capacitive, inductive-input, or pi filters (see **Figure 2**).

Voltage Regulation: Line and Load

Most power supplies' output voltages vary with the AC mains voltage and with the load current drawn. The variation in output voltage is quantified

by the line regulation and load regulation.

$$\text{line regulation} = \frac{V_{\text{out,max line}} - V_{\text{out,min line}}}{V_{\text{out,normal line}}} \times 100\%$$

In other words, the line regulation represents the total variation in output voltage between high-line-voltage and low-line-voltage conditions, as a percentage of the normal output voltage.

$$\text{load regulation} = \frac{V_{\text{out,no-load}} - V_{\text{out,max load}}}{V_{\text{out, rated}}} \times 100\%$$

So the load regulation specifies the percentage by which the output voltage varies as the load current is changed from none to maximum. A lower percentage voltage regulation indicates better performance. Although load regulation depends

Figure 1: This is a block diagram of an unregulated power supply.



upon the component values used, line regulation is generally not quantified for typical power supplies used for hollow-state amplifiers, since very few such supplies incorporate voltage regulator stages. An exception is that the screen grid supply for some power amplifiers is regulated, but even this is the exception rather than the rule.

Hum

Any power supply's output voltage contains some AC ("hum" or "ripple") voltage in addition to the DC, which is just another way of saying that no power-supply filter is perfect. Hum can be specified as a ratio of the AC component to the DC component, either as a percentage or in decibels. Usually the hum is specified with the amplifier producing a given output power level, perhaps full output.

Rectifier Types

As mentioned earlier, rectifiers can be built as either half-wave or full-wave types. These are illustrated in **Figure 3** and **Figure 4**, respectively. The full-wave bridge rectifier, which uses four diodes and does not require a center-tapped transformer, is not illustrated, even though it is a common design in circuits in which solid-state diodes are used rather than hollow-state ones.

Full-wave rectifiers provide better load regulation and lower hum than half-wave ones, assuming the same filter design is used for both. This difference in performance results from the full-wave rectifier producing an output pulse on each half-cycle, giving a higher average output voltage and also a higher fundamental hum frequency. The higher the hum frequency, the lower the value of capacitors and inductors needed to provide a given degree of filtering.

Whether a half-wave or a full-wave rectifier is used, the rectifier introduces a resistance in series with the load, worsening load regulation. For a solid-state diode, this resistance is about 33 Ω , and does not vary much in the 100-mA-to-300-mA range typical of hollow-state power amplifier B+ currents. A hollow-state full-wave rectifier may have a resistance of about 260 Ω per section for a plate current of 100 mA, dropping to 186 Ω for a plate current of 300 mA.

Filters

The output voltage of a power supply using a given transformer secondary voltage depends upon the type of filter. If the first filter component the input voltage encounters is a capacitor, as in the

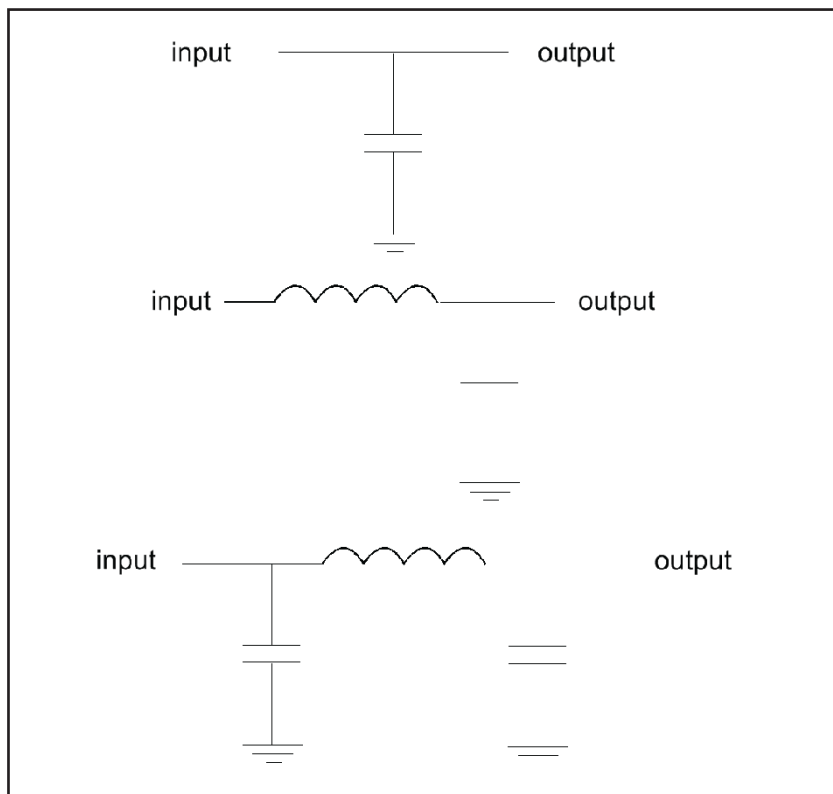


Figure 2: These are the basic power supply filter types: capacitive (a), LC (b), and pi (c).

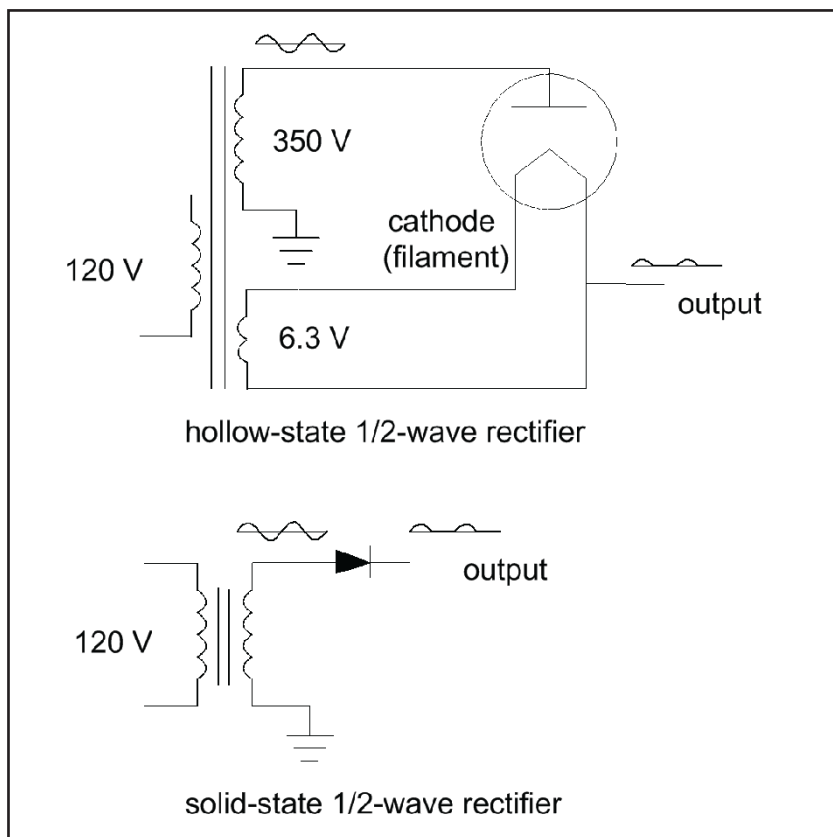


Figure 3: Here are hollow- and solid-state half-wave rectifiers.

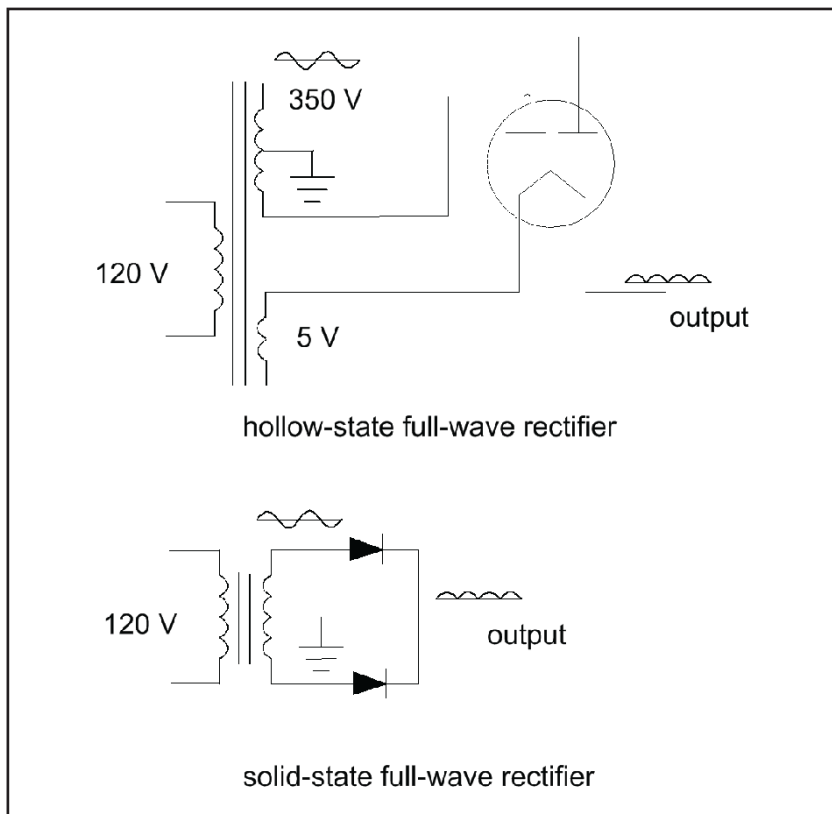


Figure 4: These are hollow- and solid-state full-wave rectifiers.

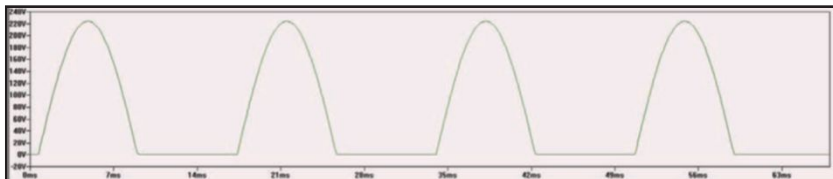


Figure 5: This is the output wave of a half-wave rectifier, unfiltered.

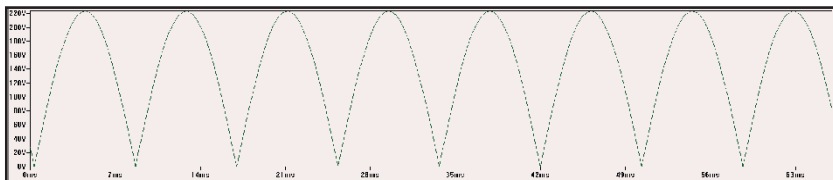


Figure 6: Here is the output wave of a full-wave rectifier, unfiltered.

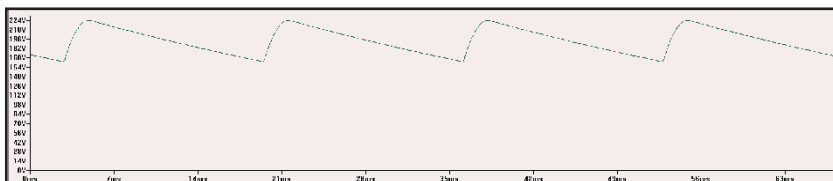


Figure 7: The output wave of a half-wave rectifier is shown with a capacitive filter and 100 mA of load current.

capacitive and pi filters, the capacitor will charge to the peak value (1.414 times the RMS value) of the input voltage, and that will be the no-load voltage of the power supply. If the first component is an inductor, the output voltage will be the average value (0.9 times the RMS or 0.636 times the peak value).

Capacitor-input filters have poorer voltage regulation than inductive-input ones, because the input capacitor discharges into the load between peaks of the input voltage. The filter choke in an inductor-input filter always has some DC resistance, causing the output voltage to drop as more load current is drawn. Since the resistance is not very large, this effect is less pronounced than a capacitive filter's drop in output voltage due to capacitor discharge.

The hum present in the output voltage after filtering depends upon the fundamental frequency of the ripple wave in the input voltage: 100 Hz or 120 Hz for a full-wave rectifier, or 50 Hz or 60 Hz for a half-wave rectifier.

Since the capacitive reactance is twice as high for the ripple frequency from a half-wave rectifier, the capacitor filters half as well as it would for a full-wave rectifier. Adding additional series inductors or resistors, interspersed with shunt capacitors, will decrease the hum.

Most hollow-state power amplifiers use a capacitor-input filter for the output tubes, followed by an inductor and/or RC sections to clean up the B+ for the input and phase-splitter stages. These additional filter stages have little effect upon voltage regulation and hum, because the Class-A tube stages draw nearly constant current, unlike push-pull output stages.

Waveforms and Performance

Having discussed the basics, let's now look at some actual examples showing output voltages and waveforms. All of the examples use a 159 VRMS transformer secondary. **Figure 5** and **Figure 6** show the output waves of half-wave and full-wave rectifiers, respectively. Notice that the output wave of the half-wave rectifier spends half of its time at zero volts, reducing the average output voltage and allowing a longer time for filter capacitors to discharge through the load.

Figures 7–10 show the output waves for half-wave and full-wave rectifiers filtered by a 20 μF capacitor. Notice that when the half-wave rectifier's load current increases from 100 mA to 300 mA, the P-P ripple voltage increases from 58 V to 116 V, the average DC voltage decreases from 195 V to

172 V, and the peak voltage remains constant at about 225 V. For the full-wave rectifier at 100 mA load, P-P ripple is 27 V, average DC voltage is 210 V, and the peak voltage is 223 V. For a load current of 300 mA, the P-P ripple voltage is 65 V, the average DC voltage is 190 V, and the peak voltage remains unchanged. None of these simulations included the effects of resistance in the transformer or rectifier.

Figure 11 illustrates the effect of increasing the filter capacitor from 20 μF to 60 μF , using a full-wave rectifier with a 300 mA load. Comparing this to the circuit with the 20 μF capacitor used to generate Figure 10, we see that P-P ripple voltage is decreased to 31 V and average DC voltage is increased to 208 V.

If we add the effects of resistance in the rectifier, the transformer, and the filter choke, the ripple and voltage regulation will change. A solid-state rectifier such as a 1N4007 has a series resistance of about 33 Ω when the forward current is in the 100-mA-to-300-mA range. A typical power transformer used in a dual-6L6GC amp adds an equivalent resistance of about 40 Ω .

Figure 12 shows the output waveform of a

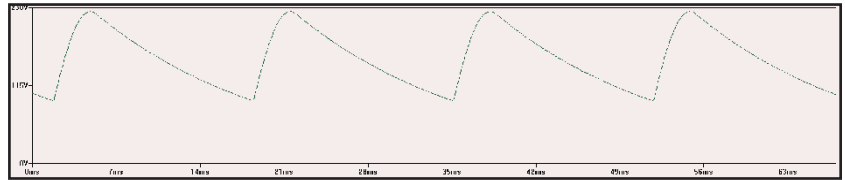


Figure 8: The output wave of a half-wave rectifier is shown with a capacitive filter and 300 mA of load current.

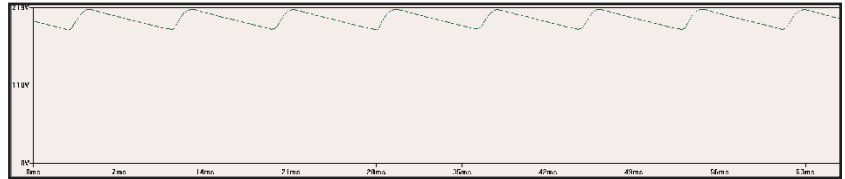


Figure 9: The output wave of a full-wave rectifier is shown with a capacitive filter and 100 mA of load current.

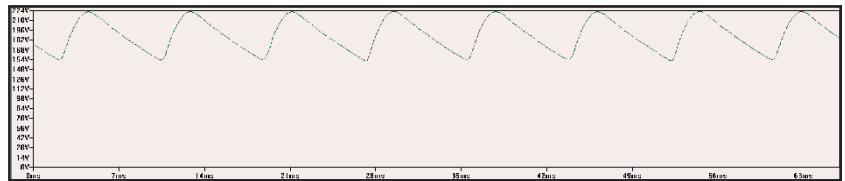


Figure 10: This is the output wave of a full-wave rectifier with a capacitive filter and 300 mA of load current.

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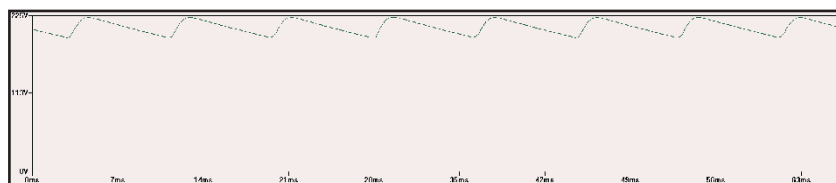


Figure 11: The output wave of a full-wave filtered rectifier is shown here with a larger filter capacitor and a load current of 300 mA.

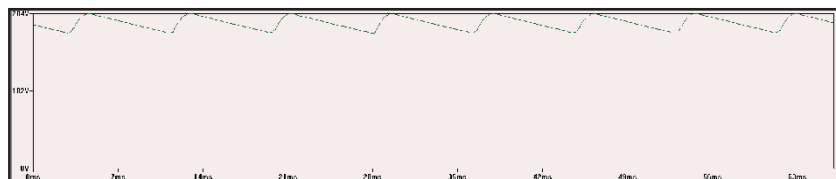


Figure 12: The output wave of a full-wave solid-state rectifier is shown with a larger filter capacitor and a 300 mA load and internal resistance.

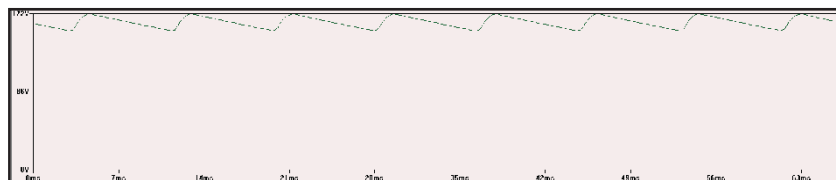


Figure 13: This output wave of a full-wave hollow-state rectifier is shown with a capacitor filter and a 300 mA load and internal resistance.

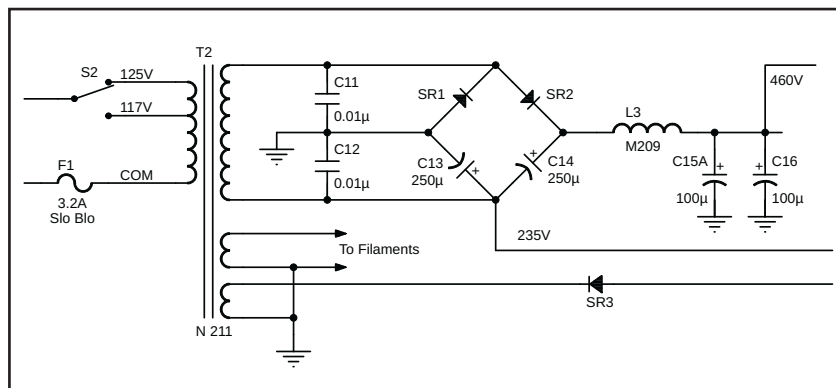


Figure 14: This is the high-voltage section of the power supply of the McIntosh MC75.

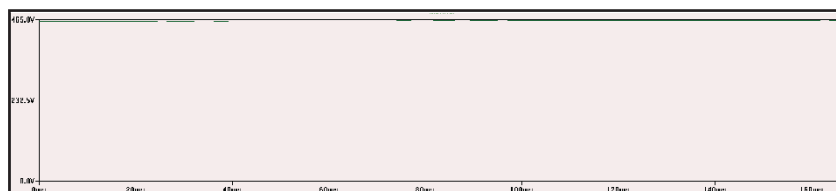


Figure 15: Here is the output waveform of the MC75 power supply's high-voltage section.

solid-state (not bridge) full-wave rectifier with a 60 μF capacitor filter and a 300 mA load. Comparing this to the waveform in Figure 11, we see that the resistances decrease the P-P ripple voltage to 27 V, and also decrease the average DC voltage to 191 V.

Since a vacuum diode has a higher resistance (260 Ω at 100 mA and 186 Ω at 300 mA), the performance of a power supply using a hollow-state rectifier will be different than that of one using a solid-state rectifier. **Figure 13** shows the effect of using a hollow-state rectifier. Again, both the ripple voltage and the average voltage decrease—this time, to 10 V and 131 V, respectively.

You would expect that the nonlinear resistance of the hollow-state rectifier would cause the change in average output voltage to be less for a hollow-state rectifier than for a solid-state one, and in fact it is. Using a full-wave vacuum diode, the power supply delivers only 19 V less into a 300 mA load (with P-P ripple of 4.5 V) than into a 100 mA load, compared to a 27 V drop (with 7 V P-P ripple) for the solid-state rectifier. But the output voltage is 17 V lower with the hollow-state rectifier.

One of the highest-performing power supplies for a hollow-state power amplifier is the one used in the McIntosh MC75. Its schematic, shown in **Figure 14**, reveals an inductive-input LC filter using an 8 H choke and two paralleled filter capacitors totaling 200 μF . At first glance, the rectifier looks like a bridge, but it is actually just a full-wave circuit with solid-state diodes and noise-reducing capacitors.

The output wave for a 300 mA load, shown in **Figure 15**, looks almost like pure DC; the P-P ripple voltage is only 0.7 V, and the DC output voltage is 460 V. Reducing the load current to 100 mA allows the B+ to rise to 507 V. Comparing the load regulation of the MC75 supply to that of the solid-state full-wave supply with a 60 μ F filter capacitor, we find that the MC75 has a load regulation of 9.3%, vs. 13.7% for the other circuit.

Although the load regulation of a hollow-state amplifier's power supply is not a critical factor, it is true that the peak unclipped output power is less when the B+ voltage drops under load. In a future article, we will see how large this effect is in actual amplifiers. 

Editor's Note: All audioXpress articles from 2001 to present can be found on the aX Cache, a USB drive available from www.cc-webshop.com.

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

July 21–22, 2018

CanJam London

**Park Plaza Westminster Bridge,
London, United Kingdom**

**[www.canjamglobal.com/
london2018](http://www.canjamglobal.com/london2018)**



CanJam is a network of headphone and personal audio shows, produced by Head-Fi.org, the world's largest community for headphone and personal audio enthusiasts. Since 2006, CanJam SoCal became a well-established event for one of the fastest growing segments of the audio industry, dedicated to everything that goes on and around our ears. CanJam is not only about headphones and in-ears, but also increasingly about all the audio players, preamps, and electronics that complete the experience. Attendees are always encouraged to bring their own music in order to get the best possible experience. And with products at all price ranges, educational seminars, and prize giveaways, there is something for everyone at CanJam.

CanJam returns to the heart of Central London at the ideally located Park Plaza Westminster Bridge Hotel. Featuring more than 100 leading brands in over 20,000 ft² of exhibition space, CanJam London 2018 will showcase the latest products and innovations in headphones and personal audio electronics in all price ranges—from \$50 earbuds to the world's finest headphone audio systems at \$50,000.

June 27–29, 2018

California Audio Show

**Hilton Oakland International
Airport, 1 Hegenberger Rd.,
Oakland, CA**

www.caaudioshow.com



The California Audio Show (CAS), produced by DAGOGO, is entering its eighth year and will take place once again at the Hilton Oakland from July 27–29, 2018. Response of the 2017 CAS from exhibitors and attendees was overwhelmingly positive. Exhibitors reported unprecedented attendance in each room and attendees reported extended listening time in exhibits.

It is now apparent what small shows such as the CAS can contribute to the health of the audio industry. The good American tradition of promoting audio shows in all the main cities and attract both the manufacturers, retailers, and potential buyers to regular meetings is keeping the high-end and consumer audio sectors afloat.

While other SoCal high-end shows have been struggling in result of the promoters' ambitions to grow nationwide and even internationally, the CAS has been consolidating its reputation as a small, local, but enjoyable show for visitors and successful for exhibitors. Now the second longest-running audio show in the US, CAS offers plenty of rooms for the presentations, accommodations for fair prices, and plenty of parking. With 50-plus brands already confirmed to attend, the show also includes an entire section dedicated to headphones and related products, and a CD, vinyl, and HRA sale shopping area.

For CAS 2018, the promoters have announced the creation of the HeadMasters Auditioning Room located in the Pacific Room, a quiet environment for CAS Attendees to sample high-end headphones and related products. Companies, each with a 6' x 30" long table will display their products inside the room. A waiting area will be set up for attendees who want their turn in a listening chair. All-access all-days Pass includes access to the HeadMasters Auditioning Room.

Recently renovated, the Hilton Oakland Airport consists of four buildings. Each three-story building has 9' ceilings with thick walls,

ideal for sound demonstration purposes. The show takes the entire Building V, which features the Seminar Room, seven large and 38 standard sound rooms. A neighboring building, Building IV, is a single-story convention center with a cafe/bar and a restaurant that houses the CAS Registration, the Concert/Seminar Space, vendor booth area and two additional sound rooms. Exhibitors will be able to continue music-playing in both Buildings well after show hours into the evenings.

July 28–29, 2018

Summer NAMM

700 Korean Veterans Blvd.

Music City Center, Nashville, TN

www.namm.org/summer/2018



The National Association of Music Merchants (NAMM), established in 1901, is the not-for-profit association that strengthens the \$17 billion global music, sound and event technology products industry. The association and its trade shows serve as the crossroads for professionals wanting to seek out the newest innovations in music, recording technology, sound, stage and lighting products.

At Summer NAMM attendees can develop relationships with the more than 500 exhibitors; dive into product demonstrations from more than 1,600 brands; find unique products; experience the new Retail Training Summit with six tracks for all areas of a company's retail team; learn best practices during NAMM U sessions, TEC Tracks, and the Top 100 Dealer Awards; discover the future of audio with A3E and Dante Certification Training; and experience the annual Opening Night Party featuring the 35th Annual American Eagle Awards honoring jazz legends Chick Corea and The Manhattan Transfer. Visitors can also network with industry leaders and peers during receptions, concerts, and explore Nashville.

August 7–9, 2018

2018 AES International

**Conference on Spatial
Reproduction—Aesthetics
and Science**

Tokyo, Japan

**[www.aes.org/conferences/2018/
spatial](http://www.aes.org/conferences/2018/spatial)**



Many practical technologies have been developed in the area of spatial audio since the last international conference in Tokyo, AES International Conference on Spatial Audio—Sense the Sound of Space—was held in 2010. The industry and all research institutions globally have managed to handle the technology and used it to create countless works since then. In the light of such technology-driven content creation, it may be the time to address the aesthetics in audio production.

The AES committee for this conference is expecting discussions in the following topics:

- How can the Spatial Audio technology contribute to music and other artistic productions?
- What technologies and scientific knowledge are expected from musicians and the content creators?
- What are the keys for connecting Science and Aesthetics in this area?

For more information about this and other conferences, visit www.aes.org.



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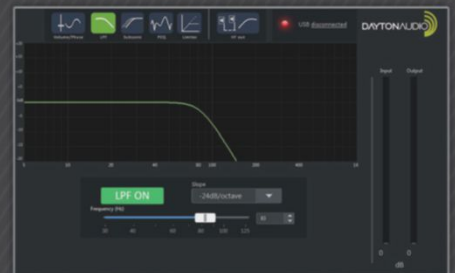
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